ISDN Primary Rate Interface Features Fundamentals — Book 2 of 3
Avaya Communication Server 1000

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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 7.6.

Features

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

Other changes

There are no other changes.

Revision History

March 2013	Standard 06.01. This document is up-issued to support Avaya Communication Server 1000 Release 7.6.
November 2011	Standard 05.03. This document is up-issued to support the removal of content for outdated features, hardware, and system types.
November 2010	Standard 05.02. This document is up-issued to support Avaya Communication Server 1000 Release 7.5.
June 2010	Standard 04.01. This document is up-issued to support Avaya Communication Server 1000 Release 7.0.
March 2010	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 Release 6.0.
May 2008	Standard 02.05. This document is up-issued to update the section "Network Alternate Route Selection (NARS)/Basic Automatic Route Selection (BARS)/Coordinated Dialing Plan (CDP)" as per CR Q01872218.

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 changes in technical content: • 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. This document contains information previously contained in the following legacy document, now retired: ISDN Primary Rate Interface Features Fundamentals (NN43001-569-B2).
July 2006	Standard 5.00. This document is up-issued to reflect changes in content: • Addition of Feature Packaging on page 1539 in the Network and Distinctive Ringing chapter. • Addition to Trunk Route Optimization chapter on page 1799. • Addition of Table 137 to Engineering and Configuration Guidelines chapter on pages 1953 to 1955.
August 2005	Standard 3.00. This document is up-issued to support Communication Server 1000 Release 4.5.
September 2004	Standard 2.00. This document is up-issued for Communication Server 1000 Release 4.0.
October 2003	Standard 1.00. This document is a new technical document for Succession 3.0. It was created to support a restructuring of the Documentation Library, which resulted in the merging of multiple legacy technical documents. This new document consolidates information previously contained in the following legacy 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: E.164/ESN Numbering Plan Expansion

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

The E.164/ESN Numbering Plan Expansion feature provides the capabilities to meet the International Telegraph and Telephone Consultative Committee (CCITT) recommendation

E.164 for Integrated Services Digital Network (ISDN) and Public Switching Telephone Network (PSTN) dialing.

The following enhancements are offered by this feature:

- Support of the numbering plan for ISDN and PSTN dialing has been increased to a 15-digit maximum, from a 12-digit maximum.
- Removal of the leftwise-unique restriction imposed on ESN Supplemental Digit Restriction and Recognition (SDRR) entry codes. For example, if 555 is an existing entry, an entry of either 55 or 5551212 can be entered.
- Addition of "allow" (ALOW) as a Supplemental Digit Restriction and Recognition (SDRR) entry type in LD 90. There are now nine entry types for Supplemental Digit Restriction and Recognition. One entry type (DENY), denies a call from going through if the digits in the SDRR entry match the dialed digits. There are seven other entry types that provide special treatments, if the digits in the SDRR entry match the dialed digits. The ALOW entry allows a call to go through, if the dialed digits did not match any entry within the Supplemental Digit Restriction and Recognition table.
- Special Number Translation (SPN) screening digits scheme is changed.
- Capability of the base features to store digits is increased to 31 digits.
- Capability to send up to 32 digits to the Extended Conference and Tone & Digit Switch card, allowing an International Number to be sent in a single Automatic Number Identification (ANI) message. Previously, only 16 digits were outpulsed in a single ANI message, which included the start digit, the end digit, and an optional information digit. This left only 13 or 14 digits of the actual International Number to be outpulsed in a single ANI message. This meant that the International Number was sent in two separate ANI messages.

The specific enhancements delivered by the E.164/ESN Numbering Plan Expansion feature are provided in the tables that follow.

Summary of expanded ESN functionalities

Table 1: Summary of ESN changes on page 22 summarizes the changes made to the ESN functionality.

The components that are not affected are not listed.

Table 1: Summary of ESN changes

Component	Old	Expanded
Maximum number of digits for a Special Numbering Translation screening.	11	19
Maximum number of Digit Manipulation Index (DMI) deletion digits.	15	19
Maximum number of Digit Manipulation Index (DMI) insertion digits.	24	31

Component	Old	Expanded
Maximum number of Supplemental Digit Restriction and Recognition (SDRR) tables, with BARS.	256	1500
Maximum number of Supplemental Digit Restriction and Recognition (SDRR) tables, with NARS.	512	1500
Maximum number of words for each Supplemental Digit Restriction and Recognition (SDRR) entry.	3	4
Maximum number of digits in each Supplemental Digit Restriction and Recognition (SDRR) entry.	7	10
Maximum length of Flexible Numbering Plan (FNP) Flexible Digit Number Length (FLEN) numbers for Special Number Translation (SPN).	16	24
Maximum length of Flexible Numbering Plan (FNP) Flexible Digit Number Length (FLEN) numbers for Trunk Steering Codes (TSCs).	16	24
Maximum number of digits for each Free Special Number Screening (FSNS) Special Number.	11	19
Total number of digits for screening under Free Special Number Screening (FSNS).	14	22
Maximum number of possible Supplemental Digit Restriction and Recognition (SDRR) entry types.	8	9 (ALOW is added). ALOW is an entry type in LD 90. This entry allows a call to go through, as if the dialed digits did not match any entry within the Supplemental Digit Restriction and Recognition table.
Restriction imposed on Supplemental Digit Restriction and Recognition (SDRR) entry codes.	Leftwise Unique.	None. The leftwise-unique restriction, imposed on SDRR entry codes, is removed. For example, if 555 is an existing entry, an entry of either 55 or 5551212 can be entered.

Summary of base features enhancements

Table 2: Summary of base features changes on page 24 summarizes the changes made to the base features functionality.

Note:

The components that are not affected are not listed.

Table 2: Summary of base features changes

Component	Old	Expanded
Maximum number of digits for an Autodial entry.	23	31
For the No Hold Conference functionality, the maximum number of digits for a Conference Autodial target DN entry.	23	31
Maximum number of digits for the Call Forward All Calls DN. Note that this enhancement does not apply to the M2317 telephone.	23	31
Maximum number of digits for the Internal Call Forward DN.	23	31
For the Phantom DN functionality, the maximum number of digits for the Default Call Forward DN.	23	31
Maximum number of digits for the Customer Call Forward DN.	16	23
Maximum number of digits sent in an ANI message.	16	32

Summary of Automatic Call Distribution (ACD) enhancements

Table 3: Summary of system ACD changes on page 24 summarizes the changes made to the ACD functionality.

The components that are not affected are not listed.

Table 3: Summary of system ACD changes

Component	Old	Expanded
Maximum number of digits for the ACD Night Forward DN.	23	31
Maximum number of digits for the ACD Interflow DN.	23	31

Summary of Customer Controlled Routing (CCR) enhancements

Table 4: Summary of CCR changes on page 25 summarizes the changes made to the Customer Controlled Routing functionality.

Note:

The components that are not affected are not listed.

Table 4: Summary of CCR changes

Component	Old	Expanded
Maximum number of Termination DN digits in the CCR ITR To-Route-A-Call message.	24	31

Summary of ISDN features enhancements

Table 5: Summary of ISDN changes on page 25 summarizes the changes made to the ISDN functionality.

Note:

The components that are not affected are not listed.

Table 5: Summary of ISDN changes

Component	Old	Expanded
For ISDN BRI basic call service line access (BRIL), the maximum number of Called Party Information Element digits.	24	31
For the ISDN QSIG interface, the maximum number of Called Party Information Element digits.	24	31
For the EuroISDN interface, the maximum number of Called Party Information Element digits.	20	31
For the ISDN BRI NI-1 Call Forward All Calls feature, the maximum number of digits for the forward DN.	20	31
For the ISDN BRI ETSI Call Forward All Calls feature, the maximum number of digits for the forward DN.	20	31
For the Overlap Signaling feature, the maximum number of digits that a user can dial before a SETUP message is sent over the D-channel.	16	24

Component	Old	Expanded
Note: Overlap Signaling is not supported on North American interfaces.		

Operating parameters

All existing operating parameters that apply to the system ESN functionality, base system features and ISDN PRI and ISDN BRI networking features apply to the E.164/ESN Numbering Plan Expansion feature.

The enhancements introduced by the E.164/ESN Numbering Plan Expansion feature, pertaining to Called Party Information Element digits (CDPN), and forwarded and redirected DNs do not pertain to the M5317 and M5209 ISDN BRI terminals. This is because these terminals follow the BRI ETSI standard, placing a maximum limit of 24 digits on Called Party Information Element digits and redirected DNs.

The 31-digit expansion, pertaining to the Call Forward All Calls DN, does not apply to the M2317 telephone, since its firmware limits entering a CFAC DN from the phone to a maximum of 23 digits.

Since the Network Call Trace feature can outpulse a maximum of 21 digits to be printed on the TTY, the 31-digit expansion does not apply to Network Call Trace.

The 16 digit maximum, pertaining to the Calling Party Number, Connected Party Number, Redirecting Number and Redirection Number, has not been expanded by the E.164/ESN Numbering Plan Expansion feature.

If all the digits of a DN are dialed automatically, such as for a Speed Call or for any redial feature, only the first 31 digits are processed. This means that any digits that are inserted by, for example, network signaling, pretranslation or digit manipulation, are counted as part of these 31 digits. Therefore, even fewer than the automatically dialed digits are processed.

Flexible Feature Codes (FFC) DNs have a length of 1-7 digits. Call Forward All Calls (CFAC), Internal Call Forward All Calls (IFC) and Remote Call Forward (RCFW) are supported by FFCs, and can be activated, deactivated and verified using FFCs. When activating CFAC, using the Remote Call Forward FFC (RCFA), or when verifying the Call Forward DN, using the Call Forward All Call FFC (CFWV), the Remote Call Forward FFC (RCFV), or the Internal Call Forward All Calls FFC (ICFV), the FFC DN is kept, using eight of the available 31 digits. For the CFAC DN or ICF DN, this means that only 23 possible digits can be stored or verified using FFCs. As a result, this limitation does not accommodate the 31 digit CFAC DN and IFC DN expansion, introduced by the E.164/ESN Numbering Plan Expansion feature.

Calling Party Privacy (CPP) allows a caller to block his or her name from appearing on the called party display. The call is marked as a CPP call by entering a CPP FFC, up to seven

digits long. This means that up to seven additional digits have to be stored for automatic dialing. These digits are stored in sequence, and subsequent digits are not be processed.

In the case of CPP outpulsing, the CPP outpulsed digits (typically *67 or 1167) are stored after CPP has been activated. If this call goes over a trunk, the CPP outpulsed digits are sent to the far end to indicate that the call is a CPP call. For ISDN calls, a CPP flag is marked in the ISDN message. For non-ISDN CPP calls, digits that are already stored, such as for Call Forward, Autodial, and Speed Call, can be pushed out when the CPP digits are inserted, since only 31 digits can automatically be processed. This means that up to four digits can be removed from the digits that are outpulsed.

The SPN expansion, provided by the E.164/ESN Numbering Plan Expansion feature, applies to the DPNSS1/DASS2 UDP Interworking feature.

The System Access Enhancement (SAE) feature impacts the accepted values for the Call Forward All Calls DN, and the Internal Call Forward DN, when entered in LD 10 and LD 11. This feature allows all values between 4-31 to be accepted, rather than the normal accepted values of 4, 8, 12, 16, 20, 24, 28, 31. Also, the default is 4, rather than 16.

The Called Party Information Elements expansion, provided by the E.164/ESN Numbering Plan Expansion feature, applies to ISDN QSIG GF Transport feature.

Feature interactions

The E.164/ESN Numbering Plan Expansion feature does not interact with the base system features and ISDN PRI and BRI networking features, other than as described in Table 1: Summary of ESN changes on page 22 to Table 1: Summary of ESN changes on page 22 and in the operating parameters.

Feature packaging

The E.164/ESN Numbering Plan Expansion feature does not introduce a new package. However, the following packages are required to support the new expansions.

For ESN expansions:

- Basic Alternate Route Selection (BARS) package 57
- Network Alternate Route Selection (NARS) package 58
- Flexible Numbering Plan (FNP) package 160

For ISDN expansions:

- Digit Display (DDSP) package 19
- Integrated Services Digital Network (ISDN) package 145
- Primary Rate Access (PRI) package 146
- Integrated Service Digital Network Signaling Link (ISL) package 147
- 2.0 Mbit Primary Rate Interface (PRI2) package 154
- Overlap Signaling (OVLP) package 184
- International Primary Rate Access (IPRA) package 202
- Meridian 1Extended Peripheral Equipment (XPE) package 203
- Basic Rate Interface (BRI) package 216
- Multi-purpose Serial Data Link (MSDL) package 222
- Integrated Services Digital Network Basic Rate Interface Trunk Access (BRIT) package 233
- Basic Rate Interface Line Application (BRIL) package 235
- EuroISDN (EURO) package 261
- QSIG Interface (QSIG) package 263

For base system features expansions:

- Optional Features (OPTF) package 1
- 2500 Phone Features (SS25) package 18
- Basic Automatic Call Distribution (BACD) package 40
- Automatic Call Distribution Basic, Package B (ACDB) package 41
- Automatic Call Distribution Advanced, Package A (ACDA) package 45
- 500 Phone Features (SS5) package 73
- Controlled Class of Service (CCOS) package 81
- Background Terminal (BGD) package 99
- Flexible Feature Codes (FFC) package 139
- Phantom Terminal Numbers (PHTN) package 254

Feature implementation

Task summary list

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

1. Table 6: LD 10 - Configure expansion for analog (500/2500-type) phones. on page 29
2. Table 7: LD 11 - Configure expansion for Meridian 1 Proprietary Phones. on page 30
3. Table 8: LD 12 - Configure expansion for Attendant Consoles pertaining to the maximum length of the Autodial DN. on page 31
4. Table 9: LD 15 - Configure the expansion pertaining to the Customer Call Forward DN. on page 31
5. Table 10: LD 23 - Configure the expansion pertaining to the ACD Night Call Forward DN and the ACD Interflow DN. on page 32
6. Table 11: LD 86 - Configure the expansion for the ESN data block. on page 32
7. Table 12: LD 86 - Configure the expansion for the Digit Manipulation data block. on page 33
8. Table 13: LD 86 - Configure the expansion for the Route List data block. on page 33
9. Table 14: LD 87 - Configure the expansion for a Free Special Number Screening Index. on page 34
10. Table 15: LD 87 - Configure the expansion for a Trunk Steering Code. on page 34
11. Table 16: LD 90 - Configure the expansion for Network Translation tables. on page 35

Note:

For analog (500/2500-type) phones, configure the expansion pertaining to the Autodial, Call Forward All Calls, Default Call Forward (for Phantom TNs) and Internal Call Forward DNs.

Table 6: LD 10 - Configure expansion for analog (500/2500-type) phones.

Prompt	Response	Description
REQ	NEW	Add new data.

Prompt	Response	Description
	CHG	Change existing data.
TYPE	500	Type of phone.
TN	l s c u	Terminal Number for Large System and CS 1000E.
FTR	ADL nn x..x	Autodial where nn = number of digits up to a maximum of 31 in Autodial DN and x..x = Autodial DN.
	CFW (4)-31	Enter the maximum number of digits in the Call Forward All Calls DN. Note that there is no default value.
	DCFW ll xxx...x	Enter the maximum number of digits in the Default Call Forward DN for a Phantom DN. ll = 4,8,12,16,20,24,28,31 (entries are rounded up to the next valid length.) xx...x = the Default Call Forward All Calls DN.
	ICF (4)-31	Enter the maximum number of digits in the Internal Call Forward DN for a Phantom DN.

Note:

LD 11– For Meridian 1 proprietary telephones, configure the expansion pertaining to the Autodial, Call Forward All Calls, Default Call Forward for Phantom TNs and Internal Call Forward DNs.

Table 7: LD 11 - Configure expansion for Meridian 1 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	l s c u	Terminal Number for Large System and CS 1000E.
KEY	nn ADL ll xxx...x	Phone function key assignments. Enter the maximum number of Autodial digits that can be stored. nn = Key number ll = 4,8,12,(16),20,24,28,31,NUL Entries are rounded up to the next valid length. An entry of NUL will disable the Autodial feature xx...x = the Autodial target DN (this is an optional entry.)
	nn CA ll xxx...x	Enter the maximum number of digits that can be stored for the combined Conference-Autodial DN. nn = Key number ll = 4,8,12,(16),20,24,28,31 (entries are rounded up to the next valid length.) xx...x = the Conference-Autodial target DN (this is an optional entry.)

Prompt	Response	Description
	nn CFW ll xxx...x	Enter the maximum number of digits that can be stored for the Call Forward All Calls DN. nn = Key number ll = (4)-31 (for M2317 phones, the accepted range is (4)-23) xx...x = the Call Forward All Calls DN (this is an optional entry.)
	nn ICF ll xx...x	Enter the maximum number of digits that can be stored for the Internal Call Forward DN. nn = Key number ll = (4)-31 xx...x = the Internal Call Forward DN (this is an optional entry.)

Table 8: LD 12 - Configure expansion for Attendant Consoles pertaining to the maximum length of the Autodial DN.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	2250	Attendant Console type.
TN	l s c u	Terminal Number for Large System and CS 1000E.
KEY		Attendant console function key assignments.
	nn ADL xxx...x	Enter the maximum number of Autodial digits that can be stored. nn = Key number xxx...x = the Autodial DN, up to 31 digits (this is an optional entry.)

Table 9: LD 15 - Configure the expansion pertaining to the Customer Call Forward DN.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	RDR	Redirection data.
CUST	0-99	Customer number as defined in LD 15.
...		
- CCFWDN	xxx...x X	Enter the Customer Call Forward DN, up to 23 digits (this entry is made for a new CCFWDN, or if the length of an existing CCFWDN has been expanded.) Enter X for no entry.

Table 10: LD 23 - Configure the expansion pertaining to the ACD Night Call Forward DN and the ACD Interflow DN.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	ACD	Automatic Call Distribution data block.
CUST	0-99	Customer number as defined in LD 15.
ACDN	xxx...x	ACD DN.
...		
NCFW	xxx...x	Enter the ACD Night Forward DN, up to 31 digits (this entry is made for a new Night Forward DN, or if the length of an existing Night Forward DN has been expanded.)
...		
IFDN	xxx...x	Enter the ACD Interflow DN, up to 31 digits (this entry is made for a new Interflow DN, or if the length of an existing Interflow DN has been expanded.)

Note:

LD 86 – For the ESN data block, configure the expansion pertaining to the maximum number of Supplemental Digit Restriction and Recognition blocks, and the maximum number of Digit Manipulation tables.

Table 11: LD 86 - Configure the expansion for the ESN data block.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
CUST	0-99	Customer number as defined in LD 15.
FEAT	ESN	ESN data block feature.
...		
MXSD	(0)-1500	Enter the maximum number of Supplemental Digit Restriction and Recognition (SDRR) blocks, for BARS or NARS. Enter 0 (the default) if no SDRR blocks are required.
...		
MXDM	(0)-2000 (0)-256	Enter the maximum number of Digit Manipulation tables.

Prompt	Response	Description
	(0)-32	(0)-2000 = for Flexible Numbering Plan (0)-256 = for BARS/NARS (0)-32 = for Coordinated Dialing Plan Enter 0 (the default) if no Digit Manipulation tables are required.

Note:

LD 86 – For the Digit Manipulation data block, configure the expansion pertaining to the maximum number of leading digits to be deleted and inserted.

Table 12: LD 86 - Configure the expansion for the Digit Manipulation data block.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
CUST	0-99	Customer number as defined in LD 15.
FEAT	DGT	Digit Manipulation data block feature.
...		
DMI	1-1999 1-255 1-31	Enter the Digit Manipulation table. 1-1999 = for Flexible Numbering Plan 1-255 = for BARS/NARS 1-31 = for Coordinated Dialing Plan. The maximum number of Digit Manipulation tables is defined using prompt MXDM, in the ESN data block.
DEL	(0)-19	Enter the number of leading digits to be deleted. Enter 0 (the default) for none.
INST	x...x x...x*y...y X	Enter the number of leading digits to be inserted. x...x = 1-31 x...x*y...y = for Special Common Carriers (SCCs), 1-30, including access number (x...x), delimiter (*), and authorization code (y...y) Enter X for none.

Note:

LD 86 — For the Route List data block, configure the expansion pertaining to the minimum Overlap Digit Length.

Table 13: LD 86 - Configure the expansion for the Route List data block.

Prompt	Response	Description
REQ	NEW	Add new data.

Prompt	Response	Description
	CHG	Change existing data.
CUST	0-99	Customer number as defined in LD 15.
FEAT	RLB	Route List data block feature.
RLI	0-1999	Route List Index.
DMI	1-1999 1-255 1-31	Enter the Digit Manipulation table. 1-1999 = for Flexible Numbering Plan 1-255 = for BARS/NARS 1-31 = for Coordinated Dialing Plan. The maximum number of Digit Manipulation tables is defined using prompt MXDM, in the ESN data block.
...		
OVLL	(0)-24	Enter the minimum Overlap Digit Length, pertaining to Overlap Sending. If 0 (the default) is entered, then the Flexible Digit Number Length (FLEN) determines whether Overlap Sending takes place.

Note:

LD 87 – For a Free Special Number Screening Index, configure the expansion pertaining to the maximum number of Special Numbers to be screened.

Table 14: LD 87 - Configure the expansion for a Free Special Number Screening Index.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
CUST	0-99	Customer number as defined in LD 15.
FEAT	FSNS	Free Special Number Screening.
FSNI	1-255	Free Special Number Screening Index.
SPN	1-19	Special Number to be screened.

Note:

LD 87 – For a Trunk Steering Code, configure the expansion pertaining to the maximum number of Flexible Numbers.

Table 15: LD 87 - Configure the expansion for a Trunk Steering Code.

Prompt	Response	Description
REQ	NEW	Add new data.

Prompt	Response	Description
	CHG	Change existing data.
CUST	0-99	Customer number as defined in LD 15.
FEAT	CDP	Coordinated Dialing Plan.
TYPE	TSC	Trunk Steering Code.
TSC	1-7	Trunk Steering Code.
- FLEN	0-24	Flexible Number.

Note:

LD 90 – For Network Translation tables, configure the expansion pertaining to the maximum number of Special Numbers digits, the maximum number of digits for Flexible Numbers, the maximum number of Supplemental Digit Restriction and Recognition digits for each entry, and the addition of "allow" (ALOW) as a new Supplemental Digit Restriction and Recognition type.

Table 16: LD 90 - Configure the expansion for Network Translation tables.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
CUST	0-99	Customer number as defined in LD 15.
FEAT	NET	Network Translation Tables data.
...		
TYPE	SPN	Special Number Translation.
SPN	xxxx xxxx x...x	Enter the Special Number Translation digits. Up to 19 digits can be entered. The SPN digits must be entered in groups of at most four digits, separated by a space. Up to five groups can be entered. The restriction of allowing one less digit if the first digit of a digit string is not "1", is removed.
- FLEN	(0)-24	Enter the number of Flexible Digits (the number of digits that the system expects to receive before accessing a trunk, and outpulsing the digits.)
...		
- SDRR	a...a DENY LDID LDDD DID DDD	Type of Supplemental Digit Restriction and Recognition. Restricted codes. Recognized local DID codes. Recognized local DDD codes. Recognize remote DID codes.

Prompt	Response	Description
	ITED ARRN STRK ALOW	Recognize remote DDD codes. Incoming trunk group exclusion digits. Alternate routing remote number. Allowed codes for ADM/MDM. Allowed codes. The maximum number of digits entered in response to the DENY, LDID, LDDD, DID, DDD and ALOW prompts, which follow, must be less than 10, or 7-m (8-m for 1+ dialing) for Central Office translation data (NXX), 10-m (11-m for 1= dialing) for Numbering Plan Area Code translation data (NPA), or 19-m for Special Numbers Translation data (SPN), where m = the number of digits entered for the prompt NXX, NPA or SPN. These numbers are no longer required to be leftwise unique. For non leftwise unique numbers, the longer number takes precedence over the shorter number. However, the exact same numbers (non leftwise unique and the same length) are still blocked.
- - DENY	x...x	Restricted number to be denied.
- - LDID	x...x	Local DID number to be recognized.
- - LDDD	x...x	Local DDD number to be recognized.
- - DID	x...x	Remote DID number to be recognized.
- - DDD	x...x	Remote DDD number to be recognized.
- - ALOW	x...x	Code to be allowed.

Feature operation

There are no operating procedures specified for this feature.

Chapter 4: Electronic Lock Network Wide/ Electronic Lock on Private Lines

Contents

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

Feature description

The basic Electronic Lock feature enhancement provides the following capabilities:

- The feature can be implemented network wide.
- The capability to define (for the locked phones), on a single customer basis, a Controlled Network Class of Service (CNCS) different from the Network Class of Service (NCOS), which is defined on a single phone basis.
- Locking can be implemented for Private DNs.

In a Meridian Customer Defined Network (MCDN) environment, Electronic Lock Network Wide can be used to change the Class of Service of a phone in a remote location. Electronic Lock Network Wide is activated or deactivated from any node by dialing the Electronic Lock Flexible Feature Code (FFC), the password, and the DN of the phone to be changed. Since the password length defined at the destination node is not known at the originating node, the

Station Control Password (SCPW) length (defined in LD 15) must be defined the same for all network nodes.

If the originating node has the FFC Confirmation Tone option selected, a confirmation tone is given when the feature is successfully activated or deactivated. Overflow tone is given if the operation is unsuccessful. There is no FFC verify code for Electronic Lock.

When a locked phone makes an outgoing trunk call, and Controlled Network Class of Service is defined, the Network Class of Service (NCOS) defined by CNCS is used instead of the NCOS defined in LD 10 or 11 for the phone. If network signaling is configured for the trunk that normally transmits the NCOS of the phone between Electronic Switched Network (ESN) nodes, the CNCS is transmitted instead of the NCOS. This prevents a locked phone from reaching the exchange network by tandeming through a TIE trunk using ESN.

A prompt (PELK) is introduced in the Customer Data Block to implement Electronic Lock on private lines. If this option is enabled, an outgoing call on a private line of a locked phone is subject to the same restrictions as all other DNs on the phone. The same intercept treatment would be given as for a regular DN. The restrictions for private lines, as well as other DN keys on the phone, are controlled by the Controlled Class of Service (CCOS), and by the CNCS if defined. Therefore, for outgoing calls, the Class of Service restrictions and/or New Flexible Code Restriction (NFCR) apply to private line keys on locked phones. Only outgoing calls are affected. The Class of Service of a non-locked phone has no affect on private lines.

Operating parameters

The Network Dialing Plan must be either a Coordinated Dialing Plan (CDP) or a Uniform Dialing Plan (UDP).

The phone password lengths must be equal for all nodes in the network.

Network wide operation is only supported through an MCDN ISDN network.

The Electronic Lock feature must be equipped on both originating and remote nodes.

The FFC used is defined on the node from which Network Electronic Lock is being activated. To activate or deactivate Network Electronic Lock from any remote node, the user has to use the FFC Electronic Lock Activate (ELKA) or Electronic Lock Deactivate (ELKD) code defined on that remote node.

ISDN Basic Rate Interface (BRI) phones cannot be used to lock another phone, nor can they be locked themselves.

An analog (2500/500-type) phone with a private line DN or a digital phone with a private line on the Prime DN cannot be locked.

The following hardware is required:

- Primary Rate Interface – D-channel Handler Interface (DCHI)/ Multipurpose Serial Data Link (MSDL) and PRI2 cards.
- Integrated Services Digital Network Signaling Link (ISL) – DCHI cards and TIE trunks.
- Virtual Network Services (VNS) – DCHI cards and any trunks.

Feature interactions

Automatic Call Distribution (ACD)

An ACD phone cannot be locked.

Call Forward (CFW)

Call Forward No Answer (CFNA)

For Call Forwarding, the COS and NCOS used for the forwarding call can be taken from either the forwarding phone or from the forwarded phone, depending on the option defined in the Customer Data Block.

For example, Phone B call forwards all calls to an external trunk. Phone A calls Phone B. If OPT = CFF in LD 15 (Call Forward forwarded to party's COS and NCOS), the COS and NCOS of Phone B are used for forwarding the call to the trunk. If OPT = CFO (Call Forward originating party's COS and NCOS), the COS and NCOS of Phone A are used for forwarding the call to the trunk.

Direct Inward System Access (DISA)

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

Digital Private Network Signaling System (DPNSS1)

Digital Access Signaling System (DASS2)

Analog Private Network Signaling System (APNSS)

Electronic Lock Network Wide is not supported on DPNSS1, DASS2, or APNSS trunks.

Electronic Switched Network Authcode

If a station user enters an authcode on the phone, regardless of the status of the phone being locked or not, the NCOS defined for the authcode is used. The ESN Authcode feature overrides the Electronic Lock Network Wide feature.

Flexible Numbering Plan

If a network is equipped with a flexible numbering plan (i.e., not all the network DNs are the same length), hanging up before the usual end-of-dialing timeout cancels the request for activation or deactivation of Electronic Lock. Dialing an octothorpe (#) after the network DN will cause the request for activation or deactivation of the Electronic Lock to be sent immediately, instead of waiting for the usual end-of-dialing timeout to send it.

Multiple Appearance DN

The same locked or unlocked state applies to all Terminal Numbers with the same primary DN and the same SCPW. Terminal Numbers with the same DN, but not having the same SCPW, cannot be locked or unlocked.

New Flexible Code Restriction (NFCR)

With NFCR, toll denied stations are allowed or denied calling privileges according to the Facility Restriction Level (FRL) assigned to the NCOS defined in the protected line block. For a locked phone, NFCR uses the FRL assigned to the CNCS to determine its calling privileges if one is defined; if no CNCS is defined, the NCOS of the locked phone will be used.

Scheduled Access Restrictions (SAR)

The SAR feature overrides Electronic Lock.

Virtual Network Services (VNS)

Electronic Lock functions in a VNS environment.

Feature packaging

Electronic Lock Network Wide/Electronic Lock on Private Lines is packaged under Flexible Feature Codes (FFC) package 139, and it has the following dependencies:

- Network Class of Service (NCOS) package 32
- Controlled Class of Service (CCOS) package 81
- Integrated Services Digital Network (ISDN) package 145

Feature implementation

Task summary list

Thefollowing is a summary of the tasks in this section:

1. Table 17: LD 15 - Modify the Customer Data Block for Controlled Class of Service data. on page 42
2. Table 18: LD 15 - Modify the Customer Data Block for Flexible Feature Code data. on page 42
3. Table 19: LD 10 - Set the Station Control Password and Controlled Class of Service Allowed for each analog (500/2500-type) phone. on page 43
4. Table 20: LD 11 - Set the Station Control Password and Controlled Class of Service Allowed for each Meridian 1 proprietary phone. on page 43
5. Table 21: LD 57 - Configure the Flexible Feature Codes for Electronic Lock. on page 44
6. Table 22: LD 87 - In the ESN 2 overlay, define all NCOS to be used. on page 44

Electronic Lock Network Wide

Usethe following overlays to implement Electronic Lock Network Wide.

Table 17: LD 15 - Modify the Customer Data Block for Controlled Class of Service data.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	CCS	Controlled Class of Service data.
CUST	0-99	Customer number as defined in LD 15.
- CCRS	aaa	Controlled Class of Service (CCOS) Restricted Service.
- ECC1	aaa	Enhanced Controlled Class of Service level 1.
- ECC2	aaa	Enhanced Controlled Class of Service level 2.
- CNCS	0-99	Network Controlled Class Of Service for Electronic Lock.
- PELK	YES	Electronic Lock on Private Lines.

Table 18: LD 15 - Modify the Customer Data Block for Flexible Feature Code data.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	FFC	Flexible Feature Code data.
CUST	0-99	Customer number as defined in LD 15.
- CCRS	aaa	Controlled Class of Service (CCOS) Restricted Service.
- SBUP	(YES) NO	Enable use of station control passwords for phone based administration user level access

Prompt	Response	Description
- - PWD2	xxxx	PWD2 password for confirmation
- FFCS	(NO) YES	Change Flexible Feature Code end-of-dialing indicator
- - STRL	1-3	String Length of end-of-dial indicator
- - STRG	xxx	String to indicate end-of-dialing
- ADLD	(0)-20	Auto Dial Delay in seconds
- SCPL	0-8	Station Control Password Length

Table 19: LD 10 - Set the Station Control Password and Controlled Class of Service Allowed for each analog (500/2500-type) phone.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	500	Type of phone.
CUST	0-99	Customer number as defined in LD 15.
TN	l s c u	Terminal Number for Large System and CS 1000E.
SCPW	xxxxxxxx	Station Control Password (1 to 8 digits, as defined in LD 15).
CLS	CCSA	Controlled Class of Service Allowed.

Table 20: LD 11 - Set the Station Control Password and Controlled Class of Service Allowed for each Meridian 1 proprietary phone.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	a...a	Telephone type. Type **?** for a list of possible responses.
CUST	0-99	Customer number as defined in LD 15.
TN	l s c u	Terminal Number for Large System and CS 1000E.
SCPW	xxxxxxxx	Station Control Password (1 to 8 digits, as defined in LD 15).
CLS	CCSA	Controlled Class of Service Allowed.

Table 21: LD 57 - Configure the Flexible Feature Codes for Electronic Lock.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	FFC	Flexible Feature Codes.
CUST	0-99	Customer number as defined in LD 15.
FFCT	YES	Provide FFC confirmation tone.
CODE	ELKA	New/change Electronic Lock Activate FFC.
ELKA	xxxx	Enter the new or changed Electronic Lock Activate FFC
CODE	ELKD	New/change Electronic Lock Deactivate FFC.
ELKD	xxxx	Enter the new or changed Electronic Lock Deactivate FFC

Table 22: LD 87 - In the ESN 2 overlay, define all NCOS to be used.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
CUST	0-99	Customer number as defined in LD 15.
FEAT	NCTL	Network Control.
...		
NRNG	0-99	NCOS range.
NCOS	0-99	Network Class of Service group number.
...		
– FRL	(0)-7	Facility Restriction level.

Electronic Lock on Private Lines

Use the following overlays to implement Electronic Lock Network on Private Lines.

Table 23: LD 15 - Modify the Customer Data Block for Controlled Class of Service data.

Prompt	Response	Description
REQ	CHG	Change existing data.

Prompt	Response	Description
TYPE	CCS	Controlled Class of Service data.
CUST	0-99	Customer number as defined in LD 15.
- CCRS	aaa	Controlled Class of Service (CCOS) Restricted Service.
- ECC1	aaa	Enhanced Controlled Class of Service level 1.
- ECC2	aaa	Enhanced Controlled Class of Service level 2.
- CNCS	0-99	Network Controlled Class Of Service for Electronic Lock.
- PELK	YES	Electronic Lock on Private Lines.

Table 24: LD 15 - Modify the Customer Data Block for Flexible Feature Code data.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	FFC	Flexible Feature Code data.
CUST	0-99	Customer number as defined in LD 15.
- CCRS	aaa	Controlled Class of Service (CCOS) Restricted Service.
- SCPL	0-8	Station Control Password Length.
- SBUP	(YES) NO	Enable use of station control passwords for phone based administration user level access
- - PWD2	xxxx	PWD2 password for confirmation
- FFCS	(NO) YES	Change Flexible Feature Code end-of-dialing indicator
- - STRL	1-3	String Length of end-of-dial indicator
- - STRG	xxx	String to indicate end-of-dialing
- ADLD	(0)-20	Auto Dial Delay in seconds
- SCPL	0-8	Station Control Password Length

Table 25: LD 16 - Set the Facility Restriction Level (FRL) number and the New Flexible Code Restriction (NFCR) tree number for private lines.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	RDB	Route Data Block.
...		

Prompt	Response	Description
PRIV	YES	Private Line Route.
...		
FRL	0-7 0-254	Facility Restriction Level (FRL) and New Flexible Code Restriction (NFCR) tree number.

Table 26: LD 10 - Set the Station Control Password and Controlled Class of Service Allowed for each analog (500/2500-type) phone.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	500	Type of phone.
CUST	0-99	Customer number as defined in LD 15.
TN	l s c u	Terminal Number for Large System and CS 1000E.
SCPW	xxxxxxxx	Station Control Password (1 to 8 digits, as defined in LD 15).
CLS	CCSA	Controlled Class of Service Allowed.

Table 27: LD 11 - Set the Station Control Password and Controlled Class of Service Allowed for each digital proprietary phone.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	a...a	Telephone type. Type **?** for a list of possible responses.
CUST	0-99	Customer number as defined in LD 15.
TN	l s c u	Terminal Number for Large System and CS 1000E.
SCPW	xxxxxxxx	Station Control Password (1 to 8 digits, as defined in LD 15).
CLS	CCSA	Controlled Class of Service Allowed.
...		
KEY	xx PVR yyyy(yyy)	Key number and DN for Private Line Ringing, where xx = the key number and yyyy(yyy) = the Private Line Ringing DN (up to four digits; up to seven digits if the Directory Number Expansion (DNXP) package 150 is equipped).

Prompt	Response	Description
		Private Line Directory Number (PRDN) must be defined in LD 14.

Feature operation

Electronic Lock Network Wide

In this example, an MCDN ISDN network is set up connecting Node 1 to Node 2. Phone A is the Controlling phone. Phone B is the phone to be locked and unlocked by Phone A. The digits that Phone A would dial to ring Phone B are to be in the format of a CDP or UDP dialing plan. This is standard dialing for ISDN features.

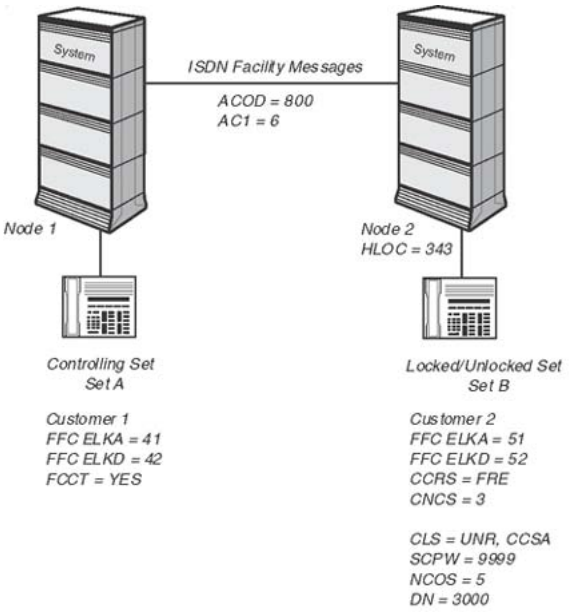


Figure 1: Network Electronic Lock

To lock Phone B (in Node 2) from Phone A (in Node 1), the user goes off-hook and dials the Electronic Lock Activate (ELKA) FFC defined in the Customer Data Block of Node 1, followed by the Station Password (SCPW) defined for Phone B, and the digits that Phone A would normally dial to ring Phone B (e.g., 41 + 9999 + 6-343-3000). If the FFCT option is configured as YES in LD 57 in the Customer Data Block of Node 1, confirmation tone is given to Phone A to confirm that the lock operation has been successful. Phone B becomes locked if it was previously in an unlocked state.

If Phone B is already locked, the above operation is ignored and Phone B remains locked; however, a confirmation tone is provided to verify that the phone is locked.

If the lock operation is unsuccessful, overflow tone is given.

To unlock Phone B (in Node 2) from Phone A (in Node 1), the user goes off-hook and dials the Electronic Lock Deactivate (ELKD) FFC defined in the Customer Data Block of Node 1, followed by the SCPW defined for Phone B and the digits that Phone A would normally dial to ring Phone B (e.g., 42 + 9999+ 6-343-3000). If the FFCT option is configured as YES in LD 57 in the Customer Data Block of Node 1, confirmation tone is given to Phone A to confirm that the unlock operation has been successful. Phone B becomes unlocked if it was previously in a locked state.

If Phone B was already unlocked, the above operation is ignored and Station B remains unlocked; however, a confirmation tone is provided to verify that the phone is unlocked.

If the unlock operation was unsuccessful, overflow tone is given.

Electronic Lock on Private Lines

Feature operation of Electronic Lock for Private Lines is the same as for the basic Electronic Lock feature.

Controlled Network Class of Service

No specific operating procedures are required to use this feature.

Chapter 5: Emergency Services Access

The Emergency Services Access (ESA) feature places a customer in compliance with new federal legislation that requires the Private 911 type of functionality provided by ESA. Please note, however, that the ESA feature is also generally useful for users who are not subject to legislation, and is broad enough to be used in different countries.

For example, it will be appreciated by any customer who wants to route emergency calls in a special manner, or who wants to be notified when a phone user makes an emergency call. It would also appeal to a customer who wishes to have ESA calls answered on-site, on the business premises, rather than being forwarded to the Public Services Answering Point (PSAP).

For complete information on ESA, *Avaya Emergency Services Access Fundamentals, NN43001-613*.

Chapter 6: Equi-distribution Routing Network Attendant Service

Contents

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

Feature description

This feature provides a new algorithm for Network Attendant Service (NAS) routing. The algorithm attempts to distribute routed calls evenly among network attendants, based on loading levels. Calls are routed across a network according to NAS routing tables. These tables define the primary attendant node and alternatives in case the attendant at the primary node is busy and the call waiting threshold has been exceeded. In the case of all attendants busy, as routing is attempted to each alternate defined in the NAS routing table, the Equi-distribution NAS Routing feature reads the load level at each location. The call is then routed to the alternate with the lowest load level.

The load level is a value defined as the customer-defined efficiency factor, multiplied by the number of calls waiting for an attendant, divided by the number of attendants in service. The efficiency factor is a value between 0-8064. The lower the efficiency factor, the more efficient the attendants are considered to be at a specific location (however, if a value of 0 is entered,

then the Equi-distribution NAS Routing feature is not activated, and normal NAS routing occurs).

Another capability of the Equi-distribution NAS Routing feature deals with an attendant at a node going into Night Service. All calls in the attendant queue are routed to another location, based on the NAS routing schedule, instead of to the local night DN.

Operating parameters

This feature applies only to systems using Meridian Customer Defined Network (MCDN) signaling over Integrated Services Digital Network Signaling Link (ISL)/ISDN TIE links.

All nodes in the system network should have the Equi-distribution NAS Routing feature.

This feature is not compatible with the Console Presentation Group (CPG) or Network Tenant Service (NTS) features.

Feature functionality takes into account a maximum number of four alternate locations for each time period.

Feature interactions

When the attendant goes into Night Service, calls presented to the attendant receive NAS routing in an attempt to reach another attendant that is in day service, rather than being routed to the local night DN.

Feature packaging

The following packages are required for Equi-distribution Routing and Network Attendant Service:

- Network Alternate Route Selection (NARS) package 58
- Integrated Services Digital Network Signaling (ISDN) package 145, or ISDN Signaling Link (ISL) package 147
- Network Attendant Service (NAS) package 159

Feature implementation

Task summary list

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

1. Table 28: LD15 - Configure the EFLL. on page 53
2. Table 29: LD86 - Define the attendant alternatives for NAS routing in the ElectronicSwitched Network. on page 53
3. Table 30: LD 93 - Configure the EFLL. on page 54

Table 28: LD15 - Configure the EFLL.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	ATT	Attendant Console options.
CUST	0-99	Customer number as defined in LD 15.
...		
EFLL	(0)-8064	Efficiency Factor Loading Level. The efficiency factor loading level. If the default value of 0 is entered, then the Equi-distribution NAS Routing feature is not activated, and normal NAS routing occurs.

Table 29: LD86 - Define the attendant alternatives for NAS routing in the ElectronicSwitched Network.

Prompt	Response	Description
REQ	CHG	Change.
CUST	0-99	Customer number as defined in LD 15.
FEAT	NAS	Network Attendant Service
...		
ALST	1-7	Alternatives List. Attendant alternatives for NAS routing (up to 4 alternatives are defined for each schedule period).
DBK	(N) Y	Drop Back busy option. YES = Drop Back Busy allowed, NO = Drop Back Busy denied.

Table 30: LD 93 - Configure the EFLL.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	aaa	Type of data block.
CUST	0-99	Customer number as defined in LD 15.
...		
EFLL	(0)-8064	Efficiency Factor Loading Level. The efficiency factor loading level. If the default value of 0 is entered, then the Equi-distribution NAS Routing feature is not activated, and normal NAS routing occurs.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 7: Error Handling on ISDN Signaling Link Analog E and M TIE Trunks

Contents

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

Feature description

The Integrated Services Digital Network Signaling Link (ISL) Analog E&M TIE Trunks feature identifies error conditions that are generated by external equipment and ensures that faulty trunks cannot be used. When the error condition is removed, the trunk is put back into service.

When an error condition is identified while the trunk is idle, the trunk is put into a maintenance busy mode and a trunk error message is printed. If an error occurs while a call is being sent, an error message is printed and the call proceeds; the trunk unit is put into maintenance busy mode after the call is disconnected. Outgoing calls cannot be made on units that are in maintenance busy and incoming calls are rejected.

Once the external equipment removes the error condition, a message is printed and a restart message is sent on the D-channel to indicate that the unit is usable.

Operating parameters

Error Handling on ISL Analog E&M TIE Trunks only functions on ISL Analog E&M TIE Trunks and not on other types of ISL TIE trunks. This function only operates when the D-channel is established.

E&M types applicable to this feature include two-wire and four-wire with TY1, TY2 and British Post Office E&M signaling types. The feature applies to immediate start, delay dial start and wink start arrangement types.

This feature is designed to operate only when ISL E&M TIE trunks are used in conjunction with transmission equipment.

To implement Error Handling on ISL Analog E&M TIE trunks the FALT prompt response in the Route Data Block (LD 16) must be YES. If FALT = NO, then no error handling will occur on the route.

Error handling exists on ISDN Signaling Link Digital Trunk Interface (2Mbit).

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

The following package is required for Error Handling on ISL Analog E&M TIE Trunks:

- ISDN Signaling Link (ISL) package 147, which is dependent on ISDN Signaling (ISDN) package 145.

Feature implementation

Table 31: LD 16 - Configure ISL on Analog Trunks.

Prompt	Response	Description
REQ	NEW	Add new data.

Prompt	Response	Description
	CHG	Change existing data.
TYPE	RDB	Route data block.
...		
DTRK	NO	Digital trunk route.
ISDN	YES	ISDN Primary Rate Interface option.
MODE	ISLD	Mode of D-channel controlling the route.
...		
FALT	YES	Enables the error handling and recognition of E lead error for ISL E&M TIE Trunks. If FALT= NO, then no error handling with occur.

Feature operation

Overlay Changes

The enabling/disabling commands in the maintenance overlay is changed so that if the trunk unit is in an error state the following occurs:

- The trunk unit cannot be enabled until the external error condition is removed. The message TRK533 is printed to warn the user that the enable command is cancelled because an error exists on the ISL E&M TIE Trunks. When this occurs, the trunk unit will remain in maintenance busy mode.

These overlay changes ensure that error states on both sides of the link are not affected by enabling/disabling the unit.

Error Applied to Unit

- An error condition can be generated by the external equipment when a unit is idle, during call setup or when a call is established.

Error applied, Unit Idle

- If an error is detected the unit is put in maintenance busy mode and the error message TRK531 is printed.

Error applied, Call Setup

- If an error is detected during call setup, the error message TRK531 is printed.
- After the call is disconnected, the trunk is put into maintenance busy mode.

Error applied, Call Established

When an error is applied while a call is established, it is detected after the call is disconnected. If the release message is not received within the time defined by the supervision timer then the E lead remains grounded and causes an error condition. The following occurs:

- The unit is put into maintenance busy mode and the error message TRK531 is printed.

ErrorState

When the unit is an error state then it is not available for incoming or outgoing calls. The error state is maintained until it is cleared by external equipment.

Outgoing Calls

Normal call processing selects the next available channel for outgoing calls.

Incoming Calls

Far end attempts to make calls will send a SETUP message. If the unit is in an error state, a Release Complete message is sent. The far end trunk will be put into a LOCKOUT state.

Restart Message

During an error state, a RESTART message is received from the far end. This message is ignored so that the error state of the unit is preserved.

Error Removed

Ifthe error condition is removed by external equipment, the trunk ismarked IDLE, the error message TRK532 is printed and a RESTART OneChannel message is sent on the D-channel. The far end trunk unit changesfrom LOCKOUT to IDLE state.

Nospecific operating procedures are required to use this feature, howeverthe following system messages will occur throughout feature operation.

Table 32: System messages

Message Number	Events
TRK531 tn	An error has been detected on ISL E&M TIE, unit will be put to Maintenance Busy.
TRK532 tn	Error Condition on ISL E&M TIE removed.
TRK533 tn	Enable command is cancelled on this unit because an error exists on the ISL E&M TIE trunks.

Chapter 8: ESN compatibility on ISDN

Contents

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

Feature description

TheElectronic Switched Network (ESN) group of features is designed tosupport voice and circuit-switched voiceband data telecommunicationneeds for multiple-location customer applications.

Note:

ESN is supported for connections from system to system, to SL-100, and to DMS-100 Centrex.

Electronic Switched Network (ESN) applications range from a single network node (combined PBX and network switching system) to a widely dispersed network with up to 256 locations. For complete information on Electronic Switched Network (ESN), refer to *Avaya Electronic Switched Network: Signaling and Transmission Guidelines, NN43001-280*.

ISDNenhances Electronic Switched Network (ESN), by allowing ESN capabilitiessuch as Network Class of Service, Network Alternate Route Selection,ESN queuing, and Network Authorization Codes across a system ISDNnetwork.

ISDNprovides a series of messages containing imbedded information; forexample, NCOS information is imbedded in the SETUP message to providethe means to control a user's eligibility to:

- access routes
- access queuing
- receive Expensive Route Warning Tone (ERWT)
- access network speed call

Table 33: ESN features available with ISDN on page 60 provides a list of ESN features and capabilities that are configured on ISDN.

Table 33: ESN features available with ISDN

ESN features	Available on ISDN?
Network Class of Service	yes
Network Control	yes
—NCOS	yes
—TCOS (ETN)	yes
Network Alternate Route Selection	
—NARS Access Codes	yes
—Uniform Dialing Plan	yes
—Time of Day Routing	yes
—Network Routing Control	yes
—Satellite Link Control	yes
—Digit Screening	yes
—Digit Manipulationyes	yes
—Auto on-net to offnet overflow	yes
—Automatic Least Cost Routing	yes
—Network Speed Call	yes
—Automatic OCC Access	yes
ESN Queuing	
Local queuing	
—Offhook queuing	yes
—Call-back queuing	yes
Network Queuing	
—Coordinated Call-Back Queuing	no

ESN features	Available on ISDN?
—Coordinated Call Back Queuing to conventional main	no
—Remote Virtual Queuing	
	yes
Network Authorization Codes	yes
Coordinated Dialing Plan for 10 digits	yes
Flexible Numbering Plan	yes
ESN Network Call Transfer	no

Operating parameters

There are no operating parameters associated with this feature

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is included in base System Software.

Feature implementation

There are no specific implementation procedures for this feature.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 9: ESN interworking

Contents

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

Feature description

The Electronic Switched Network (ESN) operates within ISDN, supporting Network Class of Service (NCOS) capability. The tables in this module describe which ESN features are available over ISDN and illustrate ESN capabilities over ISDN.

Network Class Of Service (NCOS) information is embedded in SETUP messages, and provides the means to control a user's eligibility for the following:

- access routes
- access queuing

- receive Expensive Route Warning Tone (ERWT)
- access network speed call

When NCOS information, which includes Traveling Class of Service (TCOS) information, is sent to a DMS-100, the DMS switch can provide access to AT&Ts Electronic Tandem Network (ETN).

ESN translation

Currently, a system switch with a BARS or NARS package can have two (or one) separate ESN translators for handling BARS/NARS calls. NARS can have two translators and BARS can have only one translator (AC1). This is in addition to its standard translator, which handles all other call types. Each NARS/BARS translator has its access code (ESN AC1/AC2), which is defined on the standard translator. The remaining digits, for example, NPA, NXX, LOC, and SPN, are defined on the NARS/BARS translator.

Therefore, a receiving switch must determine which translator to use and insert the needed NARS/BARS access code (unless the receiving number is complete, that is, it includes the proper NARS/BARS access code in its digit stream already). This can be done by configuring an ESN digit manipulation to insert the ESN access code in the sending switch.

To insert the needed BARS/NARS access code in the receiving switch, two route options in LD 16 can be used:

- INST: An incoming route option that inserts the specified digits to the incoming digit stream for all the calls received on that route.
- INAC: An incoming ISDN route option that inserts the needed NARS/BARS access code to the incoming digit stream. The insertion is based on the Type of Number of the received Called Number Information Element, bypassing the digit insertion (INST) of the route. The AC2 option in LD 15 provides a mapping between the incoming Type of Number and the desired translator. This is used to determine which NARS/BARS access code is inserted.

ESN access code configuration with package 148

For system configurations that utilize ISDN networking features, ESN access code insertion must be performed on the receiving system. The originating system must not use digit manipulation to insert and send the ESN access code to the terminating system.

Therefore, in order to accomplish networking feature transparency (requiring package 148), the ESN access code insertion must be performed in the receiving system, by means of the INAC prompt in LD 16. Although the INST or the INAC prompt can be used, the INAC prompt is the recommended method.

When the PBX is upgraded, the above database conversion is required to allow networking applications (such as NRAG) to function properly.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature is included in base System Software.

Feature implementation

Tasksummary list

Thefollowing is a summary of the tasks in this section:

1. Table 35: LD 86 - Configure digit manipulation. on page 66
2. Table 36: LD 16 - Enable the INAC option. on page 67
3. Table 37: LD 15 - Map the NARS/BARS access code to the incoming call types. on page 67

ESNdefault call types

When an ISDN call is made, a default Electronic Switched Network (ESN) call type is set initially. The call type is set depending on the dialing method used (that is, how the call is made). Table 34: Default Call Types on page 66shows the ISDN default call types. LD 86 describes how to define the default call types.

Note:

When a trunk access code is used to dial on an ISDN route, the caller dials all digits without waiting for subsequent dial tones.

Table 34: Default Call Types

Dial method	Default call type	Comment
Direct dial	UKWN (unknown)	Direct dial to trunks through access code
CDP (DSC/ TSC)	CDP (Coordinated dialing plan)	Dial CDP number through DSC/TSC code
CDP (LSC)	UKWN	Dial CDP number through LSC code
ESN (LOC)	LOC (location code)	Dial NARS location code
ESN (HLOC)	UKWN	Dial ESN HLOC number
ESN (NPA)	NPA (national)	Dial ESN NPA number
ESN (NXX)	NXX (local)	Dial ESN NXX number
ESN (SPN)	SPN (special number)	Dial ESN SPN number

Table 35: LD 86 - Configure digit manipulation.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
CUST	0-99	Customer number as defined in LD 15.
FEAT	DGT	Digit manipulation data block
DMI	1–255	Digit manipulation table index numbers for NARS/BARS
CTYP		Call type. Enter the call type to the manipulated digits. This call type must be recognized by the far end switch.
	(NCHG)	The call type will not be changed
	INTL	Special number in international format
	NPA	NPA
	NXX	NXX
	LOC	Location Code
	CDP	Coordinated Dialing Plan
	SPN	Special number
	UKWN	Unknown call type

ESNaccess code insertion

LD 16 and LD 15, starting on Table 36: LD 16 - Enable the INAC option. on page 67, describe how to insert the needed NARS/BARS access code to the incoming digit stream. This is required for ISDN networking features such as Network Ring Again (NRAG) and Network Message Services (NMS).

ESNaccess code insertion is performed in the receiving switch eitherby the INST or INAC option in LD 16. The INAC option is recommended.The called number can go through ESN digit manipulation, which mustnot include the ESN access code in the sending switch.

Table 36: LD 16 - Enable the INAC option.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	RDB	Route data block
INAC	(NO) YES	Insert Access Code. Permit an ESN access code to be automatically added to an incoming ESN call from a private network. If INAC is YES, the digit insertion option (INST) is bypassed. This prompt only appears if the route type is a tie trunk.

Table 37: LD 15 - Map the NARS/BARS access code to the incoming call types.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	NET	Networking data.
AC2		Access code 2. Enter call types (type of number) that use access code 2. Multiple responses are permitted. If a call type is not entered here, it is automatically defaulted to access code 1.
	NPA	E.164 National
	NXX	E.164 Subscriber
	INTL	International
	SPN	Special number
	LOC	Location code
ISDN	YES	

Prompt	Response	Description
PNI	1–32700	Customer private identifier—unique to a customer. Within one network, use the same value for PNI in both the Customer Data Block (LD 15) and the Route Data Block (LD 16). For different customers, the PNI in LD 15 should match the PNI in LD 16 at the far end switch (M1/SL-1 only).

Private Network Hopoff

Private Network Hopoff is provided by the ESN feature. Private Network Hopoff allows ISDN callers to use the private network to complete public calls. The call is routed through the private network as far as possible before "hopping off" onto the public lines (see Figure 2: Private Network Hopoff on page 68).

Users select this service, on a single call basis, by dialing the private network facilities prefix digit, followed by the public network number. For example, the number 9 is often used as a private network facilities prefix digit before dialing a local number.

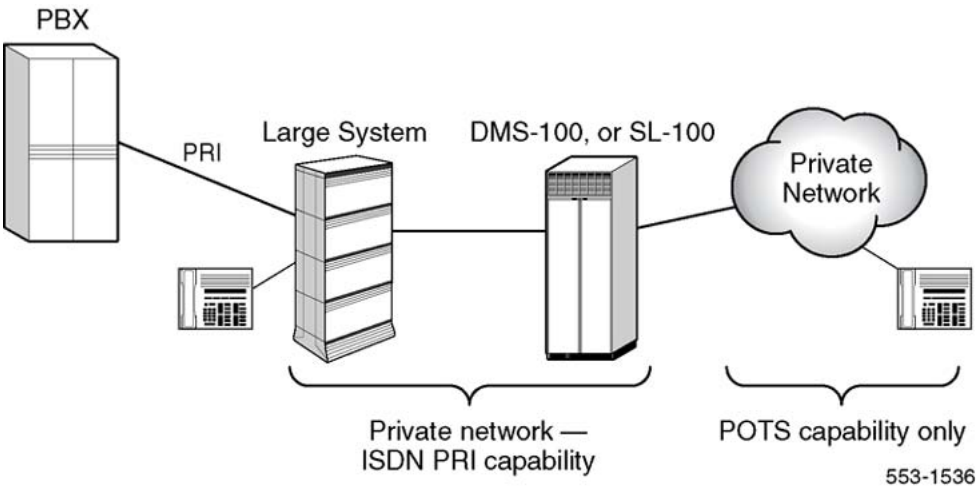


Figure 2: Private Network Hopoff

Private Network Overflow

This service allows overflow from a private network onto public network facilities (see Figure 3: Private Network Overflow on page 69). On outgoing calls over a DMS-100 PRI connection, the DMS provides network overflow when all of a customer's private trunks are busy.

For outgoing calls that encounter blocking (all B-channels are busy), the network provides the capability to overflow to alternate trunks. This is provided by the Electronic Switched Network (ESN) feature. This access can be restricted using a trunk or set class of service.

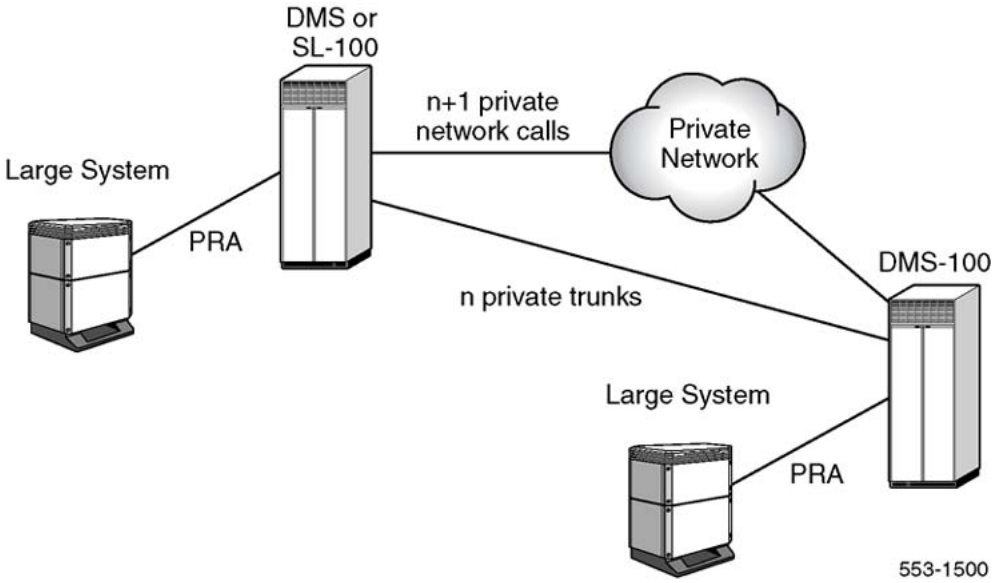


Figure 3: Private Network Overflow

Feature operation

No specific operating procedures are required to use this feature.

Chapter 10: ESN Location Code Expansion

Contents

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

Feature description

The ESN (Electronic Switched Network) Location Code Expansion feature expands the number of Location Codes that can be configured on CS 1000and Meridian 1systems. Previously, ESN supported a maximum of 999 Location Codes. With the ESN Location Code Expansion feature, ESN supports a maximum of 16 000 Location Codes. This expansion is required to support large multi-site customers with more than 1000 sites that use the Location Code Dialing Plan.

ESN Location Code Expansion enhances the printing of translation codes such as LOC, NPA, NXX, and SPN. Previously, the system printed only exact matches of these codes. With ESN Location Code Expansion, the system also prints partial matches of translation codes.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

The ESN Location Code Expansion feature requires the following packages:

- Flexible Numbering Plan (FNP) package 160
- Location Code Expansion (LOCX) package 400

Feature implementation

Task summary list

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

1. Table 38: LD 86 - Define the maximum number of Location Codes that can be configured on the system. on page 72
2. Table 39: LD 90 - Configure the Location Codes. on page 73

⚠ Caution:

Service Interruption

Carefully choose the MXLC value based on the required number of LOCs in the network. If you choose a number greater than the number of LOCs needed for configuration in LD 90, then valuable system memory is wasted.

Table 38: LD 86 - Define the maximum number of Location Codes that can be configured on the system.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
CUST	0-99	Customer number as defined in LD 15.
FEAT	ESN	Electronic Switched Network
MXLC	0-999 0-16000	Maximum number of LOC codes Maximum number of LOC codes (with the ESN Location Code Expansion feature and the FNP feature enabled)

Table 39: LD 90 - Configure the Location Codes.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
CUST	0-99	Customer number as defined in LD 15.
FEAT	NET	Network translation table
TRANS	AC1 AC2 SUM	Access Code 1 Access Code 2 Summary tables
TYPE	LOC	Location Code
LOC	xxx xxx y..y	Location Code Extended Location code

Feature operation

No specific operating procedures are required to use this feature.

Chapter 11: EuroISDN 7kHz/ Videotelephony Teleservices

Contents

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

Feature description

The 7 kHz/Videotelephony teleservices feature supports the 7 kHz teleservice, and the Videotelephony teleservice.

The 7 kHz teleservice is an enhanced telephony teleservice offering better voice quality. It is a real-time teleservice in which a voice call uses one circuit-mode 64 Kbit/s connection with 7 kHz bandwidth.

The Videotelephony teleservice is a real-time, audiovisual teleservice in which voice and moving pictures are interchanged by means of one or two 64 Kbit/s circuit-mode connections.

The 7 kHz/Videotelephony teleservices are supported over EuroISDN PRI2 and Basic Rate Trunk interfaces, and on ETSI Basic Rate Interface sets.

This feature also supports the fall-back capability. Fall-back allows a request for the telephony 7 kHz teleservice or for the videotelephony teleservice, to use the alternate teleservice. In the case of the 7 kHz teleservice, the alternative is the 3.1 kHz teleservice. In the case of the

Videotelephony teleservice, the alternative is the 7 kHz teleservice (if supported), or the 3.1 kHz teleservice.

This feature supports the following types of communication:

- videotelephone terminal to videotelephone terminal
- 7 kHz terminal to 7 kHz terminal
- videotelephone terminal to 7 kHz terminal or 3.1 kHz terminal, if fall-back allowed
- 7 kHz terminal to 3.1 kHz terminal, if fall-back allowed

Originating and terminating terminals can be local or remote, but calls using one of these teleservices with at least one remote terminal must be made over a EuroISDN PRI2 or EuroISDN BRI link.

When a user originates a call requesting a 7 kHz or videotelephony teleservice, the user can choose if fall-back is allowed or not, that is, whether or not the call is to be established with the alternative teleservice if the requested teleservice is not available. Such requests can be made either from a local ETSI BRI phone supporting the teleservice, or from a remote phone whose request is received over a EuroISDN link. The terminating party can accept the call, if the teleservice is supported end to end, or reject the call, if the requested teleservice is not supported.

Note:

A videotelephone terminal does not support the telephony 7 kHz teleservice.

7 kHz or videotelephony teleservice request with fall-back not allowed

If fall-back is not allowed, the call is established with the requested teleservice or rejected.

If the call request reaches a non EuroISDN link, or if the requested phone is a non-ETSI BRI phone, the call is rejected. For analog (500/2500-type) phones and Meridian Proprietary Phones, the call is rejected. The call is also rejected for calls over a EuroISDN link and for ETSI BRI phones not configured to support the 7 kHz/Videotelephony teleservices (the 7VOD or 7DTA teleservice EuroISDN for routes, or the VID7 supplementary service for ETSI BRI phones).

7 kHz or videotelephony teleservice request with fall-back allowed

If fall-back is allowed, the call is established with the preferred teleservice or with the alternate teleservice.

If the call request reaches a non EuroISDN link, or if the destination phone does not support the requested teleservice, the call is established with the alternate teleservice (3.1 kHz teleservice). The originator is notified that fall-back has occurred.

Establishment of a second connection for videotelephony calls

When a call is established with the videotelephony teleservice, a second connection can be established to provide a better video quality. Only the calling party can initiate a request for a second connection, and this only after the first connection is established.

Operating parameters

This feature is not supported over QSIG, DPNSS1, or MCDN networks. If a EuroISDN call gets routed over a QSIG, DPNSS1, or MCDN link, the call is not established with 7 kHz or videotelephony teleservice. If fall-back is allowed, call is established with the alternate (3.1 kHz) teleservice, or else the call is dropped.

This feature requires that the EuroISDN ETS 300-403 compliance update feature be implemented (IFC = E403 in LD 16).

Feature interactions

Call Forward All Calls

Fall-back is applied if Call Forward All Calls was activated. If fall-back is not allowed, the call is treated as a normal operation.

If fall-back occurred, a call request for the alternate (3.1 kHz) teleservice is sent to the forwarded-to phone or trunk. If fall-back has not already occurred, the call is treated as a normal operation for the forwarded-to phone or trunk.

Call forward Busy

If Call Forward Busy feature has been activated on a busy phone, the call is established with the alternate (3.1 kHz) teleservice, if fall-back is allowed. If fall-back is not allowed, the call is rejected.

Call Forward No Answer

When Call Forward No Answer has been activated on a phone, the call is treated as a normal operation for the forwarded-to phone or trunk, if fall-back is not allowed.

If fall-back is allowed and has already occurred, a call request for the alternate (3.1 kHz) teleservice is sent to the forwarded-to phone or trunk. If fall-back is allowed and has not already occurred, the call is treated as a normal operation for the forwarded-to phone or trunk.

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 all public Integrated Services Digital Network (ISDN) interfaces. When the Privacy Indicator Ignore (PII) prompt is YES in LD 16, the Calling Line Identification (CLID) Presentation Indicator and the Calling Party Name Display (CPND) Indicator override the blocking of the number and name details of the calling party. The name and number of the calling party appear on the receiving telephone.

The CPPE feature introduces a new route option AUXP for Auxiliary processor applications in LD 16 (RDB – Route Data Block). AUXP enhances the ability of the system to honor or ignore the Privacy Indicator for a Calling Party Privacy call, for each incoming route. If AUXP is 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, there will be no change to the CLID Presentation Indicator.

Hunt

If Hunt has been applied to a busy destination, destination phone is busy, and if the Hunt feature is configured, the call is treated as a normal operation for the hunted-to phone or trunk.

DISA call

For a DISA call requesting 7 kHz or videotelephony teleservice, fall-back occurs, if allowed, or the call is rejected.

ACD DN

For a request for 7 kHz or videotelephony teleservice terminating on an ACD DN, fall-back occurs, if allowed, or the call is rejected.

Advice Of Charge

If a second connection is established for a videotelephony teleservice, the charge information is received separately for the two connections, since there is no way to identify that the two connections relate to the same videotelephony call.

Multiple Appearance Directory Number

An incoming call to a Multiple Appearance DN associated to several (non-BRI) phones is presented to all phones. For an incoming call to a Multiple Appearance DN requesting the 7 kHz or videotelephony teleservice, fall-back occurs if allowed, or the call is rejected.

Network Alternate Route Selection

NARS route selection is done by looking at the Data Selection (DSEL) values of the different routes. An outgoing call requesting the 7 kHz or videotelephony teleservice can only use route defined with 7DTA or 7VOD at DSEL prompt, whether fall-back is allowed or not.

If fall-back is allowed and if no route has been found for the preferred teleservice, then fall-back occurs and a new route selection is started for the alternate teleservice.

If fall-back occurs further in the network, a voice call can be established using a route defined for data (DSEL = 7DTA).

Customer Controlled Routing

For a call request for the 7 kHz or videotelephony teleservice terminating on a CCR Control DN (same as an ACD DN), fall-back occurs if allowed, or else the call is rejected.

Symposium Call Center Server

For a call request for the 7 kHz or videotelephony teleservice terminating on a Symposium Call Center Server, fall-back occurs if allowed, or else the call is rejected.

Meridian IVR

For a call request for the 7 kHz or videotelephony teleservice terminating on a Meridian IVR application, fall-back occurs if allowed, or else the call is rejected.

Meridian Link

For a call request for the 7 kHz or videotelephony teleservice terminating on a Meridian link application, fall-back occurs if allowed, or else the call is rejected.

Feature packaging

This feature requires the following packages:

For ISDN PRI2:

- Integrated Services Digital Network (ISDN) package 145
- 2.0 Mbit/s Primary Rate Interface (PRI2) package 154
- ISDN Supplementary Features (ISDN_SUPP) package 161
- International Primary Rate Access (IPRA) package 202
- Multi-Purpose Serial Data Link (MSDL) package 222
- EuroISDN (EURO) package 261
- Calling Party Privacy (CPP) package 301

For ISDN Basic Rate trunking:

- Integrated Services Digital Network (ISDN) package 145
- ISDN Supplementary Features (ISDN_SUPP) package 161
- International Primary Rate Access (IPRA) package 202
- Basic Rate Interface (BRI) package 216
- Basic Rate Interface Trunking (BRIT) package 233
- EuroISDN (EURO) package 261

For ISDN Basic Rate line application:

- Basic Rate Interface Line (BRIL) package 235

Feature implementation

Tasksummary list

Thefollowing is a summary of the tasks in this section:

1. Table 40: LD17 - Configure D-Channel for the ETS 300 403 interface. on page 81
2. Table 41: LD 16 - Configure the 7 kHz/Videotelephony teleservices for a PRI2 EuroISDN E403 route. on page 82
3. Table 42: LD 14 - Configure the 7 kHz/Videotelephony teleservices PRI2 EuroISDN E403 trunks. on page 83
4. Table 43: LD 16 - Configure the 7 kHz/Videotelephony teleservices for an ISDN BRI EuroISDN E403 route, in the Route Data Block. on page 83
5. Table 44: LD 27 - Configure the ETSI ISDN BRI phones, on the Digital Subscriber Loop (DSL), for voice and data. on page 85
6. Table 45: LD 27 - Configure the Terminal Service Profile (TSP) for voice and data, and the 7 kHz/Videotelephony teleservices on the ETSI ISDN BRI phones. on page 85

ISDNPRI2 implementation

Table 40: LD17 - Configure D-Channel for the ETS 300 403 interface.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	ADAN	Action Device and Number.
- ADAN	CHG DCH xx	Change existing D-channel data.
...		
- IFC	E403	EuroISDN Interface conforming to the ETS 300 403 or ETS 300 102 protocol.
- - CNTY	xxxx	Enter country pertaining to the E403 interface.

Note:

The EuroISDN route using the 7 kHz/Videotelephony teleservices must be configured as using the interface type of ETS 300 403 (IFC = E403), even if the link to the Central Office is still based on ETS 300 102 implementation.

Table 41: LD 16 - Configure the 7 kHz/Videotelephony teleservices for a PRI2 EuroISDN E403 route.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	RDB	Route Data Block.
CUST	0-99	Customer number as defined in LD 15.
ROUT	0-511	Route number for Large System and CS 1000E system.
TKTP	aaaa	Trunk type.
...		
DTRK	YES	EuroISDN route is digital.
- DGTP		Digital trunk type.
	PRI2	PRI2 = 2.0 Mbit/s Primary Rate Interface.
- IFC	E403	EuroISDN Interface conforming to the ETS 300 403, or the extended ETS 300 102 version for country-specific interfaces. Refer to the Note above.
- - CNTY		Enter country pertaining to the E403 interface. If either ETSI or NET are entered, the interface is fully compliant with ETS 300 403. The entered value must match the entry in LD 17.
	ETSI NET	ETS 300 403 for the user side. ETS 300 403 for the network side. If any of the countries listed on the following page are entered, the interface functions with the extended ETS 300 102 capabilities.
	AUS DEN FIN GER ITA NOR POR SWE EIR DUT SWI BEL	Austria. Denmark. Finland. Germany. Italy. Norway. Portugal. Sweden. Ireland. Holland. Switzerland. Belgium.

Prompt	Response	Description
	ESP UK FRA CIS	Spain. United Kingdom. France. Commonwealth of Independent States (Russia and the Ukraine).
...		
- DSEL		Data selection.
	7VOD 7DTA	The route supports voice and data calls, and the telephony 7 kHz/Videotelephony teleservices. The route supports data calls, and the telephony 7 kHz/ Videotelephony teleservices.

Table 42: LD 14 - Configure the 7 kHz/Videotelephony teleservices PRI2 EuroISDN E403 trunks.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	aaaa	Trunk type. Must match the trunk type configured in LD 16.
CUST	0-99	Customer number as defined in LD 15
RTMB		Route number and Member Number
	0-511 1-4000	Range for Large System and CS 1000E system.

ISDNBasic Rate Interface implementation

Note:

The EuroISDN route using the 7 kHz/Videotelephony teleservices must be configured as using the interface type of ETS 300 403 (IFC = E403), even if the link to the Central Office is still be based on ETS 300 102 implementation.

Table 43: LD 16 - Configure the 7 kHz/Videotelephony teleservices for an ISDN BRI EuroISDN E403 route, in the Route Data Block.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	RDB	Route Data Block.
CUST	0-99	Customer number as defined in LD 15.

Prompt	Response	Description
ROUT	0-511	Route number for Large System and CS 1000E system.
TKTP	aaaa	Trunk type.
DTRK	YES	EuroISDN route is digital.
- DGTP	BRI	Digital trunk type is Basic Rate Interface.
- IFC	E403	EuroISDN Interface conforming to the ETS 300 403, or the extended ETS 300 102 version for country-specific interfaces. Refer to the Note above.
- - CNTY		Enter country pertaining to the E403 interface. If either ETSI or NET are entered, the interface is fully compliant with ETS 300 403. The entered value must match the entry in LD 17.
	ETSI NET	ETS 300 403 for the user side. ETS 300 403 for the network side. If any of the countries listed on the following page are entered, the interface functions with the extended ETS 300 102 capabilities.
	AUS DEN FIN GER ITA NOR POR SWE EIR DUT SWI BEL ESP UK FRA CIS	Austria. Denmark. Finland. Germany. Italy. Norway. Portugal. Sweden. Ireland. Holland. Switzerland. Belgium. Spain. United Kingdom. France. Commonwealth of Independent States (Russia and the Ukraine).
...		
- DSEL		Data selection.
	7VOD 7DTA	The route supports voice and data calls, and the telephony 7 kHz/Videotelephony teleservices. The route supports data calls, and the telephony 7 kHz/ Videotelephony teleservices.

Table 44: LD 27 - Configure the ETSI ISDN BRI phones, on the Digital Subscriber Loop (DSL), for voice and data.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	DSL	Digital Subscriber Loop data block.
DSL		Digital Subscriber Loop
	l s c dsl#	DSL location, where: • lll (superloop) = 0-156 (must be zero or a number divisible by 4) • s (shelf) = 0-1 • cc (card) = 0-15 and for CS 1000E 1–20 • dsl# (DSL location) = 0-7 Assign 31 DSLs for each MISP if DCH or BDCH were specified at the PH prompt in "MISP configuration procedures".
CUST	0-99	Customer number as defined in LD 15.
CTYP	SILC UILC	Interface card type is SILC or UILC.
APPL	BRIL	Basic Rate Interface.
MISP	0-158 0-9	MISP card slot number has to be 1-4, 7-10 for Media Gateway 1000 1-10 for Media Gateway 1010
MODE	(TE) NT	Mode, user side (TE) or network side (NT).
B1CT	VCE DTA	B-Channel 1 call type is voice and data. **Note:** At least one of the B-Channels must be configured for voice and data.
B2CT	VCE DTA	B-Channel 2 call type is voice and data. **Note:** At least one of the B-Channels must be configured for voice and data.

Table 45: LD 27 - Configure the Terminal Service Profile (TSP) for voice and data, and the 7 kHz/Videotelephony teleservices on the ETSI ISDN BRI phones.

Prompt	Response	Description
REQ	NEW	Add new data.

Prompt	Response	Description
	CHG	Change existing data.
TYPE	TSP	Terminal Service Profile data block.
DSL		Digital Subscriber Loop
	l s c dsl#	DSL location, where: • lll (superloop) = 0-156 (must be zero or a number divisible by 4) • s (shelf) = 0-1 • cc (card) = 0-15 and for CS 1000E 1–20 • dsl# (DSL location) = 0-7
...		
BCH	1 2	B-Channel (either 1 or 2) to which the TSP is associated.
...		
- CT	VCE DTA	The call type that the TSP supports is voice and data.
...		
SSRV_ETSI	VID7 XVID7	The ETSI ISDN BRI phone supports the 7kHz/ Videotelephony teleservices. Precede with an X to remove the configured 7kHz/ Videotelephony teleservices.

Maintenance and diagnostics guidelines

Call Trace (LD 80)

LD 80 is used to gather information pertaining to a call. If the call originated or terminated over an ISDN trunk, the following information is printed, in addition to the basic call information:

- call reference number
- bearer capability
- call state
- calling number
- called number
- Virtual Network Services Directory Number, if applicable

New bearer capability print information

The new printed bearer capability information, which is printed for the 7kHz/Videotelephony teleservices feature, is 'UDI-TA', which pertains to the 'Unrestricted Digital Information with Tones/Announcement' Information Transfer Capability Information Element (IE).

Bearer capability print format when fall-back is not allowed

If fall-back is not allowed, there is only one bearer capability Information Element in the received SETUP message. The print format is not changed. The format is 'BEARER CAP = aaaaaa', where 'aaaaaa' = one of 'VOICE, 64KCLR, 64KRES, 56KRES, 3.1KHZ, DATAUN, UDI-TA, or DATARS'.

Bearer capability print format when fall-back is allowed

If fall-back is allowed, there are two bearer capability IE contained in the received SETUP message. There are two print formats for the bearer capability.

When the used teleservice has not yet been selected, both values are printed in the order that they are received. When the used teleservice has been selected (and whether fall-back occurs or not), only the selected value is printed.

Examples follow:

BEARER CAP = VOICE UDI-TA (if a bearer capability has yet to be selected)

BEARER CAP = UDI-TA (if fall-back has not occurred)

BEARER CAP = VOICE (if fall-back has occurred)

High Layer Compatibility print information

If a bearer capability 'UDI-TA' IE is received, then the High Layer Compatibility (HLC) information is printed on the line below the bearer capability information, whatever the value of the bearer capability.

The HLC print format is 'HLC = aaaaa', where aaaaa = VIDEO or TELE.

For other values, nothing is printed, since these values do not define a teleservice when the bearer capability receives a UDI-TA IE. If two HLC IEs are received, in the case of fall-back, the print format is the same as for the bearer capability print format.

No correlation between two connections for the same call

There is no messaging information within a EuroISDN network to correlate two connections established for the same call. The only way to gather information pertaining to two connections for the same call, is to analyze the calling and called number information associated with each connection.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 12: EuroISDN Continuation

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

EuroISDN provides Primary and Basic Rate Interfaces to Central Offices/PublicExchanges that comply with the specifications for Layer 3, as describedin ETS 300-102. The EuroISDN Continuation feature expands the scopeof the original EuroISDN feature to include application support forSwitzerland, Spain, Belgium and the United Kingdom.

In addition, the following functionalities are provided by the EuroISDN Continuation feature:

- intercept treatment upon reception of an invalid or incomplete called party number
- interception to an attendant for EuroISDN voice calls terminating on a data device
- the capability to listen to tones and announcements provided by the Central Office on call clearing (T306 supported)
- the capability of transferring outgoing EuroISDN calls after completion of dialing (only for Italy)
- Calling Line Identification and Connected Line Identification transparency to or from EuroISDN to or from a Basic Rate Interface (BRI) phone
- the configuration of Connected Line Presentation (COLP) on each D-channel basis (remote capability)
- 3.1 KHz audio bearer capability for outgoing fax calls, based on a Class of Service assigned to the analog (500/2500-type) phone
- the capability of defining the bearer as "Voice" or "3.1 KHz" on a system basis
- the addition and display of national or international prefix in front of the received Calling/ Connected Line Identification on incoming/outgoing EuroISDN calls
- flexible national and local prefixes in addition to the Calling/Connected Line Identification on incoming/outgoing calls based on the route configuration
- User-to-User information transparency in call control messages
- capability of mapping a PROGRESS message or Progress Indicator in a CALL PROCEEDING message into an ALERT or a CONNECT message on a single configuration basis
- capability of using different options to build the Calling Line Identification (CLID)
- called party number size increase to 31 octets

Note:

The older ESTI standards support a length of 23 octets only.

Thisdevelopment also provides a subset of the ETSI Generic functionalprotocol for the support of supplementary services. Implementationrelies on the Generic Functional (GF) Transport platform in the ISDNQSIG GF Transport feature. The ETSI GF subset provides two types ofsupplementary service control entities as follows:

- Bearer-related transport with a point-to-point transport mechanism. This service is used to transport supplementary service protocol information in association with a basic call.
- Bearer-independent transport with a point-to-point (connection-oriented) mechanism. This service is used to transport supplementary service protocol information, which is entirely independent of any existing basic call.

TheETSI GF provides a generic transport platform that supports ETSI-compliantsupplementary services. The ETSI GF protocol is implemented on theETSI, Swiss, German and Danish EuroISDN interfaces.

Operating parameters

Layer 1 and Layer 2 compliance with ETSI requirements are also supported.

If more than one Channel Identification Information Element (IE) is received in a SETUP message, only the first one is used by the system. If the Information Element is not available, the call is processed as each channel negotiation configuration. If no negotiation is allowed, the call is released.

The transfer of an unanswered EuroISDN call to a remote ringing phone requires disconnect supervision from a TIE trunk. If disconnect supervision is not available when an external user goes on-hook, the trunk could be locked out. Therefore, it is important to ensure that trunks used for a EuroISDN call have disconnect supervision.

With ETSI GF, DN address translation requires association with a customer number. In an ETSI basic call establishment, the customer number association is located through the B-channel, identified in the channel identification IE. If the DN address is not associated with a basic call, the customer number is determined through other means.

For Connected Line Identification Presentation (COLP) supplementary service, sending or restricting connected line identification is not supported on ISDN BRI phones on an individual call basis.

Feature interactions

Auto terminate Call

This feature is not supported.

Call Back Queuing and Off-hook Queuing

This feature is not supported with Overlap Signaling.

Call Completion Supplementary Service

Call Completion interacts with Call Transfer Over EuroISDN SN3. Call Completion Free Notification can only be presented to the Call Completion originating station and cannot be transferred to any other station. However, once the Call Completion call is established, it behaves as a normal call with respect to Call Transfer.

End-to-End Signaling

End-to-End Signaling is supported on all outgoing EuroISDN routes as soon as the CALL PROCEEDING message with a Progress Indicator is received.

Flexible Hotline

This feature is not supported with Overlap Signaling.

Incoming Digit Conversion Enhancement

The Incoming Digit Conversion Enhancement (IDC) feature converts incoming digits from a DID route. This feature is supported on the incoming EuroISDN DID routes. Digits received as a called party number are converted if the IDC feature is activated on the route. Digit analysis is then performed on the converted digits by the system.

ISDN Basic Rate Interface Connected Line

Presentation/Restriction

The EuroISDN Continuation capability adds National and Local prefixes to the connected number being sent. This is programmed on a route basis and is applicable to connected numbers received from a ISDN BRI terminal and sent over a ISDN trunk.

ISDN Calling Line Identification Enhancements

The EuroISDN Continuation feature allows Home National Numbers and Home Local Numbers to be configured on a route. When an ISDN call is made from a phone to a EuroISDN interface, the Calling Line Identification (CLID) constructed by EuroISDN, based on the outgoing route, takes precedence over the CLID constructed for the calling station phone.

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 all public Integrated Services Digital Network (ISDN) interfaces. When the Privacy Indicator Ignore (PII) prompt is YES in LD 16, the Calling Line Identification (CLID) Presentation Indicator and the Calling Party Name Display (CPND) Indicator override the blocking of the number and name details of the calling party. The name and number of the calling party appear on the receiving telephone.

The CPPE feature introduces a new route option AUXP for Auxiliary processor applications in LD 16 (RDB – Route Data Block). AUXP enhances the ability of the system to honor or ignore the Privacy Indicator for a Calling Party Privacy call, for each incoming route. If AUXP is 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, there will be no change to the CLID Presentation Indicator.

KD3 Signaling

Interworking with KD3 signaling is not supported.

Integrated Service Access (ISA)

Integrated Service Access is not supported.

Interworking Notification

Direct Interworking

When "direct interworking" occurs, the EuroISDN trunk terminates directly on the analog trunk. As a result, the CALL PROCEEDING message is sent to indicate that no further called party information is required. No further call control message is initiated by the analog trunk until answer supervision is provided. If the trunk is a supervised trunk, the resulting CONNECT message permits the Public Switched Telephone Network (PSTN) to start charging the caller. However, an unsupervised trunk can also be called. In this case, the CONNECT message is sent up on the expiry of the pseudo answer supervision timer.

The destination non-ISDN trunk can be busy instead of ringing; therefore, it is not effective to provide an ALERTING message, since the PSTN provides the caller with ringback. However, the PSTN will be running a T310 timer. Expiry of this timer causes call clearing, and a message must be provided.

The following three alternatives can be configured in LD 16 by responding to the PROG prompt to stop the network T310 timer:

1. A PROGRESS message is sent to the network after the CALL PROCEEDING message.
2. An ALERT message is sent to the network after the CALL PROCEEDING message.
3. A CONNECT message is sent to the network after the CALL PROCEEDING message.

Tandem Interworking

When tandem interworking occurs, the EuroISDN trunk terminates on an ISDN link terminating on an analog trunk. When the tandem SETUP message is sent, the CALL PROCEEDING message is sent to the PSTN to indicate that no further called party information is required. The tandem ISDN link returns a PROGRESS message, indicating interworking with a non-ISDN trunk.

Since the PSTN does not implement Annex N, proposed by ETSI, it runs a T310 timer. A message must be provided to stop the T310 timer.

The following three alternatives can be configured in PROG prompt LD 16 to stop the network T310 timer:

1. A PROGRESS message is sent to the network after the CALL PROCEEDING message.
2. An ALERT message is sent to the network after the CALL PROCEEDING message.
3. A CONNECT message is sent to the network after the CALL PROCEEDING message.

When interworking with an analog trunk without answer supervised signaling, the system should be configured to send a CONNECT message rather than an ALERT message. An ALERT or CONNECT message stops the receipt of digits, which implies that all digits have been received.

Networking Features

Some networking features currently exist on more than one ISDN interface implemented on the system. These features are listed in Table 46: Networkingfeatures supported by more than one ISDN interface on page 94. Any networking feature that does not appear in the table is only supported on one ISDN interface and is rejected by the gateway when the service is requested. That is the case for all Meridian Customer Defined Network (MCDN) features that are not supported over the EuroISDN or QSIG interface.

Table 46: Networkingfeatures supported by more than one ISDN interface

	Euro-ISDN	MCDN	QSIG	BRI phones	DPNSS	MFC	DTI2	Analog
Calling Line ID	Yes	Yes	Yes	Yes	Yes	No	No	No
Connected Number	Yes	Yes	Yes	No	Yes	No	No	No

	Euro-ISDN	MCDN	QSIG	BRI phones	DPNSS	MFC	DTI2	Analog
Transit Counter	No	Yes	Yes	No	Yes	No	No	No

Special Dial Tone after Dialed Numbers

Special dial tones after dialed numbers are not supported for incoming calls.

Transfer of Unanswered Call

The EuroISDN Continuation feature supports the transfer of a call from a phone dialing an external number, before the external phone answers, to both local and remote phones upon ringing.

Virtual Network Services

A EuroISDN link can be used as a B-Channel for the Virtual Network Services feature.

R2MFC Calling Number Identification/Call Detail Recording Enhancements

The outgoing Calling Line Identification (CLID) element of the EuroISDN Continuation feature is mutually exclusive with the R2MFC CNI/CDR Enhancements feature. If the CLID is to be composed from the EuroISDN Continuation feature, it will not contain the Calling Number Identification (CNI). If the CLID is to be composed from the CNI, no prefixes will be added to the number.

Feature packaging

EuroISDN Continuation is included in EuroISDN (EURO) package 261.

For Primary Rate Interface (PRI), the following software packages are required:

- Digit Display (DDSP) package 19
- Integrated Services Digital Network (ISDN) package 145
- 2.0 Mbit Primary Rate Interface (PRI2) package 154
- Integrated Services Digital Network Supplementary Features (ISDN INTL SUP) package 161
- International ISDN PRI (IPRA) package 202
- Multi-purpose Serial Data Link (MSDL) package 222
- Calling Party Privacy (CPP) package 301

For Basic Rate Interface (BRI), the following software packages are required:

- Digit Display (DDSP) package 19
- Integrated Services Digital Network (ISDN) package 145

- Integrated Services Digital Network Supplementary Features (ISDN INTL SUP) package 161
- International ISDN PRI (IPRA) package 202
- Meridian 1 Extended Peripheral Equipment (XPE) package 203
- Basic Rate Interface (BRI) package 216
- Integrated Services Digital Network Basic Rate Interface Trunk Access (BRIT) package 233

Overlap Signaling is included in (OVLP) package 184. For the Uniform Dialing Plan (UDP), the following packages are also required:

- Basic Routing (BRTE) package 14
- Digit Display (DDSP) package 19
- Network Call of Service (NCOS) package 32
- Basic Alternate Route Selection (BARS) package 57
- Network Alternate Route Selection (NARS) package 58

For the Coordinated Dialing Plan (CDP), package 59 and the following packages are required:

- Integrated Services Digital Network (ISDN) package 145
- 2.0 Mbit Primary Rate Interface (PRI2) package 154
- Flexible Numbering Plan (FNP) package 160
- Integrated Services Digital Network Supplementary Features (ISDN INTL SUP) package 161
- International Primary Rate Access (IPRA) package 202

The following packages are required for EuroISDN to interact with Multifrequency Compelled Signaling (MFC), Multifrequency Compelled Signaling for Socotel (MFE), Direct Inward Dialing (DID), Digital Access Signaling System #2 (DASS2), and Digital Private Network Signaling System #1 (DPNSS1):

- Universal ISDN Gateway (UIGW) package 283
- Enhanced DPNSS1 Gateway (DPNSS189I) package 284

Feature implementation

Tasksummary list

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

1. Table 47: LD 10 - Configure Fax Allowed Class of Service for analog (500-type) phone. on page 97
2. Table 48: LD 17 - Create a new D-channel. on page 97
3. Table 49: LD16 - Configure new DID routes (enter the same responses to the IFC and CNTY prompts that were used in LD 17). on page 98
4. Table 50: LD14 - Configure new DID trunks. on page 101

Table 47: LD 10 - Configure Fax Allowed Class of Service for analog (500-type) phone.

Prompt	Response	Description
REQ	NEW	Add new data.
TYPE	500	Type of phone.
...		
CLS	FAXA	Fax Class of Service allowed for phone or modem. ISDN call is generated with 3.1 KHz bearer capability. FAXD = Fax Class of Service denied.

Table 48: LD 17 - Create a new D-channel.

Prompt	Response	Description
REQ	NEW	Add.
TYPE	ADAN	Action Device and Number.
- ADAN	NEW DCH x	New D-channel at port number x.
- CTYP	MSDL	Card type, where: MSDL = Multipurpose Serial Data Link.
- GRP	0-4	Network Group Number.
- DNUM	0-15	Device number for I/O ports.
- PORT	0-3	Port number on MSDL card.
- USR	PRI	D-channel for ISDN PRI only. Precede with X to remove.

Prompt	Response	Description
IFC	EURO	Interface type for D-channel, where: EURO = EuroISDN.
	SWIS	IFC SWIS is for PRI2 (SN2).
- - CNTY	AUS DEN (ETSI) FIN GER ITA NOR POR SWE EIR DUT SWI BEL ESP UK	Austria. Denmark. ETS 300-102 basic protocol. Finland. Germany. Italy. Norway. Portugal. Sweden. Ireland. Holland. Switzerland. Belgium. Spain. United Kingdom.
...		
- - DCHL	0-159	PRI loop number.
- RCAP	COLP	Enter COLP to support Connected Line Identification Presentation as a remote capability. Enter XCOL to remove Connect Line Identification Presentation.
PINX_CUST	xx	This customer number will be used for DN address translation, associated with bearer independent connection messages received on this D-channel.
...		
PARM	CHG	Change system parameters.
BCAP	(SPEE) 31KH	Speech (the default). 3.1 Khz bearer capability setting for outgoing voice calls.

Table 49: LD16 - Configure new DID routes (enter the same responses to the IFC and CNTY prompts that were used in LD 17).

Prompt	Response	Description
REQ	NEW	Add.
TYPE	RDB	Route Data Block.
CUST	0-99	Customer number as defined in LD 15.
ROUT	0-511	Route number for Large System and CS 1000E system.

Prompt	Response	Description
TKTP ...	DID	Trunk type, where: DID = Direct Inward Dialing.
DTRK	YES	Digital trunk route.
- DGTP	PRI2 BRI	Digital trunk type for route. 2.0 Mbit Primary Rate Access. ISDN Basic Rate Access.
- IFC	EURO SWIS	Interface type for route EuroISDN For PRI2 (SN2).
- - CNTY	AUS DEN (ETSI) FIN GER ITA NOR POR SWE EIR DUT SWI BEL ESP UK	Austria. Denmark. ETS 300-102 basic protocol. Finland. Germany. Italy. Norway. Portugal. Sweden. Ireland. Holland. Switzerland. Belgium. Spain. United Kingdom.
CLID	OPT0	Prefix = 0, for North American dialing plan. This is the default value for ESIG and ISIG interfaces.
	OPT1	Prefix = 1, for international PFXs in CLID, any numbering type supported. This is the default value for all EuroISDN interfaces.
	OPT2	Prefix = 2, for international PFXs in CLID, CCITT numbering type supported: UKWN, INTL, NPA, and NXX. Default value for CO/DID routes for the Telecom New Zealand interface.
	OPT3	Prefix = 3, for international PFXs in CLID, only NXX number type supported. Default value for TIE routes for the Telecom New Zealand interface.
	OPT4	For international COs, if the call originates from a CO trunk type, add nothing. Otherwise, add PFX1 and PFX2. This is the default value for the Hong Kong, Singapore, and Thailand interfaces.
	OPT5	This is the same as OPT4, except it supports a maximum of 10 digits in the CLID. This is the default value for the Austrian interface.

Prompt	Response	Description
PROG	NCHG	Send PROGRESS (default value for all interfaces, except Austria).
	MALE	Send ALERT after CALL PROCEEDING.
	MCON	Send CONNECT after CALL PROCEEDING (this is the default value for the Austrian interface).
...		
RCAP	COLP	Connected Number IE Presentation is supported on the far end (enter XCOLP to remove COLP).
- CPFXS	(YES) NO	Customer-defined Prefixes option. This prompt is added as one of the sub-prompts of ISDN = YES. It is added at the end of the ISDN sub-prompts. If CPFXS = YES, when constructing the Calling or Connected Line Identification, the prefixes are retrieved from the Customer Data Block via the PFX1 and PFX2 prompts in LD 15, as is currently done. This is the default response. If CPFXS = NO, when constructing the Calling or Connected Line Identification, the prefixes are retrieved from the Route Data Block via the HNTN and HLCL prompts in LD 16.
- - HNTN	0-9999	Home National Number. This number is similar to the PFX1 number prompted in LD 15. It is added to this overlay so that this prefix can be configured on a route basis as required in some countries (e.g., Italy). As is the case with PFX1, the HNTN prefix can be from one-to-four digits long. This prompt is displayed only if CPFXS = NO. If only a <CR> is entered, this prompt keeps its previous configuration. If no value was configured previously, no value will be configured. Enter X to delete the digits.
HLCL	0-9999	Home Location Number. This number is similar to PFX2 number prompted in LD 15. It is added to this overlay so that this prefix can be configured on a route basis as required in some countries (e.g., Italy). As is the case with PFX2, the HLCL prefix can be from one-to-four digits long. This prompt is displayed only if CPFXS = NO. If only a <CR> is entered, this prompt keeps its previous configuration. If no value was configured previously, no value will be configured. Enter X to delete the digits.
ADDP	(NO) YES	If ADDP = YES, the prefixes 0 (national) or 00 (international) are added to the Calling Party Number if the Type of Number (TON) is public.

Prompt	Response	Description
		If ADDP = NO, the Calling or Connected Party Number displayed is not modified.

Table 50: LD14 - Configure new DID trunks.

Prompt	Response	Description
REQ	NEW xx	Add a new data block to the system. Follow NEW with a value of 1-255 to create that number of consecutive trunks.
TYPE	DID	Direct Inward Dialing data block.
TN	l s c u	Format for Large System and CS 1000E system, where l = loop, s = shelf, c = card, u = unit.
RTMB	0-511 1-4000	Route number for Large System and CS 1000E system.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 13: EuroISDN Continuation Phase III

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

The EuroISDN feature provides ISDN PRI and ISDN BRI connectivity between the system and Public Exchanges that comply with the European Telecom Standards Institute (ETSI) specification ETS 300 102 for the Layer 3. The interfaces provided by this feature also comply with the country-specific Application Documents for Austria, Denmark, Finland, Germany,

Holland, Ireland, Italy, Norway, Portugal, Sweden, and Switzerland (only ISDN BRI connectivity).

The EuroISDN Continuation feature expanded the scope of the original EuroISDN feature to provide application support for Belgium, Spain, Switzerland (ISDN PRI connectivity added to the ISDN BRI connectivity), and the United Kingdom.

EuroISDN Continuation Phase III adds France, Russia and the Ukraine to the list of countries having country-specific Application Documents compliance for EuroISDN.

In addition to providing all of the functionalities provided by the original introduction of EuroISDN and the EuroISDN Continuation, EuroISDN Continuation Phase III introduces the following functionalities for all EuroISDN interfaces:

- optional sending of last forwarding DN as CLID
- Trunk Route Optimization - before Answer applied to incoming EuroISDN trunks
- Numbering Plan Identification (NPI) and Type of Number (TON) included in CDR tickets for EuroISDN calls

Optional sending of last forwarding DN as CLID

In a call redirection scenario (Call Forward All Calls, Call Forward No Answer, Hunt, and the ACD redirections, Night Call Forward and Interflow) occurring at a gateway node, the (optional) capability, on a per-system basis, of sending the last forwarding DN as Calling Line Identification (CLID) information. This functionality is controlled by a new prompt, the OCLI (Outgoing CLID) prompt, that has been introduced in LD 15 and which has three possible responses (NO, EXT, or ALL).

This functionality applies to the following types of calls being redirected at a gateway node.

Note:

There is a limitation on the functionality if the last redirection is due to ACD NCFW or Interflow, and if the redirection is preceded by one or more redirections on the same node. In such a case, the called number on the node is sent as CLID rather than the ACD DN.

- Incoming EuroISDN DID or Central Office calls coming from a Public Network Central Office or a Private Exchange over a standard ISDN connectivity, are forwarded back to the Public Network.

  Note:

  Standard ISDN connectivity means all supported ISDN connectivities including, EuroISDN, Asia Pacific, and NI-2.

  - Station A, located at a Public Network Central Office or a Private Exchange, calls Station B at a Gateway node. Station B is redirected to Station C, located at the same or another Public Network Central Office or Private Exchange as Station A. If OCLI = EXT or ALL,

the CLID of Station B is sent to Station C. Refer to Figure 4: Incoming EuroISDN call redirected back to the Public Network/PBX on page 105.

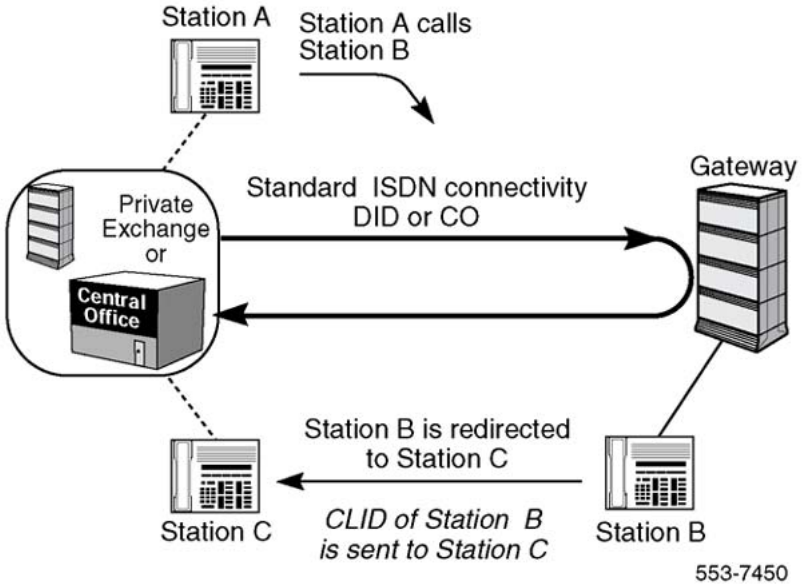


Figure 4: Incoming EuroISDN call redirected back to the Public Network/PBX

- MCDN/QSIG calls from an originating node, are being redirected to a Public Network Central Office or a Private Exchange over a standard ISDN connectivity.
  - Station A, located at an originating node, calls Station B located at a gateway node. Station B is redirected to Station C, located at a Public Network Central Office or a Private Exchange. If OCLI = ALL, the CLID of Station B is sent to Station C. Refer to Figure 5: MCDN/QSIG call redirected to the Public Network/PBX on page 106.

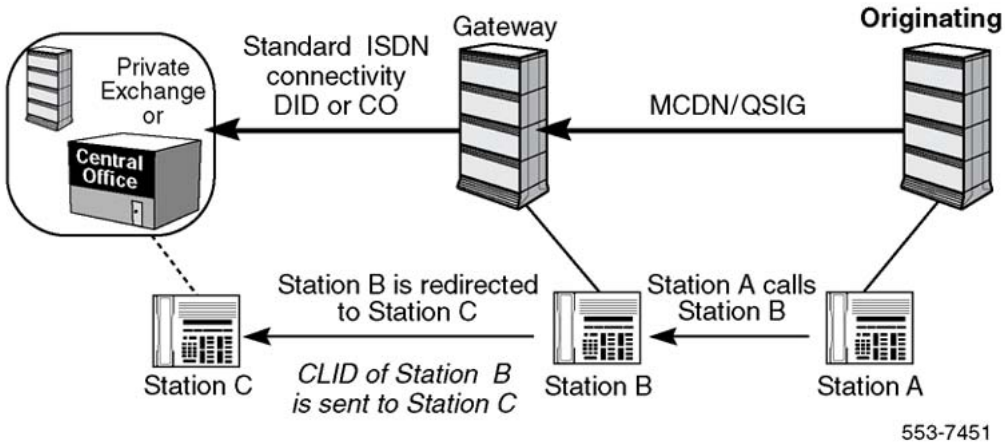


Figure 5: MCDN/QSIG call redirected to the Public Network/PBX

- Internal calls (in a stand-alone scenario) are being redirected to a Public Network Central Office or a Private Exchange over a standard ISDN connectivity.

  - Station A, Station B, and Station C are all located on the same gateway node. Station A calls Station B, which is redirected to Station C. Station C is redirected to Station D located at a Public Network Central Office or a Private Exchange. If OCLI = ALL, the CLID of Station C is sent to Station D. Refer to Figure 6: Internal call (stand-alone case) redirected to the Public Network/PBX on page 107.

    Note:

    If Station B is redirected to an ACD queue through ACD NCFW or Interflow across a Standard ISDN connectivity, then the CLID sent to Station C will be that of the number called by Station A rather than that of the ACD DN. There is a limitation on the functionality if the last redirection is due to ACD NCFW or Interflow, and if the redirection is preceded by one or more redirections on the same node. In such a case, the called number on the node is sent as CLID rather than the ACD DN..

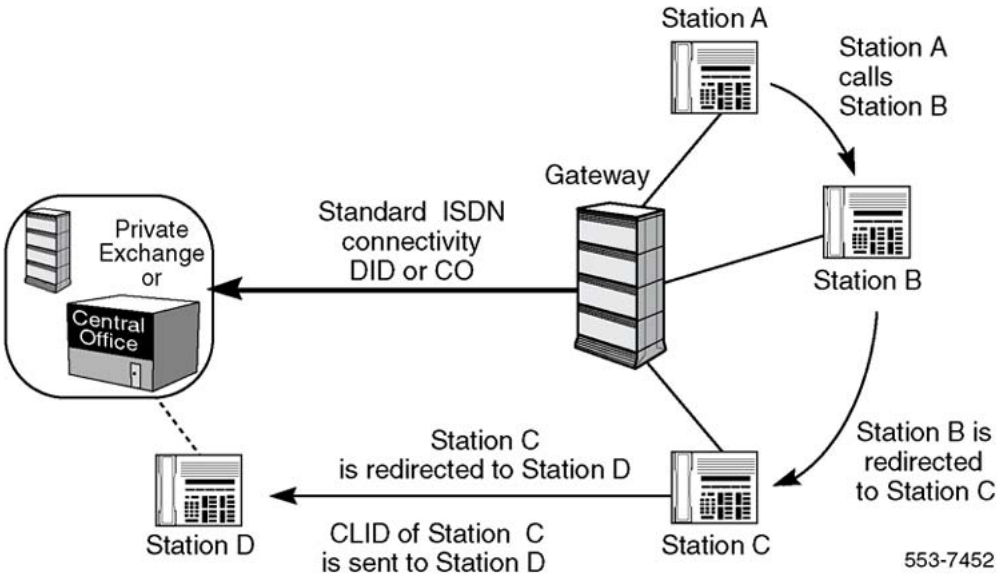


Figure 6: Internal call (stand-alone case) redirected to the Public Network/PBX

When the redirection does not occur at the gateway node for calls redirected to a Public Network Central Office or a Private Exchange over a standard ISDN connectivity, the outgoing CLID is that of the redirecting number information included in the SETUP message and received at the gateway node. The redirecting number information is the first redirecting DN on the last redirecting node.

- Station A and Station B are located at the same originating node. Station A calls Station B, which is redirected to Station C located at a transit node. Station C is redirected to Station D on the same transit node. Station D is redirected through a gateway node, to Station E located at a Public Network Central Office or a Private Exchange. If OCLI = ALL, the CLID of Station C is sent to Station E, since the redirecting number information is the first redirecting DN (that of Station C) on the last redirecting node (at the transit node). Refer to Figure 7: MCDN call redirected to the Public Network/PBX from a non-gateway node on page 108.

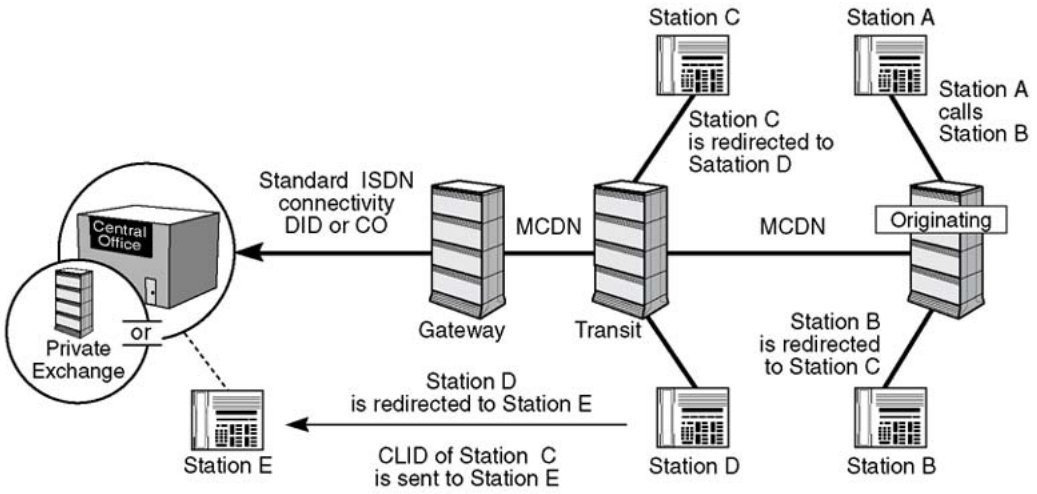


Figure 7: MCDN call redirected to the Public Network/PBX from a non-gateway node

Trunk Route Optimization - before Answer on EuroISDN trunks

Trunk Route Optimization before Answer (TRO-BA) is enhanced to allow TRO-BA to be applied to EuroISDN trunks. The trunk optimization occurs within an MCDN network to eliminate redundant trunks due to a call redirection scenario (Call Forward All Calls, Call Forward No Answer, Call Forward Busy, and Hunt).

- Station A, located at a Public Network Central Office or a Private Exchange, calls Station B at a terminating node over an MCDN trunk. Station B is redirected to Station C, located at a transit node, over another (redundant) MCDN trunk (refer to Figure 8: EuroISDN calling scenario before Trunk Optimization is applied on page 108).

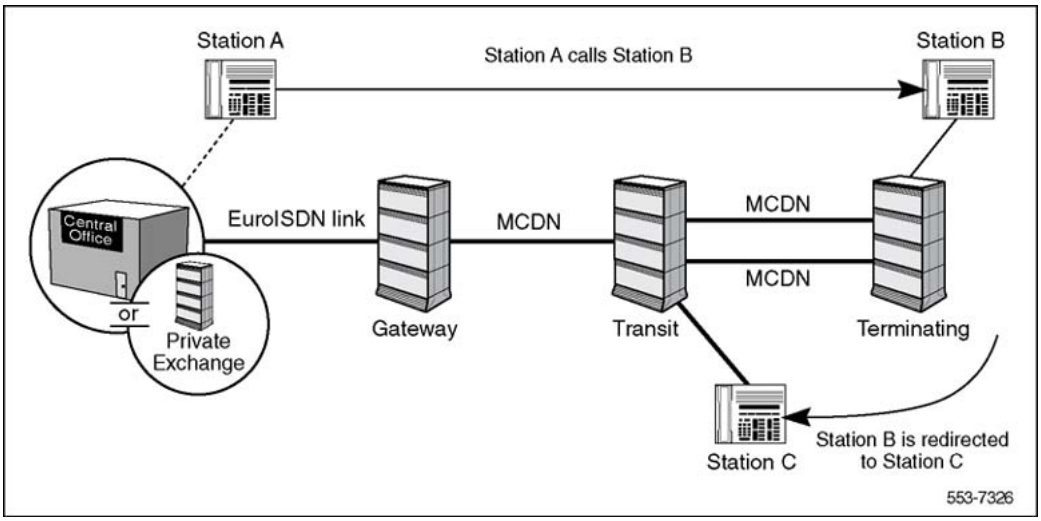


Figure 8: EuroISDN calling scenario before Trunk Optimization is applied

- As soon as the redirection occurs at the redirecting node (the terminating node), trunk optimization is invoked, the redundant trunks between the terminating node and the transit

node are dropped, and Station C and Station A connected over one MCDN trunk. Refer to Figure 9: EuroISDN calling scenario after Trunk Optimization has been applied on page 109.

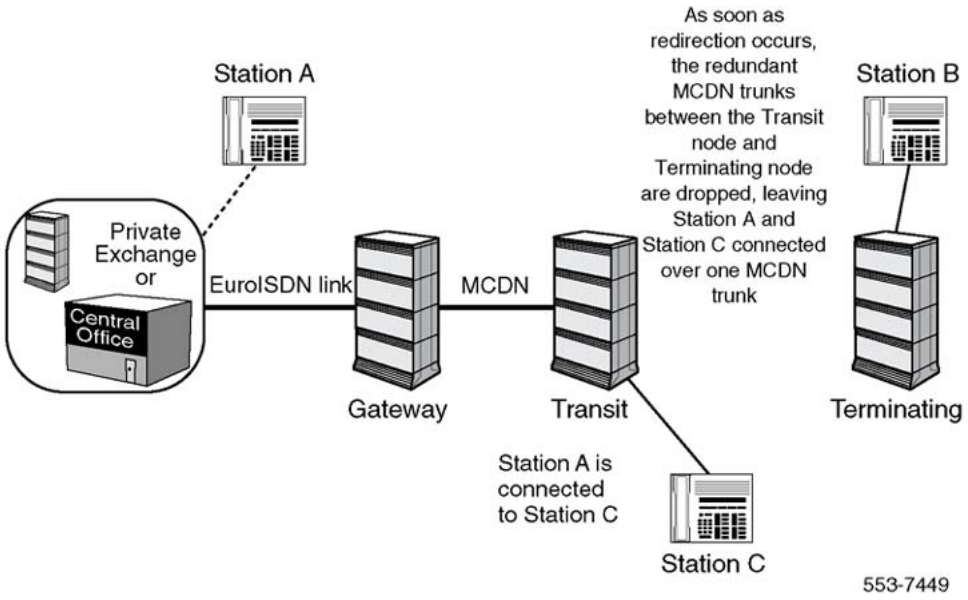


Figure 9: EuroISDN calling scenario after Trunk Optimization has been applied

Numbering Plan Identification and Type of Number in CDR tickets for EuroISDN calls

The Numbering Plan Identification (NPI) and Type of Number (TON) fields can be optionally included in the CDR tickets, for EuroISDN calls. The purpose is to allow billing the charge of a EuroISDN call to the call originator. By entering NEW against the FCDR (Format for Call Detail Recording) prompt, and YES against the CLID prompt in LD 17, the Numbering Plan Identification (NPI) and Type of Number (TON) fields are printed as part of the CDR ticket in addition to the CLID field. This functionality applies only for incoming calls over ISDN trunks, since the NPI and TON are derived from the incoming trunk.

The NPI and TON fields are left blank if the calls are internal, are over outgoing ISDN trunks, are over incoming non-ISDN trunks, or if the FCDR prompt is set to OLD or the CLID prompt is set to NO (note that the FCDR prompt takes precedence over the CLID prompt, that is, if FCDR is set to OLD and the CLID prompt is set to YES, the fields are left blank.)

The NPI and TON are both printed on the third line of the CDR ticket, at characters 44-45 and 47, respectively.

Operating parameters

EuroISDN is not supported on non-downloadable D-channel cards.

Interworking with KD3 signaling is not supported.

Integrated Service Access (ISA) is not supported.

Advice of Charge for EuroISDN calls is supported with the Advice of Charge for EuroISDN feature.

The EuroISDN to CIS MF Shuttle gateway is supported, as provided by the CIS MF Shuttle feature.

The Auto Terminate Call feature is not supported.

The Call Back Queuing and Off-hook Queuing feature and the Flexible Hotline feature are not supported with Overlap Signaling.

End-to-End Signaling is supported on all outgoing EuroISDN routes as soon as the CALL PROCEEDING message with a Progress Indicator is received.

Special dial tones after dialed numbers are not supported for incoming calls.

For Connected Line Identification Presentation (COLP) supplementary service, sending or restricting connected line identification is not supported on a single call basis on non-ISDN BRI phones.

For the Optional Sending of Last Forwarding DN as CLID functionality, if more than one ACD redirection takes place when an ACD phone is involved, the DN of the called phone is sent as the CLID rather than the DN of the last forwarding phone. Refer to Optional sending of last forwarding DN as CLID on page 104 for an example.

Feature interactions

Call Completion Supplementary Service

Call Completion interacts with Call Transfer Over EuroISDN SN3. Call Completion Free Notification can only be presented to the Call Completion originating station and cannot be transferred to any other station. However, once the Call Completion call is established, it behaves as a normal call with respect to Call Transfer.

Call Detail Recording

The Optional Sending of Last Forwarding DN as CLID functionality interacts with the CDR feature. The CLID field in the CDR ticket for the redirected outgoing EuroISDN call is modified according to the CLID information in the SETUP message, for the various scenarios described in the section Optional sending of last forwarding DN as CLID on page 104.

CLID Enhancements

The Optional Sending of Last Forwarding DN as CLID functionality interacts with the ISDN CLID Enhancements feature, in that the CLID entry programmed against the prime DN of the forwarding phone at a gateway node is used to build the outgoing CLID.

Incoming Digit Conversion Enhancement

The Incoming Digit Conversion Enhancement (IDC) feature converts incoming digits from a DID route. This feature is supported on the incoming EuroISDN DID routes. Digits received as a called party number are converted if the IDC feature is activated on the route. Digit analysis is then performed on the converted digits by the system.

Incoming Trunk Programmable CLID

The Optional Sending of Last Forwarding DN as CLID functionality interacts with the Incoming Trunk Programmable CLID feature. This feature provides the capability of assigning a billing number to incoming trunks.

In the scenario described on Figure 7: MCDN call redirected to the Public Network/PBX from a non-gateway node on page 108, whereby the redirection does not occur at the gateway node, the incoming redirecting number information is used as the outgoing CLID on the standard ISDN connectivity DID or CO trunk, even though the Incoming Trunk Programmable CLID feature is configured to send a billing number as CLID information.

Intercept Treatment on Invalid or Partial Dialing

The Partial Dial Timing (PRDL), Vacant Number Routing (VNR), and Intercept features handle dialing irregularities on the system.

The Intercept feature allows calls with dialing irregularities to be routed to an attendant or to a Recorded Announcement, or to receive overflow or busy tone. The VNR feature allows calls to a vacant number to be routed to another node, where the call is treated as a vacant number, given intercept treatment, or terminated. The PRDL feature allows incoming non ISDN DID calls to be routed to an attendant after a configured amount of time.

With respect to EuroISDN calls with dialing irregularities, the PRDL feature has been enhanced so that when the End of Dial (EOD) timer for PRDL has expired, the calls, instead of being automatically routed to an attendant, now receive the intercept treatment provided by the VNR and Intercept features. A proviso is that the EOD timer used as the PRDL timer must be configured to be at least two seconds less than the ISDN interdigit timer (T302). Otherwise, the calls are rejected if the T302 times out first.

Interworking Notification

Direct Interworking

When "direct interworking" occurs, the EuroISDN trunk terminates directly on the analog trunk. As a result, the CALL PROCEEDING message is sent to indicate that no further called party information is required. No further call control message is initiated by the analog trunk until answer supervision is provided. If the trunk is a supervised trunk, the resulting CONNECT message will permit the Public Switched Telephone Network (PSTN) to start charging the caller. However, an unsupervised trunk can also be called. In this case, the CONNECT message is sent upon the expiry of the pseudo answer supervision timer.

The destination non-ISDN trunk can be busy instead of ringing; therefore, it is not effective to provide an ALERTING message, since the PSTN provides the caller with ringback. However, the PSTN will be running a T310 timer. Expiry of this timer causes call clearing, and a message must be provided.

The following three alternatives can be configured in LD 16 by responding to the PROG prompt to stop the network T310 timer:

1. A PROGRESS message is sent to the network after the CALL PROCEEDING message.
2. An ALERT message is sent to the network after the CALL PROCEEDING message.
3. A CONNECT message is sent to the network after the CALL PROCEEDING message.

Tandem Interworking

When tandem interworking occurs, the EuroISDN trunk terminates on an ISDN link terminating on an analog trunk. When the tandem SETUP message is sent, the CALL PROCEEDING message is sent to the PSTN to indicate that no further called party information is required. The tandem ISDN link returns a PROGRESS message, indicating interworking with a non-ISDN trunk.

Since the PSTN does not implement Annex N, proposed by ETSI, it runs a T310 timer. A message must be provided to stop the T310 timer.

The following three alternatives can be configured in PROG prompt LD 16 to stop the network T310 timer:

1. A PROGRESS message is sent to the network after the CALL PROCEEDING message.
2. An ALERT message is sent to the network after the CALL PROCEEDING message.
3. A CONNECT message is sent to the network after the CALL PROCEEDING message.

When interworking with an analog trunk without answer supervised signaling, the system should be configured to send a CONNECT message rather than an ALERT message. An ALERT or CONNECT message stops the receipt of digits, which implies that all digits have been received.

Networking Features

Some networking features currently exist on more than one ISDN interface implemented on the system. These features are listed in Table 51: Networking features supported by more than one ISDN interface on page 113. Any networking feature that does not appear in the table is only supported on one ISDN interface and is rejected by the gateway when the service is requested. That is the case for all Meridian Customer Defined Network (MCDN) features that are not supported over the EuroISDN or QSIG interface.

Table 51: Networking features supported by more than one ISDN interface

	Euro-ISDN	MCDN	QSIG	ETSI BRI phones	DPNSS 1	MFC	DTI2	Analog
Calling Line ID	Yes	Yes	Yes	Yes	Yes	No	No	No
Connected Number	Yes	Yes	Yes	No	Yes	No	No	No
Transit Counter	No	Yes	Yes	No	Yes	No	No	No

Transfer of Unanswered Call

The EuroISDN Continuation Phase III feature supports the Transfer of an Unanswered Call feature. This allows a station (an attendant or phone) to transfer an unanswered outgoing EuroISDN call, after dialing has been completed and before the transferred-to phone answers (while the phone is ringing). The transferred-to phone can be local, or can be remote if the link is an MCDN link equipped with Network Attendant Services (NAS), or if the link is a QSIG link equipped with Slow Answer Recall.

Note:

The trunk type for the EuroISDN link must be either CO, DID, TIE, FEX, or WATTS for this functionality to work. Otherwise, the transferred-to phone must answer before the transfer can be done.

The transfer of an unanswered EuroISDN call interacts with the CDR feature. When the originating party completes the call transfer after the transferred-to phone answers, a CDR Start (S) is generated when the transferred-to phone answers. When one side releases the call, a CDR End (E) ticket is generated. If the originating party completes the transfer while the transferred-to phone is ringing, an S ticket will be generated only when the transferred-to phone answers.

Virtual Network Services

A EuroISDN link can be used as a B-Channel for the Virtual Network Services feature.

Feature packaging

EuroISDN Continuation is included in EuroISDN (EURO) package 261.

For Primary Rate Interface (PRI), the following software packages are required:

- Digit Display (DDSP) package 19
- Integrated Services Digital Network (ISDN) package 145
- 2.0 Mbit Primary Rate Interface (PRI2) package 154
- Integrated Services Digital Network Supplementary Features (ISDN INTL SUP) package 161
- International ISDN PRI (IPRA) package 202
- Multi-purpose Serial Data Link (MSDL) package 222

For Basic Rate Interface (BRI), the following software packages are required:

- Digit Display (DDSP) package 19
- Integrated Services Digital Network (ISDN) package 145
- Integrated Services Digital Network Supplementary Features (ISDN INTL SUP) package 161
- International ISDN PRI (IPRA) package 202
- Meridian 1 Extended Peripheral Equipment (XPE) package 203

- Basic Rate Interface (BRI) package 216
- Integrated Services Digital Network Basic Rate Interface Trunk Access (BRIT) package 233

Overlap Signaling is included in (OVLP) package 184. For the Uniform Dialing Plan (UDP), the following packages are also required:

- Basic Routing (BRTE) package 14
- Digit Display (DDSP) package 19
- Network Call of Service (NCOS) package 32
- Basic Alternate Route Selection (BARS) package 57 or Network Alternate Route Selection (NARS) package 58 (for the Uniform Dialing Plan)

For the Coordinated Dialing Plan (CDP), package 59 and the following packages are required:

- Integrated Services Digital Network (ISDN) package 145
- 2.0 Mbit Primary Rate Interface (PRI2) package 154
- Flexible Numbering Plan (FNP) package 160
- Integrated Services Digital Network Supplementary Features (ISDN INTL SUP) package 161
- International Primary Rate Access (IPRA) package 202

The following packages are required for EuroISDN to interact with Multifrequency Compelled Signaling (MFC), Multifrequency Compelled Signaling for Socotel (MFE), Direct Inward Dialing (DID), Digital Access Signaling System #2 (DASS2), and Digital Private Network Signaling System #1 (DPNSS1):

- Universal ISDN Gateway (UIGW) package 283
- Enhanced DPNSS1 Gateway (DPNSS189I) package 284

The following packages are required for the Numbering Plan Identification (NPI) and Type of Number (TON) fields in CDR:

- Call Detail Recording(CDR) package 4
- Call Detail Recording on Teletype Terminal (CDR_TTY) package 5
- Calling Line Identification in Call Detail Recording (CLID_CDR_PKG) package 118
- New Format for Call Detail Recording (CDR_NEW) package 234

Feature implementation

Task summary list

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

1. Table 52: LD 15 - Configure the Optional Sending of Forwarding CLID functionality (in response to the OCLI prompt). on page 116
2. Table 53: LD 17 - Create a new D-channel for the EuroISDN interface. on page 117
3. Table 54: LD 16 - Configure new DID routes (enter the same responses to the IFC and CNTY prompts that were used in LD 17). on page 118
4. Table 55: LD 17 - Configure the Numbering Plan Identification (NPI) and Type of Number (TON) fields to be included in the CDR tickets. on page 121

Table 52: LD 15 - Configure the Optional Sending of Forwarding CLID functionality (in response to the OCLI prompt).

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	NET	Networking data.
CUST	0-99	Customer number as defined in LD 15.
ISDN	YES	Integrated Services Digital Network.
PNI	(0)-32700	Private Network Identifier.
SATD	0-(1)-5	Satellite Delays.
OCLI	(NO)	NO = No manipulation is done on outgoing CLID for calls forwarded over EuroISDN links.
	EXT	EXT = The last forwarding DN is sent as CLID information for incoming calls over a standard ISDN (EuroISDN, APAC, NI-2) DID or CO link to a gateway system node, and redirected back over the standard ISDN (EuroISDN, APAC, NI-2) DID or CO link.
	ALL	ALL = The same as for EXT. Moreover, the last forwarding DN is sent as CLID information for internal calls from a local phone or over ISDN TIE trunks, except for DPNSS1 calls which are redirected from a gateway node over a standard ISDN (EuroISDN, APAC, NI-2) DID or CO link.

Prompt	Response	Description
		If the redirection does not occur at the gateway node, the redirecting information number, if present in the incoming SETUP message received at the gateway node from the private network, is sent as CLID.

Table 53: LD 17 - Create a new D-channel for the EuroISDN interface.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	ADAN	Action Device and Number.
- ADAN	NEW DCH x	New D-channel at port number x.
	MSDL DCHI	Multipurpose Serial Data Link or Downloadable D-Channel card.
- GRP	0-7	Network Group Number.
- DNUM	0-15	Device number for I/O ports.
- PORT	0-3	Port number on MSDL card.
- USR	PRI	D-channel for ISDN PRI only. Precede with X to remove.
- IFC		DCH interface type.
	EURO	EURO = EuroISDN
- - CNTY		Enter country pertaining to EuroISDN interface.
	AUS DEN (ETSI) FIN GER ITA NOR POR SWE EIR DUT SWI BEL ESP UK FRA	Austria Denmark ETS 300-102 basic protocol Finland Germany Italy Norway Portugal Sweden Ireland Holland Switzerland Belgium Spain United Kingdom France
PINX_CUST	xx	The customer number used for the DN address.
...		

Prompt	Response	Description
DCHL	0-159	The PRI loop number for the D-Channel.
- - CNEG	(1) 2	Channel Negotiation option. (1) = Channel is indicated and no alternative is acceptable. This is the default value for all EuroISDN Interfaces except FRA. 2 = Channel is indicated and any alternative is acceptable. This is the default value for the FRA interface.
- RCAP	COLP XCOL	Enter COLP to support Connected Line Identification Presentation as a remote capability. This is the default value for the ESIG, ISIG, NI2, and EURO interfaces. Enter XCOL to remove Connect Line Identification Presentation as a remote capability. This is the default value for the APAC, AUS, EIR, DEUT, ESP, BEL, and FRA interfaces.

Table 54: LD 16 - Configure new DID routes (enter the same responses to the IFC and CNTY prompts that were used in LD 17).

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	RDB	Route Data Block.
CUST	0-99	Customer number as defined in LD 15.
ROUT	0-511	Route number for Large System and CS 1000E system.
TKTP	 DID COT	Trunk type. DID = Direct Inward Dialing COT = Central Office.
DTRK	YES	Digital trunk route.
- DGTP	 PRI2 BRI	Digital trunk type for route. 2.0 Mbit Primary Rate Access. ISDN Basic Rate Access.
...		
- IFC	 EURO	Interface type. EURO = EuroISDN.
- - CNTY		Enter country pertaining to EuroISDN and Asia Pacific interface type.
	AUS DEN (ETSI) FIN GER	Austria Denmark ETS 300-102 basic protocol Finland Germany

Prompt	Response	Description
	ITA NOR POR SWE EIR DUT SWI BEL ESP UK	Italy Norway Portugal Sweden Ireland Holland Switzerland Belgium Spain United Kingdom
ICOG	IAO OGT ICT	Incoming and Outgoing trunk. Outgoing trunk only. Incoming trunk only.
ACOD	x..x	Access code for this trunk route.
- CLID	OPT0	Prefix = 0, for North American dialing plan. This is the default value for ESIG and ISIG interfaces.
	OPT1	Prefix = 1, for international PFXs in CLID, any numbering type supported. This is the default value for all EuroISDN interfaces.
	OPT2	Prefix = 2, for international PFXs in CLID, CCITT numbering type supported: UKWN, INTL, NPA, and NXX. Default value for CO/DID routes for the Telecom New Zealand interface.
	OPT3	Prefix = 3, for international PFXs in CLID, only NXX number type supported. Default value for TIE routes for the Telecom New Zealand interface.
	OPT4	For international COs, if the call originates from a CO trunk type, add nothing. Otherwise, add PFX1 and PFX2. This is the default value for the Hong Kong, Singapore, and Thailand interfaces.
	OPT5	This is the same as OPT4, except it supports a maximum of 10 digits in the CLID. This is the default value for the Austrian interface.
- PROG		Progress Signal.
	NCHG	Send PROGRESS (default value for all interfaces, except Austria).
	MALE	Send ALERT after CALL PROCEEDING.
	MCON	Send CONNECT after CALL PROCEEDING (this is the default value for the Austrian interface).
...		

Prompt	Response	Description
- RCAP	COLP	Enter COLP to support Connected Line Identification Presentation as a remote capability. This is the default value for the ESIG, ISIG, NI2, and EURO interfaces.
	XCOL	Enter XCOL to remove Connect Line Identification Presentation as a remote capability. This is the default value for the APAC, AUS, EIR, DEUT, ESP, BEL, and FRA interfaces.
...		
- CPFXS	(YES) NO	Customer-defined Prefixes option. This prompt is added as one of the sub-prompts of ISDN = YES. It is added at the end of the ISDN sub-prompts. If CPFXS = YES, when constructing the Calling or Connected Line Identification, the prefixes are retrieved from the Customer Data Block via the PFX1 and PFX2 prompts in LD 15, as is currently done. This is the default response. If CPFXS = NO, when constructing the Calling or Connected Line Identification, the prefixes are retrieved from the Route Data Block via the HNTN and HLCL prompts in LD 16.
- - HNTN	0-9999 X	Home National Number. This number is similar to the PFX1 number prompted in LD 15. It is added to this overlay so that this prefix can be configured on a route basis as required in some countries (e.g., Italy). As is the case with PFX1, the HNTN prefix can be from one-to-four digits long. This prompt is displayed only if CPFXS = NO. If only a <CR> is entered, this prompt keeps its previous configuration. If no value was configured previously, no value will be configured. Enter X to delete the digits.
HLCL	0-9999	Home Location Number. This number is similar to PFX2 number prompted in LD 15. It is added to this overlay so that this prefix can be configured on a route basis as required in some countries (e.g., Italy). As is the case with PFX2, the HLCL prefix can be from one-to-four digits long. This prompt is displayed only if CPFXS = NO. If only a <CR> is entered, this prompt keeps its previous configuration. If no value was configured previously, no value will be configured. Enter X to delete the digits.
ADDP	(NO)	If ADDP = NO, the Calling or Connected Party Number displayed is not modified.

Prompt	Response	Description
	YES	If ADDP = YES, the prefixes 0 (national) or 00 (international) are added to the Calling Party Number if the Type of Number (TON) is public.

For the following table for LD 17, Configure the Numbering Plan Identification (NPI) and Type of Number (TON) fields to be included in the CDR tickets, in addition to the CLID, for EuroISDN calls (this is automatically done by entering YES in response to the CLID prompt).

Table 55: LD 17 - Configure the Numbering Plan Identification (NPI) and Type of Number (TON) fields to be included in the CDR tickets.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	PARM	Change system parameters.
...		
- FCDR	NEW	Format for Call Detail Recording. Enter NEW for new format.
...		
- CLID	(NO) YES	Enter YES to include the TON and NPI fields in the CDR ticket, in addition to the CLID.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 14: EuroISDN ETS 300 403 Compliance Update

Contents

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

Feature description

The EuroISDN feature was initially introduced to provide ISDN PRI and ISDN BRI connectivity between the system and Public Exchanges that comply to the European Telecom Standards Institute (ETSI) specification ETS 300 102 for the Layer 3. The interfaces provided by this feature complied with the country-specific Application Documents for Austria, Denmark, Finland, Germany, Holland, Ireland, Italy, Norway, Portugal, Sweden, and Switzerland (only ISDN BRI connectivity).

The EuroISDN continuation feature expanded the scope of the original EuroISDN feature to provide application support for Belgium, Spain, Switzerland (ISDN PRI connectivity added to the ISDN BRI connectivity), and the United Kingdom.

EuroISDN Continuation Phase III, added France, Russia and the Ukraine to the list of countries having country-specific Application Documents comply with ETS 300 102 for EuroISDN.

The EuroISDN ETS 300 403 Compliance Update feature has been introduced to provide ISDN Primary Rate Interface and Basic Rate Interface to Central Offices that comply with the ETS 300 403-1 European Telecom Standards Institute (ETSI) standard.

Note:

In order to support countries that have not upgraded to the ETS 300 403 standard, the system still interworks with Central Offices conforming to the ETS 300 102 standard.

When programming the D-Channel for PRI2 trunks (in LD 17) or a PRI2 route for ISDN trunks (in LD 16) for an ETS 300 403 interface, the following applies:

- If IFC = E403 and CNTY = ETSI (ETS 300 403 for the user side) or NET (ETS 300 403 for the network side), then the interface is fully compliant with ETS 300 403.
- If IFC = E403 and CNTY = any of the supported country entries except ETSI and NET, then the interface behaves like an ETS 300 102 extended version, that is, in addition to the existing ETS 300 102 capabilities, the bearer capability and High Layer Compatibility selection procedures (fall-back mechanism), and the basic telecommunication service identification are also implemented in order to take advantage of teleservices offered, such as 7kHz telephony and Videotelephony.

Note:

A user can configure an interface fully compliant with the ETS 300 102 standard if IFC = EURO and CNTY = any of the supported country values.

With the EuroISDN ETS 300 403 Compliance Update feature, all of the functionalities provided by the previous offerings of the EuroISDN features are supported, along with the following enhancements:

- Support for signaling procedures for bearer capability and High Layer Capability selections by, providing repeatable Bearer capability, High layer compatibility Information Elements (IEs) in the SETUP message. ALERT, CALL PROCEEDING, CONNECT, and PROGRESS messages can be used to confirm the selected Bearer Capability/High Layer compatibility for the called user.
- This enhancement provides high-quality bearer services or teleservices, with alternate bearer capability or high layer compatibility in case of fall-back.
- Support for Basic telecommunication service identification. Each basic telecommunication service has the required Bearer capability IE encoding and, if applicable, the required High Layer compatibility IE encoding defined for that service. The requested teleservice is identified by taking the presented Bearer capability and High layer compatibility information elements in all combinations. If there is no valid combination, the presented Bearer Capability IE is considered in order to identify a bearer service.

Existing EuroISDN functionalities supported by the EuroISDN ETS 300 403 Compliance Update feature

This section summarizes the existing EuroISDN functionalities that are supported by the EuroISDN ETS 300 403 Compliance Update feature.

The ETSI GF subset provides the following types of functions to the supplementary services control entities:

- Bearer-related transport with a point-to-point transport mechanism. This service is used for the transport of supplementary service protocol information in association with a basic call.
- Bearer-independent transport with a point-to-point facility. This service is used for the transport of supplementary service protocol information which are entirely independent of any existing basic calls.

Note:

The ETSI GF does not by itself control any supplementary service but rather provides a generic transport platform that will support ETSI compliance supplementary services. The ETSI Generic Functional protocol is implemented for all ETSI interfaces.

The following call services are supported by the system on the EuroISDN connectivities:

- Basic call service (3.1 kHz, speech, unrestricted digital information)
- 64 Kbit/s bearer capability
- DID, DOD, COT and TIE call types
- Channel negotiation
- Enbloc dialing
- Overlap sending and Overlap receiving
- Flexible numbering plan
- Calling Line Identification Presentation and Calling Line Identification Restriction
- Connected Line Presentation and Connected Line Restriction
- MCID (Malicious Call Identification)
- Call Completion to a Busy Subscriber (CCBS)
- 7 kHZ telephony (with fall-back to 3.1 kHz)
- Videotelephony
- Advice of Charge

Feature operation on page 145 summarizes the networking features supported at gateways with the EuroISDN connectivity.

Any feature that is not listed in this table is not supported at gateways with EuroISDN connectivity. A "YES"/ "NO" indicates that the gateway is/is not supported. A "NS" indicates the service is not supported on the specified interface.

Table 56: Networking features supported by more than one ISDN interface

	Euro-ISDN	MCDN	QSIG	ETSI BRI phones	DPNSS1	MFC	DTI2	Analog	CIS	KD3
Calling Line ID	Yes	Yes	Yes	Yes	Yes	No	No	No	No	No[4]
ConnectedNumber	Yes	Yes	Yes	No	Yes	No	No	No	No	No[4]
Transit Counter	No	Yes[1]	Yes[2]	No	Yes[3]	No	No	No	No	No[4]

Note:

This is supported using the ICCL tandem count feature existing over the NAS feature.

Note:

This is only supported for ETSI version of QSIG. For the ISO version, it is discarded.

Note:

This is supported using the Loop Avoidance supplementary service.

Note:

The basic call is not supported at the gateway.

The following functionalities are available for EuroISDN interfaces. When not specified these functionalities are network wide:

- Intercept Treatment on reception of invalid or incomplete called party number
- Interception to an Attendant for incoming EuroISDN voice calls terminating on a data phone
- capability to listen to Tones and Announcements provided by the Central Office on call clearing
- capability of Transferring outgoing EuroISDN calls after completion of dialling (applies only for Italy)
- Calling Line Identification and Connected Line Identification transparency to/from EuroISDN to/from a BRI phone

- Connected Line Presentation service configurable on a D-Channel basis as a remote capability
- capability of providing 3.1KHz audio bearer capability for outgoing fax calls, based on a Class of Service assigned to a 500/2500 phone
- capability of defining bearer capability "Speech" or "3.1KHz" on a system basis (this functionality is also provided on ISDN interfaces)
- addition and display of National or International Prefix on top of the received Calling/ Connected Line Identification on incoming/outgoing EuroISDN calls
- Flexible National and Local prefixes addition in the Calling/Connected Line Identification for outgoing/incoming calls, based on the route configuration
- user to user information transparency in call control message (only for EuroISDN trunk interfaces)
- called party number size increase to 31 octets.
- capability of mapping a PROGRESS message or a Progress Indicator in the CALL PROCEEDING message, into an ALERT or a CONNECT message on a single configuration basis
- capability of using different options to build the CLID
- capability of sending a RELEASE message to a BRI phone, with cause and Progress Indicator (PI) Number 8. The user is provided with display of cause in addition of tone, for the following scenarios:
  - internal BRI phone originating call to busy or invalid DN.
  - when receiving a disconnect message from the Central Office through EuroISDN (if a Progress Indicator (PI) Number 8 is not present in the disconnect message, tone is provided locally. Otherwise, inband tone is remotely provided).
  - when receiving a disconnect message over an MCDN network.
- capability of immediately releasing rejected BRI data calls without providing tone provision to the BRI terminal
- capability, on a single configuration basis, of sending last forwarding DN as a CLID information, for the following types of calls:
  - incoming EuroISDN calls being forwarded back to the public network through EuroISDN
  - incoming MCDN/QSIG calls forwarded to the public network through EuroISDN
  - internal calls (stand-alone case) being forwarded to the public network through EuroISDN
- trunk optimization before call establishment (TRO) is enhanced for incoming EuroISDN calls
- Type Of Number (TON) and Numbering Plan Identification (NPI) included in CDR tickets, when the configuration record parameter CLID is set to YES and when the configuration record parameter FCDR is set to NEW. This allows incoming calls to be billed to different

accounts according to the originator of the call. The TON and NPI are required, since the CLID provides a sequence of digits that could be the same for a local, national or international number.

- EuroISDN interface also provides interworking with other ISDN or non-ISDN interfaces, such as existing ISDN Central Office connectivities (AXE10, 1TR6, SYS12, Numeris and Swiss Net 2), MCDN (1.5 and 2.0 Mbit/s), QSIG, DPNSS1, DASS2, R2MFC, MFE, DTI, DTI2 and analog trunk interfaces.
- layer 1 and layer 2 compliance with ETSI requirements are also supported

Operating parameters

The EuroISDN feature does not support non-Downloadable D-Channel cards (an NT6D80 MSDL or NTBK51AA/NTBK51CA Downloadable D-Channel card is required).

Operating parameters pertaining to Call Control Procedures:

- If more than one Channel Identification IE is received in a SETUP message, only the first one is used by the system. If it is not available, the call is processed according to the channel negotiation configuration. The call is released if no negotiation is allowed. The transfer of a unanswered EuroISDN call to a remote ringing phone requires disconnect supervision on the TIE trunk. If the disconnect supervision is not available, after the external user hangs up, the trunk could be locked out. It is the craftsman's responsibility to ensure that the trunks used for this type of call actually have disconnect supervision.

Operating parameters pertaining to ETS 300 102:

- The user-to-user compatibility checking, by the means of the Low Layer compatibility IE and/or the High Layer compatibility IE, is not supported. The LLC IE and the HLC IE are tandemed by the system, but this information is not used to perform any checking on a system node.
- Transit network selection is not supported. This IE is normally used by the user to identify a selected transit network in the SET-UP message. As no European country specifies the coding to use this IE, this service is not supported by the system. As a result, this IE is never sent by the system.
- Extension for symmetric call operation is not supported. This is normally used to implement a private network application.
- Network specific facility selection procedures are not supported.
- D-Channel backup procedures are not supported by EuroISDN.
- Message segmentation procedures are not supported. These are normally used to split messages that are too long.

- Low Layer Information coding principle is not checked by the system. No LLC is generated by the system, but this information is tandemed if received (from an ISDN BRI phone for example).
- Low layer compatibility negotiation procedures are not supported.
- The USER INFORMATION message is not implemented in the software.

Operating parameters pertaining to ETSI GF:

- ETSI GF gateways to and from other signaling systems, such as DPNSS1, QSIG, and MCDN, are not supported.
- The following ETSI GF procedures are not implemented:
  - control of supplementary services using the separate message approach (HOLD/ RETRIEVE).
  - bearer-related broadcast transport mechanism (multipoint configuration).
  - bearer-independent point-to-point connectionless transport mechanism.
  - bearer-independent broadcast transport mechanism.
  - generic notification procedures.
  - network-side channel reservation function.
  - generic procedures for supplementary service management.
  - generic status request procedure.
  - support of the Extended facility information element.
- DN address translation requires the association with a customer number. For an ETSI basic call establishment, the customer number association is found through the B-Channel identified in the channel ID IE. For DN address translation that is not associated with a basic call, the customer number association needs to be determined through other methods.
- For a BRI trunk DSL interface, there is a customer number association with the D-Channel. For a PRI interface, a prompt is added for an ETSI D-Channel configuration to create a customer number association with a given D-Channel. This implies that bearer independent messages on a primary rate D-Channel are associated with a single customer as configured in LD 17. For example, in a multi-customer configuration, if every customer on the switch wishes to use the bearer-independent transport service over ETSI PRI interfaces, then each customer requires a separate D-Channel.
- The Facility Information Element (FIE) is a repeatable IE, and its length is application dependent. However, due to system capacity considerations, such as call register usage and real time usage, there are two types of limitations enforced by the ETSI GF transport platform (actually by the ISDN GF transport platform provided by the ISDN QSIG GF development):
  - the system ETSI GF platform supports up to a maximum of eight ROSE components in one message. The eight components can be included in one Facility Information Element (FIE), or multiple FIEs. In addition, the inclusion of components in a

message is also limited by the "available message length". The available message length is the difference between the maximum message length (260 octets), and the maximum message length taken up by other mandatory and optional IEs supported in the given message.

- when a supplementary service requests the ISDN ETSI GF transport to send a component which exceed the available message length or the number of components supported, the supplementary service will be notified.

Operating parameters pertaining to ETS 300 403:

- The operating parameters pertaining to ETS 300 102 are applicable to ETS 300 403, except for the user-to-user compatibility checking, which is partially supported in some cases. For instance, as in the case of the EuroISDN 7 kHz/Videotelephony teleservices.
- Codeset 4, which is reserved for use by the ISO/IEC standards, has been added to the ETS 300 403 standard. This codeset is currently not used by the system. Therefore, it is not supported.
- The fall-back capability for multirate services is not supported.

Feature interactions

Incoming Digit Conversion

The IDC feature provides the availability to convert the incoming digits from a DID route. This feature is supported on the incoming EuroISDN DID routes. Digits received as a called party number are converted if the IDC feature is activated on the route. Digit analysis is then performed on the converted digits by the system.

Intercept Treatment on Invalid or Partial Dialing

There are three features that are involved in the treatment of calls with dialling irregularities. These are the Partial Dial Timing (PRDL), the Vacant Number Routing (VNR), and the Intercept features.

The Intercept feature allows calls with dialling irregularities to be routed to a Recorded Announcement, an attendant, or to receive overflow or busy tone. Separate treatments can be specified for DID and TIE trunks, for CDP/UDP calls, or for non CDP/ non UDP calls.

The PRDL feature allows to route non-ISDN DID incoming calls to the attendant after a configurable amount of time has expired, if the received digits cannot lead to the completion of the routing.

The VNR feature allows a call to a vacant number to be routed to another node, where the call is either treated as a vacant number, terminated, or given intercept treatment.

In order to provide as much flexibility as possible in the treatment of calls from a EuroISDN interface with dialing irregularities, the PRDL feature has been enhanced. Once the PRDL feature is activated, instead of automatically routing the call to an Attendant, the call will be given the intercept treatment provided by the VNR and Intercept features.

For the PRDL feature, when applied to EuroISDN trunks, the EOD timer used as the PRDL timer must be at least two seconds shorter than the ISDN interdigit timer (T302 = 15 seconds) used for the VNR and Intercept treatments. The call is rejected if T302 times out first. The VNR feature uses another timer which is the NIT timer. Its value is configurable between 3 - 8 seconds, for each customer.

Each time a digit is received, if the number received is an invalid number, the VNR, and then potentially the Intercept treatment feature, are activated. A number is known to be invalid under the following conditions:

- Enbloc receiving is configured and an incomplete or invalid number has been received.
- Overlap receiving is configured, and:
  - an invalid number has been received
  - an incomplete number with a Sending complete IE has been received
- If the number received is not completed and not yet invalid, one of the following timers is started:
  - EOD timer (if PRDL = YES)
  - NIT timer for UDP/CDP calls (If PRDL = NO)
  - T302 timer

Calls within a CDP or UDP Numbering plan

If an invalid number has been received, the PRDL feature is not involved as no timer has been started.

If an incomplete number has been received, and if PRDL is configured, the NIT timer has no effect. If the EOD timer times out before T302 timer, the PRDL feature is activated, otherwise the call is rejected.

Non CDP and Non UDP Calls

During enbloc receiving, if an invalid number is received in the SETUP message, the call is treated by the intercept feature.

During Overlap Receiving, if the SETUP message contains an incomplete/empty Called Party Number with the Sending complete IE, the call is treated by the intercept feature. If PRDL is configured, the EOD timer is started.

Transfer of an Unanswered Call

This improvement of the Call Transfer feature (for Italy only) allows an attendant, and an analog or a digital phone to transfer an outgoing EuroISDN unanswered call, after dialling has been completed. completion of dialling is indicated by the receipt of a CALL PROCEEDING or ALERT message. The Transfer can be performed before the internal phone (set to which the outgoing call is transferred) answers.

The transfer is allowed if NAS is configured on the MCDN link or if the link is QSIG with Slow Answer Recall configured (RTIM prompt in the CDB). If the trunk type of the EuroISDN link is neither a Central Office, DID, TIE, FEX, or WATTS, the completion of the transfer is possible only after the called party on the remote node has answered the call.

The transfer of an unanswered EuroISDN outgoing call interacts with the CDR feature. When the originating party completes the transfer, a CDR ticket type "S" (Start) is generated as soon as the transferred party answers. If one side releases the call, a second CDR ticket E is generated corresponding to the end of the communication.

If the originating party completes the transfer and both parties are still ringing, the CDR ticket type "S" (Start) will only be generated when the transferred party answers. If the external party answers when the transferred party is still ringing and then disconnects the call, the CDR ticket 'S 'and CDR ticket 'E' (End) are generated: in this case the orig ID is the transferring phone and the Ter ID is the outgoing trunk.

The Recovery of Misoperation during Call Transfer function is applicable as a normal operation.

For the Break-in feature, the same restrictions as for normal Call Transfer are apply. During the process of transferring a call, the attendant is not able to reach one of the phone involved in the transfer operation.

The changes introduced to enable the transfer before answer have an impact on the conference feature. With the introduction of transfer before answer, it is possible to add additional parties to a call before the first party called over a EuroISDN link has answered the call.

The transfer of unanswered call has no impact on the Call Forward No Answer feature.

The transfer of unanswered call has no impact on the Slow Answer Recall feature.

End-to-End signaling

End to End signaling is supported on all outgoing EuroISDN DID routes as soon as the CALL PROCEEDING message with a Progress Indicator is received.

Integrated Services Access

The Integrated Service Access (ISA) is not supported in Europe.

Advice Of Charge

AOC is implemented as part of the AOC for EuroISDN feature.

Call Completion Supplementary Service

Call Completion interacts with Call Transfer over EuroISDN. This interaction is similar to the current Network Ring Again/Network Ring Again on No Answer, and Call Back When Free/ Call back When Not in Use implementations. Call Completion Free Notification can only be presented to the Call Completion Originating station and cannot be transferred to any other station. Once the Call Completion call is established, it behaves as a normal call with respect to call transfer.

Call Back Queuing and Off-hook Queuing

This feature is not supported with Overlap signaling.

Flexible Hotline

This feature is not supported with Overlap signaling.

Auto Terminate Call

This feature is supported.

Special Dial Tone after Dialed Numbers

Special dial tones after dialed number feature is not supported for incoming calls.

Trunk Route Optimization

The purpose of this functionality is to provide trunk optimization before call establishment to incoming EuroISDN calls. The trunk optimization occurs when, on the MCDN network, the call has been redirected for one of the following reasons: Call Forward All Calls, Call Forward Busy

(CFB), Call Forward No Answer (CFNA), or Hunting. This function will be automatically triggered as soon as TRO is configured on the MCDN network.

If the call is not routed inside MCDN Network before T310 expiry (that is, an ALERT or a CONNECT message is sent back to the Central Office), the call will be released by the Central Office, with cause #102 "Recover on Timer Expiry". This situation will hardly occur since the optimization operation has to last more than T310 seconds (30 to 40 seconds).

Note:

When TRO is invoked on the redirecting node, a 2s timer is armed and if no response is received from the M1 Gateway node before expiry, TRO invocation is cancelled and the call is normally routed through the redirecting node.

Optional Sending of Forwarding CLID

The system is able, on a configuration basis, to send the last forwarding DN as CLID information for different scenarios of forwarded calls when the redirection occurs on a gateway node. The functionality is activated for the following redirections: Call Forward All Calls, Call Forward No Answer, Hunt and the two ACD redirections, Night Call Forward and Interflow.

When the redirection does not occur on the outgoing gateway node for calls coming from a private network (MCDN/QSIG) and routed to the public network through DID or Central office, the outgoing CLID is the redirecting number information included in the SETUP message received on the outgoing gateway node and coming from the private network.

This redirecting number information is the first redirecting DN on the last redirecting node.

When no redirecting IE is present, no manipulation is done and the original CLID is propagated on the DID link.

When the functionality is activated, everything occurs as if the redirecting phone had initiated the forwarded call. This means that information such as Presentation Indicator, Screening Indicator, Type Of Number and Numbering Plan are modified accordingly.

The Outgoing CLID (OCLI) prompt is defined in LD15, and can take three different values:

- NO. No manipulation is done on outgoing CLID for calls forwarded to UIPE DID or Central Office.
- EXT. The last forwarding DN is sent as CLID information for incoming UIPE DID or Central Office calls landing on gateway node and forwarded back to EuroISDN link.
- ALL. The scenario described above still applies. Moreover, last forwarding DN is sent as CLID for internal calls (MCDN/QSIG/set) redirected from gateway node to the DID or Central Office link. If the redirection does not occur on gateway node, redirecting information number is sent as CLID if present in the incoming SETUP message received on gateway node from private network.

Note:

An incoming DID/Central Office call, routed through a private ISDN network and then redirected back to the PSTN through the DID/Central Office link, is considered as an internal call. Hence, OCLI prompt has to be set to ALL for the outgoing CLID to be manipulated.

The redirecting information number sent over the MCDN link is not the Last Forwarding DN, but the first Redirecting Number on the last node where redirections occur. Therefore, the DN sent as CLID over the DID or Central Office link will be that DN which is not the Last Forwarding DN.

ISDN CLID Enhancement

The ISDN CLID enhancement interacts with the product improvement "Optional Sending of Forwarding CLID". The CLID entry programed against the prime DN key of the forwarding phone attached to the gateway node is used to build the outgoing CLID.

Calling Party Privacy Enhancement

The EuroISDN interface supports the Calling Party Privacy Enhancement (CPPE) feature.

The Calling Party Privacy Enhancement (CPPE) feature provides a route option to ignore the Calling Party Privacy Indicator for incoming calls received from all public Integrated Services Digital Network (ISDN) interfaces. When the Privacy Indicator Ignore (PII) prompt is YES in LD 16, the Calling Line Identification (CLID) Presentation Indicator and the Calling Party Name Display (CPND) Indicator override the blocking of the number and name details of the calling party. The name and number of the calling party appear on the receiving telephone.

The CPPE feature introduces a new route option AUXP for Auxiliary processor applications in LD 16 (RDB – Route Data Block). AUXP enhances the ability of the system to honor or ignore the Privacy Indicator for a Calling Party Privacy call, for each incoming route. If AUXP is 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, there will be no change to the CLID Presentation Indicator.

Incoming Trunk Programmable CLID

The "Optional Sending of Forwarding CLID" interacts with the feature "Incoming Trunk Programmable CLID". This feature provides the capability to assign a billing number for incoming trunk routes.

Even though "Incoming Trunk Programmable CLID" is configured to send a billing number (such as the ACOD of the MCDN/QSIG route) as the CLID information, the incoming redirecting information number, if known, is used for the outgoing CLID on the DID trunk.

Type of Number and Numbering Plan Identification in CDR tickets

The Type Of Number and Numbering Plan Identification fields are included in CDR tickets.

Numbering Plan Identification will be printed out on the third line of CDR at characters 44 and 45. Type Of Number will be printed out on the third line of CDR at character 47. Even if seven values are actually used by the system, as NPI information is coded on four bits in the Calling Party Number Information Element, it is advisable to reserve a two characters printout.

Both fields only appear if the system configuration parameter CLID is set to YES, if the system configuration parameter FCDR is set to NEW and if there is an incoming ISDN trunk involved in the call. This means that both NPI and TON information are taken from the incoming trunk.

For non ISDN trunks as well as for phones, two blank characters are output for the NPI field and one blank character is output for the TON field. Blank characters are also output if the system configuration parameter CLID is set to NO. If the system configuration parameter FCDR is set to OLD, the fields do not appear, whatever the value of CLID parameter.

Customer Controlled Routing

A call request for telephony 7 kHz or videotelephony teleservice terminating on a Control DN (same as an ACD DN) follows the same rules as for a Meridian Proprietary Phone, that is, fall-back occurs if allowed, otherwise the call is rejected.

Call Detail Recording

The "Optional Sending of Forwarding CLID" interacts with the CDR feature. The CLID field in the generated CDR ticket for the forwarded outgoing EuroISDN call is modified exactly the same way as the CLID information in the SETUP message.

Meridian Link

The CLID generated for DISA and incoming trunks is sent to the Meridian Link application. The prefixes 0/00 added to the display of the CLID are not sent to the Meridian Link.

Virtual Network Service

VNS provides ISDN features when no MCDN link is available between two system switches. This can be used when analog links are available, or when ISDN links that provide no supplementary services are used. The existing link is used as a B-Channel, and a separate D-Channel handles the MCDN signaling between the two end switches.

A EuroISDN link can be used as a B-Channel for the VNS feature.

Commonwealth of Independent States Phase II, MF Shuttle

The gateway between EuroISDN and the CIS MF Shuttle is supported.

Interworking with KD3

No interworking with KD3 is supported.

Advice Of Charge

Network Call Transfer, Network Call Redirection

Advice of Charge is not supported network wide. If a call is transferred to another system node, the charging information is stored against the Route meter. Relevant CDR S, X and E records are output on the TTY.

Call Completion on Busy Subscriber

The CCBS service is developed on EuroISDN interfaces as an ETSI service. The service is supported for all EuroISDN countries compliant with the ETSI CCBS protocol As a result, each country requires detailed study to insure that the CCBS service offered is compliant with the ETSI CCBS protocol.

EuroISDN Malicious Call Identification

Disconnection of the basic call

On reception of a DISCONNECT message for an incoming EuroISDN trunk which has MCID configured, clearing of the call is delayed until the MCTM timer expires or until the called user goes on-hook.

The MCTM timer will only be started depending on the class of service of the phone connected to the EuroISDN trunk. It can also be started if the MTND prompt is configured in the route data block in case of a MCDN to EuroISDN gateway. Otherwise, the connection will not be held.

In band timer (T306)

For an incoming EuroISDN call, a DISCONNECT message is sent from the network side with a Progress Indicator (PI) Number 8, so that the T306 timer should delay the sending of the release message. If for the same call the Malicious Call Trace Timer (MCTM in the Route Data Block, LD 16) is applicable, then it is the T306 timer that takes precedence in order to avoid any in-band information to be cut.

Note:

It is recommended that the MCTM timer be configured to a lower value than the T306 timer to avoid any timeout of the T306 timer on the network side.

EuroISDN 7 kHz/Videotelephony teleservices

Telephony 7 kHz and videotelephony teleservices are supported on ISDN BRI Line and EuroISDN interfaces. For any other interface, such as QSIG, MCDN, or DPNSS1, fall-back occurs if allowed. Otherwise, the call is rejected.

Feature packaging

The following packages are required for the EuroISDN ETS 300 403 Compliance Update feature:

- EuroISDN (EURO) package 261
- Multi-purpose Serial Data Link (MSDL) package 222
- International Primary Rate Access (IPRA) package 202
- ISDN Supplementary Features (ISDN INTL SUPP) package 161
- 2.0 Mbit Primary Rate Access (PRI2) package 154
- Integrated Services Digital Network (ISDN) package 145
- Digit Display (DDSP) 19

For ISDN Basic Rate Interface Trunking (BRIT), the following additional packages are required:

- ISDN Basic rate Access (BRI) 216
- ISDN Basic Rate Trunk Access (BRIT) package 233
- Meridian 1 XPE (XPE) package 203

For EuroISDN with Overlap signaling, the following additional packages are required:

- Overlap signaling (OVLP) package 184
- Flexible Numbering Plan (FNP) package 160
- Coordinated Dialing Plan (CDP) package 59, or, if the Uniform Dialing Plan is used, either Network Alternate Route Selection (NARS) package 58 or Basic Alternate Route Selection (BARS) package 57
- Network Class of Service (NCOS) package 32
- Basic Routing (BRTE) package 14

For EuroISDN to interwork with MFC, MFE, DID, and DASS2/DPNSS1 gateways, the following additional packages are required:

- New Format Call Detail Recording (FCDR) package 234
- Calling Line Identification in Call Detail Recording (CCDR) package 118
- Call Detail Recording on Teletype Terminal (CTY) package 5
- Call Detail Recording (CDR) package 4

Feature implementation

Tasksummary list

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

1. Table 57: LD16 - Configure a PRI2 loop for the ETS 300 403 interface. on page 139
2. Table 58: LD 17 - Configure a D-Channel for the ETS 300 403 interface. on page 139
3. Table 59: LD 16 - Configure a new route for the E403 interface (the same responses to the IFC and CNTY prompts, as were used in LD 17, must be entered). on page 141
4. Table 60: LD 14 - Configure the PRI2 trunks for the E403 interface. on page 144

Table 57: LD16 - Configure a PRI2 loop for the ETS 300 403 interface.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	CEQU	Make changes to Common Equipment parameters.
...		
- PRI2	0-159	PRI2 loop number.

Table 58: LD 17 - Configure a D-Channel for the ETS 300 403 interface.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	ADAN	Action Device and Number.
- ADAN	NEW DCH xx	Add a D-Channel on port 0-63.

Prompt	Response	Description
...		
- CTYP		
	MSDL DCHI	Multipurpose Serial Data Link or Downloadable D-Channel card.
- GRP	0-4	Network Group Number.
- DNUM	0-15	Device number for I/O ports.
- PORT	0-3	Port number on MSDL card.
- USR	PRI	D-channel for ISDN PRI only. Precede with X to remove.
- IFC	E403	EuroISDN Interface conforming to the ETS 300 403, or the extended ETS 300 102 version for country-specific interfaces. Refer to the Feature description on page 123.
- - CNTY		Enter country pertaining to the E403 interface. If either ETSI or NET are entered, the interface is fully compliant with ETS 300 403. See Feature description on page 123.
	ETSI NET	ETS 300 403 for the user side. ETS 300 403 for the network side.
		If any of the countries listed on the following page are entered, the interface functions with the extended ETS 300 102 capabilities. See Feature description on page 123.
	AUS	Austria.
	DEN	Denmark.
	FIN	Finland.
	GER	Germany.
	ITA	Italy.
	NOR	Norway.
	POR	Portugal.
	SWE	Sweden.
	EIR	Ireland.
	DUT	Holland.
	SWI	Switzerland.
	BEL	Belgium.
	ESP	Spain.
	UK	United Kingdom.
	FRA	France.
	CIS	Commonwealth of Independent States (Russia and the Ukraine).
...		
- RLS	xx	Software Release of the far-end switch.

Prompt	Response	Description
- RCAP	aaaa Xaaaa	Remote capabilities. Precede with an X to remove an existing configuration.
- OVLS	(NO) YES	(Do not) allow Overlap Sending.
- - OVLT	(0)-8	Duration of time, in seconds, that the sending side has to wait between INFO messages are sent. "0" means send immediately
- TIMR	(NO) YES	(Do not) change programmable timers. Only supported for interfaces supporting one of the following timers.
- - T310	(30)-100	Maximum time in seconds between an incoming CALL PROCEEDING message and the next incoming message.
- - INC_T306	0-(2)-30	Variable timer, in seconds, for received DISCONNECT message on incoming calls allowing in-band tone to be heard. The network will stop sending after this timer times out. The value is stored in two-second increments, which are rounded up.
- - OUT_T306	0-(30)	Variable timer, in seconds, for received DISCONNECT message on outgoing calls allowing in-band tone to be heard. The network will stop sending after this timer times out. The value is stored in two-second increments, which are rounded up.
- LAPD	(NO) YES	(Do not) change the parameters for the Link Access Protocol the D-Channel.

Table 59: LD 16 - Configure a new route for the E403 interface (the same responses to the IFC and CNTY prompts, as were used in LD 17, must be entered).

Prompt	Response	Description
REQ	NEW	Add new data.
TYPE	RDB	Route Data Block.
CUST	0-99	Customer number as defined in LD 15.
ROUT	0-511	Route number for Large System and CS 1000E system.
TKTP	TIE DID COT	Trunk type. TIE. Direct Inward Dialing. Central Office Trunk.
...		
DTRK	YES	Digital trunk route.
- DGTP	PRI2	2.0 MBit/s PRI digital trunk type.
...		

Prompt	Response	Description
ISDN	YES	Integrated Services Digital Network.
- MODE	PRA	ISDN PRI route.
...		
- IFC	E403	EuroISDN Interface conforming to the ETS 300 403, or the extended ETS 300 102 version for country-specific interfaces. Refer to Feature description on page 123.
- - CNTY		Enter country pertaining to the E403 interface. If either ETSI or NET are entered, the interface is fully compliant with ETS 300 403. See Feature description on page 123.
	ETSI NET	ETS 300 403 for the user side. ETS 300 403 for the network side. If any of the countries listed following are entered, the interface functions with the extended ETS 300 102 capabilities. See Feature description on page 123.
	AUS DEN FIN GER ITA NOR POR SWE EIR DUT SWI BEL ESP UK FRA CIS	Austria. Denmark. Finland. Germany. Italy. Norway. Portugal. Sweden. Ireland. Holland. Switzerland. Belgium. Spain. United Kingdom. France. Commonwealth of Independent States (Russia and the Ukraine).
ICOG	IAO OGT ICT	Incoming and Outgoing trunk. Outgoing trunk only. Incoming trunk only.
ACOD	xx..x	Access code for this trunk route.
- CLID	OPT0	Prefix = 0, for North American dialing plan. This is the default value for ESIG and ISIG interfaces.
	OPT1	Prefix = 1, for international PFXs in CLID, any numbering type supported. This is the default value for all EuroISDN interfaces.
	OPT2	Prefix = 2, for international PFXs in CLID, CCITT numbering type supported: UKWN, INTL, NPA, and NXX.

Prompt	Response	Description
		Default value for CO/DID routes for the Telecom New Zealand interface.
	OPT3	Prefix = 3, for international PFXs in CLID, only NXX number type supported. Default value for TIE routes for the Telecom New Zealand interface.
	OPT4	For international COs, if the call originates from a CO trunk type, add nothing. Otherwise, add PFX1 and PFX2. This is the default value for the Hong Kong, Singapore, and Thailand interfaces.
	OPT5	This is the same as OPT4, except it supports a maximum of 10 digits in the CLID. This is the default value for the Austrian interface.
- PROG		Progress Signal.
	NCHG	Send PROGRESS (default value for all interfaces, except Austria).
	MALE	Send ALERT after CALL PROCEEDING.
	MCON	Send CONNNECT after CALL PROCEEDING (this is the default value for the Austrian interface).
...		
- RCAP	aaaa Xaaaa	Remote capabilities. Precede with an X to remove an existing configuration.
...		
- CPFXS	(YES) NO	Customer-defined Prefixes option. This prompt is added as one of the sub-prompts of ISDN = YES. It is added at the end of the ISDN sub-prompts. If CPFXS = YES, when constructing the Calling or Connected Line Identification, the prefixes are retrieved from the Customer Data Block via the PFX1 and PFX2 prompts in LD 15, as is currently done. This is the default response. If CPFXS = NO, when constructing the Calling or Connected Line Identification, the prefixes are retrieved from the Route Data Block via the HNTN and HLCL prompts in LD 16.
- - HNTN	0-9999	Home National Number. This number is similar to the PFX1 number prompted in LD 15. It is added to this overlay so that this prefix can be configured on a route basis as required in some countries (e.g., Italy). As is the case with PFX1, the HNTN prefix can be from one-to-four digits long. This prompt is displayed only if CPFXS = NO.

Prompt	Response	Description
		If only a <CR> is entered, this prompt keeps its previous configuration. If no value was configured previously, no value will be configured. Enter X to delete the digits.
HLCL	0-9999	Home Location Number. This number is similar to the PFX2 number prompted in LD 15. It is added to this overlay so that this prefix can be configured on a route basis as required in some countries (e.g., Italy). As is the case with PFX2, the HLCL prefix can be from one-to-four digits long. This prompt is displayed only if CPFXS = NO. If only a <CR> is entered, this prompt keeps its previous configuration. If no value was configured previously, no value will be configured. Enter X to delete the digits.
ADDP	(NO) YES	If ADDP = NO, the Calling or Connected Party Number displayed is not modified. If ADDP = YES, the prefixes 0 (national) or 00 (international) are added to the Calling Party Number if the Type of Number (TON) is public.

Table 60: LD 14 - Configure the PRI2 trunks for the E403 interface.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	TIE COT DID	TIE trunk data block. Central Office Trunk data block. Direct Inward Dialing trunk data block. **Note:** Must match TKTP defined in LD 16.
TN		Terminal number
	l ch	Loop and channel for digital trunks, where: l = Previously defined PRI2 loop. ch = channel 1-24 for 1.5 Mb/s DTI/PRI or 1-30 for 2.0 Mb/s DTI/PRI
CUST	0-99	Customer number as defined in LD 15.
RTMB	0-511 1-4000	Route number for Large System and CS 1000E system.
TGAR	0 - (1) - 31	Trunk Group Access Restriction The default of 1 automatically blocks direct access.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 15: EuroISDN Malicious Call Identification

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

The EuroISDN Malicious Call Identification feature (MCID) allows the source of an incoming call to be identified and recorded by the network side (the Central Office) of a EuroISDN PRI/ BRI connection.

This request is initiated by the activation of the Malicious Call Trace feature (MCT) from analog (500/2500-type) phones using a SPRE code and a two-digit Flexible Feature MCT access code or from Meridian 1 Proprietary Phones and attendant consoles using the TRC feature key.

EuroISDN Malicious Call Identification (MCID) can be invoked during the active phase of a malicious call or after the active phase of a malicious call for a limited period of time.Figure 10: EuroISDN MCID invoked during the active phase of a malicious call on page 148 illustrates a situation where EuroISDN MCID is initiated during the active phase of a malicious call.

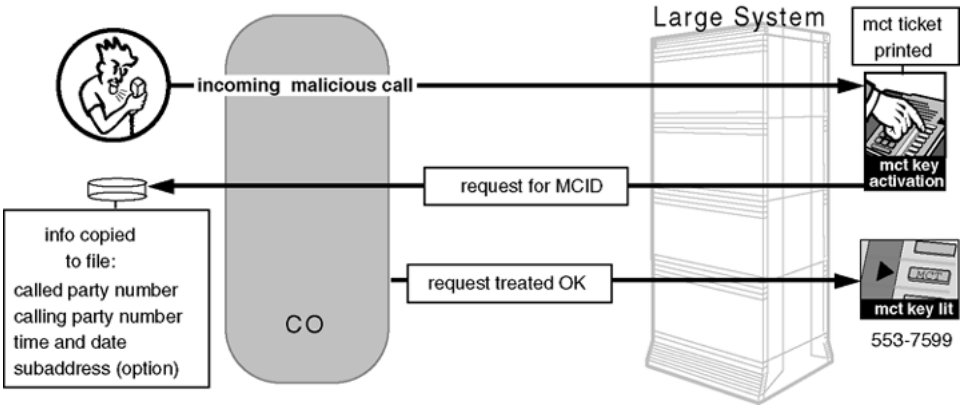


Figure 10: EuroISDN MCID invoked during the active phase of a malicious call

Figure 11: EuroISDN MCID invoked after the active phase of a malicious call on page 148 illustrates a situation where EuroISDN MCID is initiated after the active phase of a malicious call.

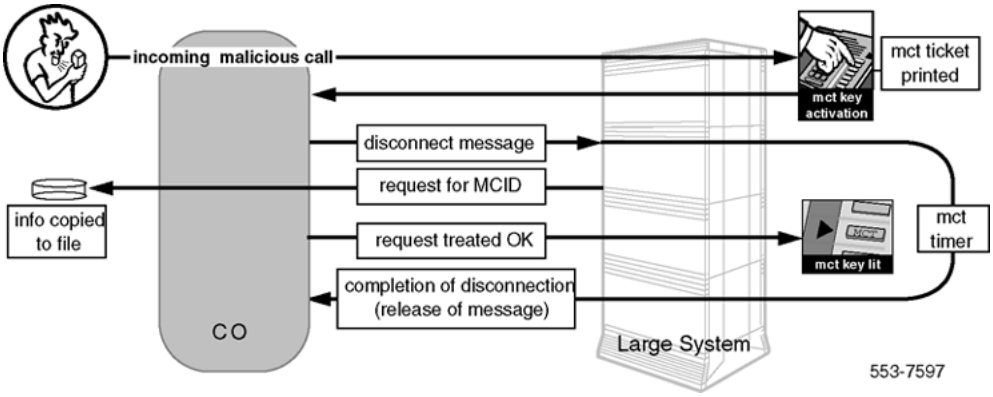


Figure 11: EuroISDN MCID invoked after the active phase of a malicious call

A report showing the results of MCID request is printed on any system Teletype Terminal (TTY) designated as a maintenance terminal. The report includes an enhanced Malicious Call Trace (MCT) ticket that shows the Number Plan Identifier (NPI) and the Type of Number (TON). A description of these output fields are provided in the section "Malicious Call Trace Record output".

The required system components of the MCID feature functionality are as follows:

- A Central Office switch supporting EuroISDN PRI/BRI connectivity (this CO is referred to as the network side).
- A system switch supporting EuroISDN PRI/BRI connectivity.
- For tandem applications, another system switch connected to the other over an MCDN link supporting the Malicious Call Trace feature and with Network Attendant Services enabled. In this case, in order for the MCID request to be processed, the gateway switch must be located between the MCDN link supporting MCT and with Network Attendant Service (NAS) enabled, and the EuroISDN Central Office interface.

When the MCID request is initiated through activation of the Malicious Call Trace feature, a MCID request message is sent to the Central Office from the called party node. The Central Office processes the MCID request and returns a message to indicate the successful operation. It registers the following relevant MCID information that is stored by the network operator:

- the called party's directory number
- the calling party's directory number
- the local time and date of invoking call trace on the network serving the called party
- the calling party's sub-address, if provided

Malicious Call Trace Record output

The NPI and TON fields have been added to the print output format, to show the Numbering Plan Identifier (NPI) and the Type of Number (TON) of the Calling Line Identifier field. The MCID ticket printed when the malicious call occurs. The output format is as follows:

```
MCI/MCE CUST# STAR1 ORIGTN STAR2 TERTN ATTPTY TIME CNI#
STAR1ORIGDN          STAR2TERDN          NPI TON
```

Note:

The second line is printed only if MCDC = YES in LD 15, the Customer Data Block. Also, for ISDN trunks, the ORIGDN/TERDN fields in line two are replaced by the CLID# field. Therefore, for ISDN trunks, the output would be as follows:

```
MCI/MCE CUST# STAR1 ORIGTN STAR2 TERTN ATTPTY TIME CNI#
STAR1CLID#           STAR2CLID#          NPI TON
```

where, for line 2:

Field	Description
NPI	For ISDN trunks. The Numbering Plan Identifier of the CLID printed above.
TON	For ISDN trunks. The Type of Number of the CLID printed above.

Operating parameters

The Malicious Call Trace (MCT) feature is only activated against one call at one time from either a phone or an attendant. If a phone activates MCT against a call, then the traced call is put on hold and a second call is activated. A trace against the second call cannot be completed until the first call is disconnected. If multiple TRC keys are defined, then this still occurs. The TRC key reflects the status of the MCT activation.

This feature is not supported on Basic Rate Interface (BRI) phones. However, it is supported over ISDN BRI trunks.

Soft keys cannot be defined as a TRC key on M2317 or M5317 digital phones.

For Large Systems, the EuroISDN Malicious Call Identification requires the following hardware:

- for Primary Rate Access, the PRI card or the NT5D97AD Dual DTI/PRI 2.0 Mbit/s card
- for D-Channel processing, NT6D80 the Multi-Serial Data Link (MSDL) card, or the NTBK51AA or NTBK51CA two-port Downloadable D-Channel Daughterboard (DDCH) with the NT5D97AD Dual DTI/PRI 2.0 Mbit/s card instead of an MSDL card
- for Basic Rate Access, the NT6D73AA MISP card and the NT6D70 SILC card for trunk access

For CS 1000E Systems, the EuroISDN Malicious Call Identification requires the following hardware:

- for Primary Rate Access, the NTBK50AA PRI card
- for D-Channel processing, the NTBK51BA Downloadable D-Channel Daughterboard
- for Basic Rate Access, the NTBK22XX MISP card and the NT6D70 SILC card for trunk access

When EuroISDN Malicious Call Identification is configured on a phone, the Attendant Administration feature cannot modify this feature.

Feature interactions

Call Detail Recording

If a DISCONNECT message is received from an incoming EuroISDN call and the Malicious Call Trace time is started, then this timer delays the output of the Call Detail Recording (CDR) record. This will not occur if the called party's phone goes on-hook.

Malicious Call Trace

EuroISDN Malicious Call Identification (MCID) is based on the existing Malicious Call Trace (MCT) feature. The existing MCT activation triggers MCID treatments.

Feature packaging

This feature requires the following packages:

- Malicious Call Trace (MCT) package 107
- International Supplementary Features (SUPP) package 131
- Flexible Feature Code (FFC) package 139 for analog (500/2500-type) phones
- Integrated Services Digital Network (ISDN) package 145
- Multipurpose Serial Data Link (MSDL) package 222
- Euro ISDN (EURO) package 261
- Universal ISDN Gateway (UIGW) package 283
- Euro Supplementary Service (ETSI_SS) package 323

Primary Rate Interface (PRI) connectivity requires the following packages:

- International Primary Rate Access CO (PRA) package 146
- 2.0 Mbit Primary Rate Interface (PRI2) package 154
- Integrated Services Digital Network Supplementary Features (ISDN INTL SUP) package 161
- International Primary Rate Access (IPRA) package 202

Basic Rate Interface (BRI) connectivity requires:

- Basic Rate Interface (BRI) package 216
- ISDN Basic Rate Interface Trunk Access (BRIT) package 233

Feature implementation

Task summary list

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

1. Table 61: LD 17 - Configure remote D-channel capability for EuroISDN interface for PRI and BRI. on page 152
2. Table 62: LD 16 - Configure remote capability for EuroISDN interface for PRI. on page 153
3. Table 63: LD 16 - Configure remote capability for EuroISDN interface for BRI. on page 153
4. Table 64: LD 15 - Modify system and software parameters. on page 153
5. Table 65: LD 10 - Configure analog (500/2500-type) phones. on page 154
6. Table 66: LD 11 - Configure Meridian 1 Proprietary Phones. on page 154
7. Table 67: LD 12 - Configure attendant consoles. on page 154
8. Table 68: LD 57 - Configure Flexible Feature Codes for analog (500/2500-type) phones. on page 154
9. Table 69: LD 16 - Configure Malicious Call Trace Timer and Tandem Delay. on page 155

Table 61: LD 17 - Configure remote D-channel capability for EuroISDN interface for PRI and BRI.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	ADAN	Action Device and Number.
- ADAN	CHG DCH xx	Change D-channel.
...		
- IFC	EURO	EuroISDN interface for D-channel.
...		
RCAP	MCID	Add Malicious Call Identification as a new remote capability.

Table 62: LD 16 - Configure remote capability for EuroISDN interface for PRI.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	RDB	Route Data Block.
...		
DTRK	YES	Digital Trunk Route.
- IFC	EURO	EuroISDN interface.
...		
RCAP	MCID	Add Malicious Call Identification as a new remote capability.

Table 63: LD 16 - Configure remote capability for EuroISDN interface for BRI.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	RDB	Route Data Block.
...		
DTRK	YES	Digital Trunk Route.
...		
- DGTP	BRI	Basic Rate Interface Digital Trunk Type.
- IFC	EURO	EuroISDN interface.
RCAP	MCID	Add Malicious Call Identification as a new remote capability.

Table 64: LD 15 - Modify system and software parameters.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	FTR	Customer Features and options.
...		
OPT	MCTA	Malicious Call Trace signal is allowed for attendants. MCTD = Malicious Call Trace is denied (default)
...		
MCDC	YES	Malicious Call DN/CLID printing allowed.

Table 65: LD 10 - Configure analog (500/2500-type) phones.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	500	Type of phone.
TN	l s c u	Terminal Number for Large System and CS 1000E.
...		
CLS	MCTA	Malicious Call Trace allowed. (MCTD) = Malicious Call Trace denied.

Table 66: LD 11 - Configure Meridian 1 Proprietary Phones.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	a...a	Telephone type. Type ? for a list of possible responses.
TN	l s c u	Terminal Number for Large System and CS 1000E.
...		
CLS	MCTA	Malicious Call Trace allowed. MCTD = Malicious Call Trace denied (default).
KEY	xx TRC	Trace key number.

Table 67: LD 12 - Configure attendant consoles.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	xxxx	Type of Attendant Console.
TN	l s c u	Terminal Number for Large System and CS 1000E.
...		
CLS	MCTA	Malicious Call Trace allowed. MCTD = Malicious Call Trace denied (default).
KEY	xx TRC	Trace key number.

Table 68: LD 57 - Configure Flexible Feature Codes for analog (500/2500-type) phones.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	FFC	Flexible Feature Code data block.

Prompt	Response	Description
CUST	0-99	Customer number as defined in LD 15.
CODE	MTRC	Malicious Call Trace code
- MTRC	xxxx	Enter Flexible Feature Code for Malicious Call Trace.

Note:

This configuration is only required to handle Malicious Call Trace during call disconnection.

Table 69: LD 16 - Configure Malicious Call Trace Timer and Tandem Delay.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	RDB	Route Data Block.
...		
MCTS	YES	Malicious Call Trace Signal. (NO) = default.
- MCTM	(0) - 30	Malicious Call Trace Request Timer ID in seconds.
- MTND	YES	Malicious Call Trace disconnect delay for tandem calls on AXE-10 Australia and EuroISDN.

Feature operation

No specific operating procedures are required to use this feature. However, the Malicious Call Identification feature is activated by one of the following methods, depending on the station:

- From an analog (500/2500-type) phone, a switch-hook flash is performed, and then a SPRE code and two-digit Flexible Feature MCT access code are dialed.
- From Meridian 1 Proprietary Phones, the TRC feature key is pressed.
- From an attendant console, the TRC feature key is pressed.

The following section describes the lamp status associated with the TRC key on Meridian 1 Proprietary Phones and attendant consoles.

Trace Number (TRC) key lamp status

The TRC key lamp status indicates the progress and success of the Malicious Call Identification request signaling to the CO. The following are the lamp states:

Lamp Winking

When the TRC key is activated, its associated lamp changes from dark to winking if the trunk involved in the call requires the signaling to be done. The lamp remains winking until the call identification request signaling to the CO has been completed.

In a Meridian Customer Defined Network (MCDN) tandem scenario, the lamp on the phone which originated the call identification remains winking until a Facility message is received from the node nearest to the Central Office. The user cannot invoke MCT again while the lamp is in the winking state.

Lamp Lit

The lamp state changes from winking to lit If the call identification request to the CO is successful.

In an MCDN tandem scenario, the lamp changes from winking to lit if a Facility message received from the node nearest to the CO indicates that the MCID request was successful. Any further activation of the TRC key during this state is ignored.

Lamp Dark

This lamp state indicates an idle TRC key or failure of the call identification request to the CO.

In an MCDN tandem scenario, the lamp goes from winking to dark if a Facility message received from the node nearest the CO indicates that the MCID request was unsuccessful.

During this lamp state, the TRC key can be activated to initiate the call identification request again.

Chapter 16: EuroISDN Trunk - Network Side

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

EuroISDN Trunk - Network Side refers to the behaviour of EuroISDN as a network interface (generally referred to as "network side"), where the EuroISDN interface has the behaviour of a terminal interface (generally referred to as the "user side").

As part of the EuroISDN Trunk – Network Side feature, the following capabilities are provided.

ETSI EuroISDN network side compliance to go along with the ETSI EuroISDN user side compliance. The ETSI EuroISDN network side interface (EuroISDN Trunk - Network Side)

provides the capability to connect the ETSI EuroISDN user side interface of a Common Peripheral Equipment (CPE) to the system. Another arrangement could be the connection of key systems to a system, to form part of the private network.

Similarly, this product also offers a trunk interface for the connection of terminal equipment, such as fax servers, routers and multiplexers, which would normally connect to the public network, but which now can be connected to the system to achieve a greater public network connection efficiency. Such access to the system can be through ISDN Primary Rate Interface (ISDN PRI) or ISDN Basic Rate Interface (ISDN BRI) trunks.

The network side interface provides all feature operations, interactions and gateways that are supported by the user side. It supports the following EuroISDN capabilities:

- Basic call
- Direct Inward Dialing
- Sub-addressing
- Calling Line Identification Presentation (CLIP)/Calling Line Identification Restriction (CLIR)
- Connected Line Identification Presentation (COLP)/Connected Line Identification Restriction (COLR)
- CCBS
    - User-to-User Signaling 1 (UUS1) information exchange transparency, which allows a calling and a called party to exchange small amounts of data over an ISDN PRI or ISDN BRI trunk's D-Channel. This data is contained in the USER_USER Information Element (IE) within the call control messages.
    - Bearer Capability-Based Routing, which allows outgoing calls to be selectively routed (over Central Office, Direct Inward Dialing, or TIE trunks for ISDN routes, and Integrated Digital Access trunks for DPNSS1 and DASS2 routes) based on its Bearer Capability. Any ISDN PRI or ISDN BRI route can be configured (in LD 16) to be dedicated to handle voice calls only, data calls only, voice with 3.1 KHz, data with 3.1 KHz, or both voice and data with 3.1 KHz.

The following protocols are supported by Bearer Capability-Based Routing:

- EuroISDN
- Asia Pacific
- Numeris
- 1TR6
- SWISSNET2
- AXE10 (for Sweden and Australia)
- SYS12
- D70

- QSIG
- DPNSS1
- DASS2
- MCDN
- All North American ISDN connectivities (system to system, system to SL-100, system to DMS-100/250,system to AT&T 4ESS, system to AT&T 4ESS/5ESS)

Operating parameters

The User-to-User transparency capability is only supported between EuroISDN trunks.

The following specific hardware is required on Large Systems.

For ISDN PRI access:

- NT8D72BA PRI2 card or the NT5D97AD Dual DTI/PRI 2.0Mbs card
- for D-Channel handling, the NT6D80 MSDL card or the NTBK51BA Downloadable D-Channel Daughterboard
- QPC414 Network Interface card, if the NT8D72BA PRI2 card
- NTRB53 Clock Controller card

For ISDN BRI access:

- NT6D73AA MISP card and the NT6D70BA SILC card for trunk access
- optionally, the NT6D71 UILC line card (the NT6D71 UILC line card requires ANSI 2B1Q line encoding and a Network Termination 1 (NT1) configuration)
- NTRB53 Clock Controller card

For ISDN BRI access:

- NTBK22XX MISP card and the NT6D70BA SILC card for trunk access

Feature interactions

Basic Call Gateways

The gateways that are supported for the EuroISDN Trunk - Network Side connectivity are the same as the ones supported for the EuroISDN connectivities,Table 70: Basic Call

Gateways on page 160 lists these gateways and the associated support for basic call functionality.

Table 70: Basic Call Gateways

Gateway	Basic Call Support
Analog (End-of-Signaling)	Basic Voice Call
DTI2 (End-of-Signaling)	Basic Voice Call Data Call (64K unrestricted)
MFE	Basic Voice Call
KD3	Not supported
R2MFC	Basic Voice Call Calling Line ID transmission (provided by the Tandem Call/CDR feature)
DPNSS1	Basic Voice Call Data Call (64K unrestricted) Calling Line ID transmission
ISDN COs (North American and non-EuroISDN interfaces)	Basic Voice Call Data Call (64K unrestricted) Calling Line ID transmission
EuroISDN and Asia Pacific ISDN	Basic Voice Call Data Call (64K unrestricted) Calling Line ID transmission
NI2	Basic Voice Call Data Call (64K unrestricted) Calling Line ID transmission
QSIG	Basic Voice Call Data Call (64K unrestricted) Calling Line ID transmission
MCDN	Basic Voice Call Data Call (64K unrestricted) Calling Line ID transmission
CIS	Basic Voice Call Calling Line ID transmission

Note:

For MFE, MFC, and CIS using DTI2, data calls are possible.

Call Completion to Busy Subscriber

This feature is supported on a EuroISDN Trunk - Network Side connectivity interface.

Call Completion on No Reply

This feature is supported on QSIG and DPNSS1 (as Call Back When Next Used) interfaces, corresponding to the MCDN Network Ring Again on No Answer feature. It is not supported on a EuroISDN Trunk - Network Side connectivity interface.

Calling Line Identification Presentation (CLIP)/Calling Line Identification Restriction (CLIR)

The EuroISDN Trunk - Network Side connectivity can generate, tandem or receive a Calling Line Identification (CLID) with presentation allowed (CLIP) or restricted (CLIR.) A CLID that is generated is constructed in the same manner as the EuroISDN user mode connectivity. A CLID that is received is displayed on the called user's display, if call presentation is allowed.

Even though the EuroISDN Trunk - Network Side connectivity acts as the network side of the Central Office connectivity, it does not provide the network functions (screening and validation) for the Calling Line Identification service.

Connected Line Identification Presentation (COLP)/Connected Line Identification Restriction (COLR)

The EuroISDN Trunk - Network Side connectivity can generate, tandem or receive a Connected Line Identification with presentation allowed (COLP) or restricted (COLR). A Connected Line Identification that is generated or tandemed is constructed in the same manner as the EuroISDN user mode connectivity. A Connected Line Identification that is received is displayed on the called user's display, if call presentation is allowed.

Even though the EuroISDN Trunk - Network Side connectivity acts as the network side of the Central Office connectivity, it does not provide the network functions (screening and validation) for the Connected Line Identification service.

Calling Party Privacy

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

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 all public Integrated Services Digital Network (ISDN) interfaces. When the Privacy Indicator Ignore (PII) prompt is YES in LD 16, the Calling Line Identification (CLID) Presentation Indicator and the Calling Party Name Display (CPND) Indicator override the blocking of the number and name details of the calling party. The name and number of the calling party appear on the receiving telephone.

The CPPE feature introduces a new route option AUXP for Auxiliary processor applications in LD 16 (RDB – Route Data Block). AUXP enhances the ability of the system to honor or ignore

the Privacy Indicator for a Calling Party Privacy call, for each incoming route. If AUXP is 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, there will be no change to the CLID Presentation Indicator.

End-to-End Signaling

End-to-End Signaling, which allows in-band dialing to be performed on ISDN trunks before and after the call has been answered, is supported on the EuroISDN Trunk - Network Side connectivity.

In the case of tandem with ISDN trunks, the necessary information to allow the End-to-End Signaling feature is tandemed to the ISDN trunk. At this point, it becomes the responsibility of the end user switch to provide the End-to-End Signaling service.

Incoming Digit Conversion (IDC) Enhancement

This feature is supported on the incoming EuroISDN Trunk - Network Side connectivity DID routes. If IDC is equipped, digits received as a called party number are converted, and digit analysis is then performed on the converted digits.

ISDN CLID Enhancements

The EuroISDN Trunk - Network Side connectivity supports all of the user side ISDN CLID enhancements.

Integrated Services Access (ISA)

ISA is not supported on a EuroISDN Trunk - Network Side connectivity. ISA is currently implemented for the North American ISDN interfaces. ISA allows a B-channel to be configured as a universal trunk. For example, the same B-channel trunk can be used once as a Central Office trunk and the next time as a Direct Inward Dial trunk.

Network Alternate Route Selection (NARS)/Basic Automatic Route Selection (BARS)/Coordinated Dialing Plan (CDP)

For NARS, the Numbering Plan Area (NPA) code and Central Office Code (NXX) cannot be used either on the ETSI network side interface or ETSI user side interface, since the codes are not supported by the European public network. Special Numbers (SPNs) are converted to "unknown", since SPNs are used only in North America. Also, when using Location Codes

(LOCs), the networking features do not accept a Digit Manipulation Index (DMI) used to insert an ESN access code. Therefore, the trunks are treated as though they were non-ISDN.

For BARS, the Numbering Plan Area (NPA) code and Central Office Code (NXX) cannot be used on the ETSI network side interface, since the codes are not supported by the European public network. Special Numbers (SPNs) are converted to "unknown", since SPNs are used only in North America.

For CDP, a CDP call can access a trunk on the EuroISDN Network Side. However, since neither the private number nor a CDP number is supported by the ETSI EuroISDN Trunk - Network Side, they get converted to a type that is supported by public network. This applies to both the called and calling number plan and type. Also, normal usage of steering codes with Distant Steering Codes (DSCs) and Trunk Steering Codes (TSCs) is supported, as is the use of Digit Manipulation Indexes (DMIs).

Network Attendant Service

Network Attendant Service signaling is not supported on a EuroISDN Trunk - Network Side connectivity. However, NAS will interwork with an incoming call from the EuroISDN Trunk - Network Side (routing and call handling).

Network Call Redirection (Call Forward, Call Forward No Answer, Hunt) and Network Call Modification (Conference, Transfer)

It is possible to have a phone Call Forward, Call Forward No Answer or Hunt to an external number over a EuroISDN Trunk - Network Side connectivity ISDN PRI or ISDN BRI trunk. It is also possible to transfer or conference a call to an external number over a EuroISDN Trunk - Network Side connectivity ISDN PRI or ISDN BRI trunk. Access restrictions can block some transfers from being completed.

Notices of call redirection or call modification are not transmitted over a EuroISDN Trunk - Network Side connectivity.

Network Call Party Name Display

This feature is not supported on a EuroISDN Trunk - Network Side connectivity.

Name Display

The transport of the name information, which is supported on QSIG interfaces and which corresponds to the Network Calling Party Name Display feature, is not supported on a EuroISDN Trunk - Network Side connectivity interface.

Networking Supplementary Features

The supplementary features that are supported on a EuroISDN Trunk - Network Side connectivity interface are Call Completion to Busy Subscriber, Calling Line Identification Presentation/Restriction, Connected Line Identification Presentation/Restriction and User-to-User Signaling 1 (UUS1) information exchange transparency.

Special Dial Tone After Dialed Numbers

This feature is not supported for incoming calls on the ETSI network side, but it is supported for outgoing calls.

Trunk Optimization

Trunk Optimization is not supported across a EuroISDN Trunk - Network Side interface. Trunk Optimization Before Answer is not supported within an MCDN network, if the call originated from a EuroISDN Trunk - Network Side connectivity interface.

Virtual Network Services

Virtual Network Services is supported on a EuroISDN Trunk - Network Side connectivity, meaning that a EuroISDN Trunk - Network Side trunk can be used as a VNS bearer trunk.

Feature packaging

To support the network side EuroISDN Trunk - Network Side connectivity, EuroISDN Trunk - Network Side feature (MASTER), package 309 is required.

The following software packages are required as dependencies to the MASTER package, or are required to support the full capabilities described in this feature module:

- 2.0 Mbit Primary Rate Access (PRI2) package 154
- Integrated Services Digital Network Supplementary Features (ISDNS) package 161
- International Primary Rate Access (IPRA) package 202
- Basic Rate Interface (BRI) package 216
- Multi-purpose Serial Data Link (MSDL) package 222
- Basic Rate Interface Trunk Access (BRIT) package 233
- EuroISDN (EURO) package 261

Feature implementation

Task summary list

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

For ISDN PRI access:

1 Table 71: LD 17 - Configure the D-Channel for the network side EuroISDN ETSI protocol. on page 165

2 Table 72: LD 16 - Configure the ISDN BRI link for the network side EuroISDN ETSI protocol. on page 166

For ISDN BRI access:

3 Table 73: LD 16 - For an ISDN or DPNSS1/DASS2 trunk, configure the route for the Bearer Capability-Based Routing call type. on page 166

For ISDN PRI access:

Note:

LD 17 – For the network side EuroISDN Trunk - Network Side connectivity, configure the D-Channel for the network side EuroISDN ETSI protocol.

Table 71: LD 17 - Configure the D-Channel for the network side EuroISDN ETSI protocol.

Prompt	Response	Description
REQ	CHG	Change the existing data.
TYPE	ADAN	Action device and number.
- ADAN	CHG DCH x	Change a specified D-Channel. x = 0-63.
- CTYP	MSDL	The card type to be used is the MSDL.
- IFC	EURO	Interface type. Enter EURO for EuroISDN.
- - CNTY	NET	Country pertaining to EuroISDN interface. Enter NET for network side.

For ISDN BRI access:

Table 72: LD 16 - Configure the ISDN BRI link for the network side EuroISDN ETSI protocol.

Prompt	Response	Description
REQ	CHG	Change the existing data.
TYPE	RDB	Route Data Block.
CUST	0-99	Customer number as defined in LD 15.
ROUT	0-511	Route number for Large System and CS 1000E system.
DTRK	YES	Digital trunk.
- IFC	EURO	Interface type. Enter EURO for EuroISDN.
- - CNTY	NET	Country pertaining to EuroISDN interface. Enter NET for network side.

Table 73: LD 16 - For an ISDN or DPNSS1/DASS2 trunk, configure the route for the Bearer Capability-Based Routing call type.

Prompt	Response	Description
REQ	CHG	Change the existing data.
TYPE	RDB	Route Data Block.
CUST	0-99	Customer number as defined in LD 15.
ROUT	0-511	Route number for Large System and CS 1000E system.
TKTP		Supported trunk type.
	CO DID TIE IDA	Central Office trunk. Direct Inward Dial trunk. TIE trunk. Integrated Digital Access trunk.
…		
DTRK	YES	Digital trunk.
…		
- DSEL	Data Selection.	
	(VOD) VCE DTA 3VCE 3DTA TDN	Route is for both voice and data. Route is voice only. Route is data only. Route is voice and 3.1 KHz. Route is data and 3.1 KHz. Transparent Data Network

Feature operation

No specific operating procedures are required to use this feature.

Chapter 17: Idle Extension Notification

Contents

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

Feature description

The Idle Extension Notification feature provides the attendant with the ability to be notified when a busy extension becomes idle. There are several applications for this feature, but the primary one is for an attendant who wants to leave a person an urgent message, but still does not want to break in on an existing call.

Operating parameters

The following limitations apply for the Idle Extension Notification feature:

- This feature is part of the SACP feature and is dependent on configuring the SACP key on the attendant console.
- The attendant can use Idle Extension Notification for extensions with ordinary HOT/MCN/ MCR/SCN/ SCR DNs. It does not work for ACD DNs or PLDNs.

- Only one Idle Extension Notification at a time can be requested for an extension.
- If a call is already camped on (or SACP camped on) to an extension, no Idle Extension Notification can be requested for that extension at the same time.
- The SACP customer option SACP = ALL or SNGL in the customer data block is not applicable for Idle Extension Notification. The Idle Extension Notification feature is always performed on a call basis.
- The Idle Extension Notification feature can only be activated on a first degree busy extension. Generally if Camp-On is not allowed on an extension Idle Extension Notification is not allowed for that extension.
- Before Idle Extension Notification is requested no indication is given to the attendant of the extension being first or second degree busy. The feature first/second degree busy indication (ABDA) does not give an indication for calls on the source side of the attendant.
- When an Idle Extension Notification request has recalled and is answered by the attendant, a destination DN cannot be dialed until there is an established call on the SRC side.
- No Camp-On/Periodic Camp-On tone is given to an extension that is supervised for Idle Extension Notification even if the customer's OPT is CTA and the extension's Class of Service is WTA.
- The feature Idle Extension Notification is implemented in MCDN (Meridian Customer Defined Network) ISDN networks. Wherever ISDN network environments are mentioned, MCDN ISDN networks are considered. The feature is not supported on DPNSS networks or through MCDN to DPNSS Gateway nodes.
- The Idle Extension Notification feature is not valid for the 2616 phone.
- If in an ISDN network environment there are nodes with the Idle Extension Notification feature active and nodes without the feature active, the feature will not work across the network. However it will work locally in the nodes where it is active. If an attempt to request for Idle Extension Notification on an extension situated in a node that does not have this feature active is made, no response from the node of the extension will be received and the SACP lamp stays dark. The attempt is canceled when the attendant presses the Release/Release SRC key. An error message (ERR5334) is printed in the node of the extension.

Feature interactions

The Idle Extension Notification feature is not supported on ACD DNs.

An Idle Extension Notification call cannot be placed on hold by the attendant. If the attendant attempts to place an Idle Extension Notification call on hold before the Release key is pressed, the Idle Extension Notification request is canceled and the call is disconnected. An answered

Idle Extension Notification recall cannot be placed on hold until after the SACP key is pressed to ring the extension.

When an attendant has requested an Idle Extension Notification, it can be canceled by then pressing the Position Busy key. This will also cancel all other Idle Extension Notification requests.

Idle Extension Notification and pre-dial Break-In are mutually exclusive. It is possible to press the Break-In key, the DN of a busy extension (to override call forward, for example), and then press the SACP key to request Idle Extension Notification. While the SACP lamp is lit, it is not possible to press the Break-In key to complete the Break-In. Conversely, if a Break-In conference is underway, it is not possible to activate Idle Extension Notification by pressing the SACP key.

If the SACP lamp is lit during the pre-dial Break-In sequence, the only available possibilities are: to press the Release key to request for Idle Extension Notification; to press the SACP key again to remove the request; or to press the Release SRC key to Release the source without requesting Idle Extension Notification. If the Idle Extension Notification request is removed by pressing the SACP key a second time (or due to any other limitation of Idle Extension Notification), Break-In is then possible.

It is not possible to do a Post-Dial Break-In on the source side of the attendant, therefore it is not possible to request for Idle Extension Notification in conjunction with Post-Dial Break-In. This means that it is not possible to a Break-In after an Idle Extension Notification recall is answered if the wanted extension becomes busy again.

When an extension that is being supervised for an Idle Extension Notification to the attendant becomes idle, it is kept busy from receiving any incoming calls. The lamp on the attendant console for that DN will display a busy status, according to the parameters of the Busy Lamp Field/Enhanced Busy Lamp Field feature.

It is not possible to request for Idle Extension Notification if the Busy Verify feature has been activated after the Busy Verify key is pressed.

When an extension that is being supervised for Idle Extension Notification becomes idle, it has the ability to make outgoing calls. If Call Forward All Calls or Intercept Call Forward are activated at the extension before the attendant presses the SACP key to ring that extension, the attendant's call will be forwarded to the Call Forward destination. The attendant display will show both the call forward DN as well as the original extension's DN.

If the Call Forward DN is busy, SACP can be activated towards the Call Forward DN, if all the requirements for allowing Idle Extension Notification are met by this DN.

If an extension has Call Waiting configured, it is not possible to request Idle Extension Notification. Call Waiting has precedence over the Idle Extension Notification feature.

When an extension is blocked for receiving calls due to the Idle Extension Notification feature, camp-on is not possible.

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

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

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

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

If the attendant dials a busy extension that has Hunting configured and where all the DNs in the hunt chain are busy, Idle Extension Notification can be requested towards the dialed extension.

ISDN BRI extensions always have the Call Waiting feature equipped, therefore Idle Extension Notification is not possible.

The Idle Extension Notification feature is available for ISDN Meridian Customer Defined Networking (MCDN) networks using BRI TIE trunks if Network Attendant Service is configured.

It is not possible to request for Idle Extension Notification towards an extension that has the Make Set Busy feature activated.

If Idle Extension Notification is requested for a Multiple Call Arrangement DN, the first extension with this DN that becomes idle will cause the recall. This extension will also be blocked from receiving calls.

An Idle Extension Notification recall will always recall to the same attendant, regardless of the configuration of the Recall To Same Attendant (RTSA) feature.

During the time that an extension is supervised or temporarily blocked from receiving calls due to the Idle Extension Notification feature, it is possible to activate Ring Again towards that extension. It is also possible to request for Idle Extension Notification on an extension that is supervised for Ring Again. When the extension becomes idle, the Idle Extension Notification will be served first.

Feature packaging

This feature requires the following packages:

- Attendant Break-in/Trunk Offer (BKI) package 127
- International Supplementary Features (SUPP) package 131
- Integrated Services Digital Network (ISDN) package 145
- Advanced ISDN Network Services (NTWK) package 148
- Network Attendant Service (NAS) package 159
- Semi-automatic Camp-on (SACP) package 181

Feature implementation

Task summary list

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

1. Table 74: LD 12 - Designate the Semi-automatic Camp-on key. on page 173
2. Table 75: LD 15 - Configure a new Incoming Call Indicator (ICI) key for customer data. on page 173
3. Table 76: LD 16 - Configure the Idle Extension Notification block timer. on page 174

Table 74: LD 12 - Designate the Semi-automatic Camp-on key.

Prompt	Response	Description
REQ	CHG	Change.
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	l s c u	Terminal Number for Large System and CS 1000E.
CUST	0-99	Customer number as defined in LD 15.
KEY	xx SACP	Key number, Semi-automatic Camp-on/Idle Extension Notification.

Table 75: LD 15 - Configure a new Incoming Call Indicator (ICI) key for customer data.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	ATT	Attendant Console options.
CUST	0-99	Customer number as defined in LD 15.
ICI	xx IEN	ICI number, Idle Extension Notification.

Table 76: LD 16 - Configure the Idle Extension Notification block timer.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	RDB	Route Data Block.
CUST	0-99	Customer number as defined in LD 15.
CNTL	YES	Changes to Controls or Timers.
- TIMR	IENB 2-(5)-10	Idle Extension Notification block timer, in minutes. The default value is five minutes. Both SACP and NAS packages must be equipped.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 18: Incoming Trunk Programmable Calling Line Identification

Contents

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

Feature description

The Incoming Trunk Programmable Calling Line Identification feature allows a billing number to be assigned to incoming trunk routes. The billing number length can be from one to 16 digits, and is only used when the incoming trunk terminates to a PRI trunk connected to a Public Exchange/Central Office. The billing number is inserted into the Calling Line Identification (CLID) field of the outgoing PRI/BRI trunk route.

An option is also provided that allows the billing number to have the "Presentation Allowed" field in the outgoing route CLID to be allowed or denied. This will allow or deny the display of the billing number when the call terminates on a phone off the Central Office.

In addition, incoming routes that support CLID can have the Billing Number replace the CLID for tandem calls only.

Operating parameters

This feature is not supported on the following outgoing interface types:

- SL-1 interfaces
- DPNSS1 interfaces

M911 trunks do not support the billing number.

If the route defined is both incoming and outgoing, the billing number will only be used for incoming calls to support outgoing Integrated Services Digital Network (ISDN) routes.

The billing number is not unique; the same number can be programmed for multiple routes.

The billing number is only inserted on tandem calls for supporting trunks to the Central Office. Therefore, the display of billing number on phones is not provided, and the CDR output will not show the billing number.

There is no control over how the Central Office uses the billing number. The presentation indicator is configurable to either allow or deny the displaying of the billing number. This is only used for the outgoing call to the Central Office, which means when the call terminates on a phone off the Central Office, the billing number will be displayed at that phone if the option is set (BDSP = YES).

Call redirection is not supported (for example, if the incoming call terminates on the system and is transferred, or call forwarded to the Central Office, the billing number is not passed on to the Central Office).

Feature interactions

R2MFC CNI/CDR Enhancements

Incoming Trunk Programmable CLID takes precedence over the R2MFC CNI/CDR Enhancements feature. If the outgoing ISDN trunk is allowed to send a billing number, the billing number is sent out as the CLID, not the CNI from the incoming trunk.

Feature packaging

The following package is required for Incoming Trunk Programmable Calling Line Identification:

- Integrated Services Digital Network (ISDN) package 145

For ISDN and networking, the following packages are required:

- Basic Routing (BRTE) package 14
- Network Class of Service (NCOS) package 32
- Basic Alternate Route Selection (BARS) package 57 or Network Alternate Route Selection (NARS) package 58 or Coordinated Dialing Plan (CDP) package 59

Feature implementation

Task summary list

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

1. Table 77: LD 16 - Configure the incoming route. on page 177
2. Table 78: LD 16 - Configure the outgoing route. on page 178

Table 77: LD 16 - Configure the incoming route.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	RDB	Route data block.
...		
BILN	(NO) YES	Billing number (is not) is required.
BLEN	1-(10)-16	Billing number length. The default is 10 digits.
BNUM	x...x	Billing number (1 to 16 digits).
BDSP	(NO) YES	(Do not) display Billing Number.

Table 78: LD 16 - Configure the outgoing route.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	RDB	Route data block.
...		
ISDN	YES	ISDN route must be YES.
...		
IFC	xxxx	Select the appropriate interface type for route (cannot be SL1 or IDA).
SBN	YES	(Do not) send billing number. Must be set to YES.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 19: Integrated Services Access

Contents

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

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

Integrated Services Access (ISA) is an ISDN PRI feature offered by the system for connecting to PSTN nodes such as DMS 100, DMS 250, 4ESS and 5ESS.

ISA allows multiple service routes to share the same common pool of B-channels. 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, and they can carry calls of different types. These call types can change for each call. In addition to taking on different trunk types,

the service routes can also be uniquely configured for other features, such as Dialed Number Identification Services (DNIS), Incoming Digit Conversion (IDC), or Auto-Terminate.

AT&T's Call-by-Call nodal service offering is similar to DMS-250 ISA. Instead of multiple call/trunk types over a single ISDN PRI channel, AT&T offers services from the 4ESS Inter Exchange Carrier (IEC) such as MEGACOM, SDN, and ACCUNET.

Establishing ISA routes

Basic PRI configuration must be performed before Integrated Services Access (ISA) is defined (see "ISDN product overview"). ISA routes and trunks must be defined to access ISA functionality or AT&T's Call-by-Call Service.

You cannot configure trunks for the service route that also accesses the ISA master route. With a single PRI (23B + D) configuration, all 23 B-channels can be configured for the ISA master route.

ISA routes cannot be accessed directly, so they must be blocked in the following programs, if applicable:

LD 19	Code Restriction
LD 27	Alternate Route Selection (ARS)
LD 86	Network Alternate Route Selection (NARS) and Basic Alternate Route Selection (BARS)
LD 93	Tenant Services

Accessing the ISA master route

The caller dials the access code of a service route to access an ISA master route. This access can also be achieved through digit manipulation by BARS or NARS or by means of direct trunk access (TARG) values in LD 10, LD 11, and LD 16. A route is configured as an ISA type through service change in LD 16. The ISA master route contains a list of B-channels over PRI that can be shared.

Different service route types such as TIE or WATS are created and associated with an ISA master route type. Service routes are not permitted to have Private trunks or B-channel members. To access call-by-call capability, the caller uses NARS/BARS or CDP. Automatic stepping from a service route to an ISA master route is performed and the ISA master route searches for an idle ISA channel. The caller is prevented from dialing the access code of an ISA master route directly.

These ISA configuration prompts are described in the following sections:

Network Services Facility option

This option is configured on the system to indicate whether it expects the Network Services Facility (NSF) Identification Element (IE) from the DMS. The NSF information dictates which service route the system uses to terminate an incoming call.

Connectivity to DMS provides Public, Private, INWATS, OUTWATS, TIE, and FX call types. When specifying COT or DID, the default is the Public call type. The call types determine the NSF values. (NSF = NO yields the Public call type.)

The values for the COT/DID service route (COTR), the WATS service route (WATR), and the TIE service route (TIER) are for incoming call termination only. Predefined NARS, BARS, or CDP allows multiple outgoing routes. The COTR can be a route number for a central office DID/DOD route. It is used for public calls. For incoming public calls, if COTR is a central office route, it is routed to the attendant. If it is a DID/DOD route, it translates the last "n" digits (as predefined in LD 15). For Private and TIE calls, the called number can be translated as defined for the service route.

With NSF = NO:

- COTR = all incoming DID/COT calls
- TIER = all incoming TIE calls

Only Public and TIE calls can be terminated. Because there is no Min/Max throttling, one service route can use all the trunks in the pool. If the NSF is sent from the central office, a master route can still be configured with NSF = NO and the NSF is ignored.

With NSF = YES:

- COTR = all incoming DID/COT calls
- WATR = incoming WATS calls without SID

Different TIE and FX calls can be terminated to multiple service routes. There can be multiple INWATS routes. These routes are distinguished by the unique SID defined in the service route. This number must be coordinated with the DMS for correct call termination.

Minimum and maximum counter function

The minimum and maximum counter (Min/Max) function allows customers more control and the ability to manage call patterns: you can define a minimum number of B-channels for a service route and you can define a maximum number of calls allowed for each service route.

The sum of all the Min values must be less than or equal to the number of B-channels in the master route. The Max values for each service route must be equal to or less than the number of B-channels in the pool less all Min values from other service routes.

The sum of all Min values for the service route cannot exceed the number of B-channels defined for the ISA master route.

When the NSF prompt is YES and the Max value is reached in a service route, the All Trunks Busy (ATB) counter is incremented. The ATB count is not incremented when NSF is NO.

The ATB value is provided during the printing of traffic reports. The ATB is pegged against individual service routes when the maximum number of B-channels reserved for that call type has been reached. The ATB counter of an ISA master route is also incremented when the last B-channel of the ISA master route is busied. This occurs whether the NSF prompt is YES or NO.

Service Identifier

In addition to the call type (route type), each call is identified by a service Identifier (SID), which is used by the system and the central office for routing. The SID prompt is available when the NSF prompt is set to YES. These service identifiers must be set up in coordination with the central office for call type delivery and routing purposes.

The Service Identifier allows a call type with multiple SIDs such as INWATS to go to prespecified Directory Numbers.

Before this feature, the SID defaults to the route number, and the value range was 0–127. With the introduction of this feature, the SID no longer has a default value. Also, the value range of SIDs is expanded to support 0–511.

ISA capability

Basic ISA functionality includes the following:

- Incoming Digit Conversion (IDC) is supported for DID call type only.
- All INWATS calls terminate to the attendant console.
- ISA supports two-way DID trunk types (as do all PRI connections to a central office).
- A total of 127 SID routes is allowed for all trunk types for each customer.
- Multiple routes can be defined.
- Calls are routed based on Service Identifier (SID) value.

System to DMS-100

WATS:

- Only one INWATS route can be defined.
- All INWATS calls terminate to the attendant console.
- OUTWATS services are not limited to a single outgoing route.

COT or DID:

- Only one incoming route can be defined: either COT or DID, but not both.
- DID trunk type supports two-way traffic (inbound and outbound).
- More than one outgoing route can be defined.
- DID operates like a two-way COT.

TIE:

- Multiple routes can be defined.
- They are routed according to the Service Identifier (SID) value.

Note:
In LD 16, if NSF = NO, then only one TIE route can be defined.

FX:

- Multiple routes can be defined.
- They are routed according to the Service Identifier (SID) value.

System to DMS-250

This includes basic ISA functionality as well as the following:

- Private call type is supported for DMS-250 interfaces to the system and is defined as TIE.

WATS:

- Only one INWATS route can be defined.
- All INWATS calls terminate to the attendant console.
- OUTWATS services are not limited to a single outgoing route.

TIE:

- Multiple routes can be defined.
- They are routed according to the Service Identifier (SID) value.

Note:

In LD 16, if NSF = NO, then only one TIE route can be defined.

FX:

- Multiple routes can be defined.
- They are routed according to the Service Identifier (SID) value.

COT or DID:

- Only one incoming route can be defined either COT or DID, but not both.
- DID trunk type supports two-way traffic (inbound and outbound).
- More than one outgoing route can be defined.
- DID operates like a two-way COT.

System to 4ESS

The system supports the following AT&T 4ESS nodal service types:

- ACCUNET, a T1 access service for both voice or data
- SDN, Software Defined Network, an AT&T virtual networking package
- MEGACOM 800, inbound domestic bundling
- MEGACOM, outbound domestic bundling
- WATB, banded WATS (the user can specify a band, such as Band 3).
- WATM (the user can specify wide area service exclusive to Band 1 through 9; for example, Band 5 means 0 to 5)
- LDS, Long Distance Service, including "World Connect" (outbound international)
- IWAT, regular 800 service
- I800, International 800, anything outside the U.S.

Enhancements

ISA has been enhanced to provide the following:

- Private, INWATS, FX, and TIE call types received from an ISA master route terminate to an individual phone with or without Incoming Digit Conversion (IDC) for the DMS and AT&T interfaces.
- ISA service routes allow an Auto-Terminate DN to be defined in the route data block. Calls can terminate to an ACD queue using Dialed Number Identification Services (DNIS) for the DMS and AT&T interfaces.

- Private call type is supported for the DMS interface for each route. (Specify Private in LD 16.)
- Service Identifier (SID) is supported with INWATS call types for the DMS interface.
- The Service Identifier (SID) value no longer defaults to route number. A unique value must be specified by the user.
- When printing the route data block for an ISA master route, all the associated service routes are printed after the new prompt, SVRT, in LD 21.

Operating parameters

Integrated Services Access (ISA) is an ISDN PRI feature offered by the system for connecting to PSTN nodes such as DMS 100, DMS 250, 4ESS and 5ESS. Do not use ISA to interface with another PBX node.

The ISA enhancements are based on the following features:

- Dialed Number Identification Services (DNIS)
- Incoming Digit Conversion (IDC) for Direct Inward Dialing (DID)

Characteristics of the ISA enhancements are as follows:

- The range of the Service Identifier (SID) for the system to DMS interface is from 0 to 511. Each SID received in the ISA trunks uniquely corresponds to a service route defined in LD 16. Also, each service route uniquely corresponds to a SID in the DMS machine type.
- The range of the Service Parameter for the system to ESS interfaces is from 0 to 9 for the parameterized WATS band service. WATR and COTR depend on the operation of service routes.
- For system to DMS interfaces, if there is no Network Specific Facility Information Element (NSF) (IE) received in the ISA master route, the system treats the call using the service route configuration defined for the COTR prompt in LD 16. If the trunk type of the COTR is COT, all the calls terminate to the system attendant. If the trunk type of the COTR is DID, each call terminates to its DID number. Similarly, if the trunk type of the WATR is WAT, all the calls terminate to the system attendant. If the trunk type of the WATR is DID, each call terminates to its DID number.
- For the system to ESS interfaces, if there is no NSF IE received in the ISA master route, system treats the call using the ISA service route configuration, which is defined as the ACCUNET service for the SRVC prompt in LD 16. If the trunk type of the ISA service route is Direct Inward Dialing (DID), each call terminates to its DID number.
- If an INWATS call or Public call received in the ISA master route is intended to be terminated to the system attendant, either IDC can be used to convert the incoming called number of the attendant DN, or the incoming called number must match the Listed Directory Number (LDN) in the Customer data block. You can also terminate one service

route for each ISA route by programming the service route number at the COTR and WATR prompts.

- For the system to DMS interfaces, the minimum and maximum counter (Min/Max) function is for each service route. For example, route 40 (call type of OUTWATS with SID 9) or route 45 (call type of OUTWATS with SID 13) can each reserve its own minimum B-channels and limit its maximum B-channels to be used in the ISA pools.
- Since AT&T is limited to one service type for each service route in a given ISA master route, for the system to ESS interfaces, the Min/Max counter function is on for each service route. That is, AT&T supports only one Min/Max function for the service types in a given ISA master route, not a Min/Max for each band. Configure only one set of the minimum and the maximum B-channels in LD 16 for all the bands of the WATB service.
- For AT&T interfaces, only nine possible services can be used for each ISA master route. With DMS products, there are 512 possibilities, one for each SID value.

Table 79: Example of Min/Max function for system to DMS on page 186 illustrates the Min/Max function, assuming 23 ISA B-channels.

Table 79: Example of Min/Max function for system to DMS

Call Type	system to DMS interface (WATR = 30, COTR = 10, NSF = YES)	MIN	MAX
Public	(rout 10, tkty DID, icog IAO)	10	13
Private	(rout 20, tkty TIE, sid 22, facy PRV, icog IAO)	1	3
INWATS	(rout 30, tkty WAT, icog ICT)	1	3
INWATS	(rout 35, tkty WAT, sid 30, icog ICT)	1	1
OUTWATS	(rout 40, tkty WAT, sid 9, icog OGT)	1	2
OUTWATS	(rout 45, tkty WAT, sid 13, icog IAO)	1	1
FX	(rout 60, tktp TIE, sid 60, facy TIE, icog IAO)	1	4
TIE	(rout 60, tkty TIE, sid 60, facy TIE, icog IAO)	2	3
TIE	(rout 65, tkty TIE, sid 65, facy TIE, icog IAO)	2	3
	Total	20	

Note:

See the Min/Max rules contained in this module.

Note:

Service types are dominant over call types with AT&T interfaces.

Table 80: Example of Min/Max function for system to ESS on page 187 illustrates the Min/Max function, assuming 23 ISA B-channels.

Table 80: Example of Min/Max function for system to ESS

Service	system to ESS interface	MIN	MAX
ACCunet	(rout 10, tkty DID, icog IAO)	10	15
SDN	(rout 20, tkty TIE, icog IAO)	1	3
IWAT	(rout 30, tkty WAT, icog ICT)	1	3
WATB	(rout 40, tkty WAT, sprm 9, icog OGT)	1	2
WATM	(rout 45, tkty WAT, icog OGT)	1	2
LDS	(rout 70, tkty TIE, icog IAO)	2	3
I800	(rout 80, tkty TIE, icog IAO)	2	3
	Total	18	

Note:
See the Min/Max rules contained in this module.

Feature interactions

Enbloc dialing

Enbloc dialing in ISDN PRI performs dialed digit analysis to determine the end of dialing before constructing the outgoing SETUP message. The Trunk Access Code applies to the service route, not to the ISA master route.

nB+D Primary Rate Interface

The system allows 16 DS1 loops controlled by one primary D-channel (on channel 24 of the first DS1 loop) and one backup D-channel (on channel 24 of the second DS1 loop). ISA can use the 382 B-channels as the ISA pool of trunks.

4ESS SDN Access

Software Defined Network (SDN) is a service offered by AT&T 4ESS for the virtual private network. SDN can be offered on dedicated channels or ISA channels.

Incoming Digit Conversion (IDC)

Part or all of the digits received in the DID route are converted and terminated at the existing internal DN in the system. With this enhancement, IDC applies to the incoming WATS route over the ISDN interface.

Dialed Number Identification Services (DNIS)

DNIS provides the last three or four digits of the dialed DN from DID or TIE trunk to be displayed in the Meridian 1 phone digit display when the DNIS call terminates on an ACD DN. DNIS length flexibility, allows 1–7 digits. DNIS is supported for each route when using the ISA feature. DNIS also applies to the INWATS service route over the ISDN interface.

Feature packaging

This feature requires the following packages:

- NI-2 Call By Call Service Selection (CBC) package 117, which requires as dependencies:
  - Integrated Services Digital Network (ISDN) package 145
  - Primary Rate Access (PRA) package 146
  - Inter Exchange Carrier (IEC) package 149
- Integrated Services Digital Network (ISDN) package 145
- Primary Rate Access (PRA) package 146, which requires as dependencies:
  - Digit Display (DDSP) package 19 (for CLID)
  - 1.5 Digital Trunk Interface (PBXI) package 75
  - Integrated Services Digital Network (ISDN) package 145
- NI-2 Call By Call (NI2 CBC) package 334

Feature implementation

Task summary list

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

1. Table 81: LD 16 - Create an ISA master route. on page 189
2. Table 82: LD 14 - Configure ISA trunks. on page 190
3. Table 83: LD 16 - Configure ISA service routes. on page 191

Basic PRI configuration must be performed before Integrated Services Access (ISA) is defined. ISA routes and trunks must be defined to access ISA functionality or AT&T's Call-by-Call Service.

Table 81: LD 16 - Create an ISA master route.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	RDB	Route Data Block
CUST	0-99	Customer number as defined in LD 15.
ROUT	0-511	Route number or Large System and CS 1000E system.
TKTP	ISA	Create an ISA trunk route **Note:** Package 117, Call-By-Call Service, is required to define an ISA master route.
DTRK	YES	Digital trunk route
DGTP	PRI	Digital trunk type
ISDN	YES	ISDN option
MODE	PRA	D-channel mode controlling the route (ISA master route)
PNI	1–32700	Customer's Private ID
IFC	aaaa	Interface type: (D100), ESS4, ESS5, SL1, S100, D250

Prompt	Response	Description
		Note: The IFC of an ISA master route and its associated service route must match.
NSF	(NO) YES	Network Service Facility (for TKTP = ISA and non-AT&T IFC; AT&T always has NSF)
COTR	xx	DID/central office route number Prompted if IFC = D100, SL1, S100, or D250. Enter CO or DID route.
TIER	xx	TIE route number Prompted if IFC = D100, SL1, S100, or D250 and NSF = NO
WATR	xx	WATS route number Prompted if IFC = D100, SL1, S100, or D250 and NSF = YES
ICOG	IAO ICT OGT	Incoming and outgoing trunk Incoming trunk Outgoing trunk
SRCH	RRB	Round robin hunting for outgoing trunks
ACOD	xxxx	Trunk route access code

Table 82: LD 14 - Configure ISA trunks.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	ISA	ISA trunk type
TN		Terminal number
	l ch	Loop and channel for digital trunks, where: l = PRI2 loop as defined in LD 17 ch = channel 1-24 for 1.5 Mb/s DTI/PRI or 1-30 for 2.0 Mb/s DTI/PRI
CUST	0-99	Customer number as defined in LD 15.
RTMB		Route number and Member Number
	0-511 1-4000	Range for Large System and CS 1000E system.
TGAR	(0)–31	Trunk Group Access Restriction
CLS	aaa	Class of service options

Table 83: LD 16 - Configure ISA service routes.

Prompt	Response	Description
REQ	NEW	Add an ISA service route
TYPE	RDB	Route data block
CUST	0-99	Customer number as defined in LD 15.
ROUT	0-511	Route number for Large System and CS 1000E system.
TKTP	aaaa	Types of service route allowed with ISDN
DTRK	YES	Digital trunk route
DGTP	PRI	Digital trunk type
ISDN	YES	ISDN option
- MODE	PRA	PRA master route for ISA
IFC	aaaa	Interface type: (D100), ESS4, ESS5, SL1, S100, D250 **Note:** The IFC of an ISA master route and its associated service route must match.
SRVC	(ACC)	Service for AT&T ESS connections: NNSF, ACC, I800, LDS, M800, MEG, SDN, IWAT, WATM, WATB Prompted if IFC = ESS4 or ESS5 **Note:** NSF refers to the services provided on a Call-by-Call basis.
- SRPM	0–(9)	WATS band. Prompted only if SRVC = WATB.
ISAR	YES (No)	ISA service route
- RTN	xx	Select route number of any configured ISA master route Prompted only when ISAR = YES.
- FACY	YES (NO)	Facility for the call type (only applies if the far-end will be sending an NSF information element, NSF = YES) Prompted when TKTP = TIE, ISAR = YES, and IFC = D100/ D250/S100 TIE connection in the NSF IE Private connection in the NSF IE
- SID	xx	Service Route Identifier number. Prompted if NSF = YES and TKTP = TIE/WAT/FX.
- MIN	xx	Minimum number of channels reserved on the ISA master route
- MAX	xx	Maximum number of channels reserved on the ISA master route

Prompt	Response	Description
- PRIM	YES (NO)	ISA route class of service (not prompted after X11 R15
ICOG	IAO ICT OGT	Incoming and outgoing trunk Incoming trunk Outgoing trunk
ACOD	xxxx	Trunk route access code

Note:

The Listed Directory Number (LDN0) value must be configured in LD 15, Customer data block, in order to terminate ISDN/DID incoming calls such as ISA.

Feature testing

Test each PRI trunk to ensure that the system completes an ISA call. Run the following test for each of the available call types: TIE, FX, COT, INWATS, OUTWATS, and DID.

Note:

COT and DID are recognized as Public call types and they must be on the same route. They can be used for incoming or outgoing routes.

To test ISA, follow steps 1 through 8.

1. Select a PRI to be tested.
2. Access LD 60. Use STAT command on all pertinent B-channels.
3. Set all the Min/Max values in all service routes to 1.
4. Make a test call on a trunk route that accesses the PRI being tested. Be sure the test call is to a valid destination.
5. Use LD 80 to ensure a B-channel from the ISA pool is in use. Then:
   - Place the call on hold.
   - Make another call. You should receive an overflow signal (fast busy).
   - Repeat this procedure for every outgoing service route.
6. Disconnect both calls.
7. Ask the terminating end to place an incoming call for each incoming call type. Then:
   - Place the call on hold.
   - Ask the terminating end to place another call. They should receive an overflow signal (fast busy).
8. Disconnect both calls.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 20: Integrated Trunk Access

Contents

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

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

Integrated Trunk Access (ITA) allows common digital transmission facilities (such as a T1 link) to be shared by:

- B-channel trunks (ISL/PRI)
- Traditional A&B bit signaling trunks

This feature removes some trunks from PRI exclusive use, to share between PRI and traditional A&B signaling. ITA supports the following configurations:

- system to system
- system to SL-100
- system to DMS-100
- system to DMS-250

Note:

Channels must match for both signaling and routing.

Note:

A separate Route Data Block (LD 16) must be configured for A/B-channels. While PRI is operational, create an RDB for the A/B-channels. Then remove the PRI trunks in LD 14. Create new trunks with the same TNs for the new route.

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 21: ISDN Calling Line Identification

Contents

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

Feature description

The Calling Line Identification (CLID) feature provides the flexibility to build and send CLID on ISDN public and private interfaces based on various numbering plans.

CLID is table-driven (when LD 15 is loaded, a customer can configure a CLID table), and virtually any of the information contained in the fields of the CLID table can be programmed against any DN or DN key for each phone. This means that the CLID that is sent from a phone is predicated on what is in the CLID table.

A CLID for any key is built by taking the information contained in a particular field in the CLID table and adding that information to the key's DN. A multi-line phone can have DN keys that each has its own CLID. Or, as explained in How CLID entries are assigned to a phone on page 201, the CLID of any one key could, on a phone, be programmed to use the CLID of any other key on the phone.

Also, the system can support multiple office codes, location codes and steering codes in CLID (this is more fully explained in the section "How a CLID table is built", which follows.) This means that any phone on one system can send a CLID that will have calls returned to another system. This type of configuration would typically be used in cases where a customer would want calls to be returned to only one central location.

Note:

Since the system does not perform verification of CLID entries that are defined in a CLID table, it is the responsibility of the system administrator to ensure that DN keys are programmed correctly.

How a CLID table is built

LD 15 is used to create a CLID table for a customer. This table contains CLID 'entries' (up to 4,000.) Each entry contains unique information pertaining to CLID, as explained in the following sections.

North American Numbering Plan

For users of a North American Numbering Plan, the system can now support multiple Home Central Office Codes (HNXXs), Home Numbering Plan Area (HNPA) codes, Home Location Codes (HLOCs) and Local Steering Codes (LSCs), on a DN or DN key basis.

For a North American Numbering Plan, each CLID entry can contain the following:

- a one-six digit national code for a home national number (HNTN)
- a one-12 digit local code for a home local number (HLCL), or a one-12 digit Listed Directory Number for a switchboard
- a one-seven digit Home Location Code (HLOC)
- a one-seven digit Local Steering Code (LSC)

International Numbering Plan

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

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

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

DIDN (which signifies "use DN as a DID number") in LD 15, allows the HLCL to be built either using the digits in the HLCL plus the digits of the active key (if DIDN is set to "YES" - the DN is considered to be a DID number and is included in the CLID), or only the digits in the HLCL

(if DIDN is set to "NO" - the DN is not included in the CLID since it is not a DID number), or based on a search on the DN keys, beginning from key 0, to find the CLID to be used (DIDN is set to "SRCH".) Please refer to Examples one to six in Examples of CLID generation on page 203.

How CLID entries are assigned to a phone

Once the CLID table has been built for a customer, any CLID entry can be assigned to any phone, for each DN or DN key (any DN or DN key can be programmed against the information defined for any CLID entry in the CLID table.) This means that for a multi-line phone, DN key 0 can send one CLID (the information contained in one CLID entry), DN key 1 can send a different CLID (the information contained in another CLID entry), and DN key 2 can send yet another CLID (the information contained in a third CLID entry). A customer can now send different CLIDs from the same phone.

Digital phones

When configuring the CLID for a DN key in LD 11, the value entered against the KEY prompt can be a CLID table entry number (0-3999), which corresponds to a CLID entry in the CLID table or 'D'. If 'D' is entered, the system initiates a search on the phone for a DN key, starting from key 0. The first found CLID is then sent as the CLID of the active DN key. This means that a call can be made on one key, and the CLID of another key is sent. This configuration is typically used in an ACD or Hotline DN key application, where, for example, the CLID for a particular key is not desired to be sent. Please refer to Examples seven, eight and nine in Examples of CLID generation on page 203.

Information display on phones and terminals

The ISDN features listed in this section can be used on all Avaya phones. However, not all phones or terminals provide information displays.

Displayphone — The Meridian 1 Displayphone allows the full display of CLID information.

Other phones and terminals — The following devices, if equipped with a digit display, can receive and display a CLID name:

- Attendant Consoles
- Meridian 1 phones with digit display
- Digital phones (M2317)
- ASCII terminals with an Add-on Data Module (ADM)
- M2008, M2016S, M2216ACD, M2616 when equipped with displays
- Virtual phones that have PCA feature

Digit display format

As shown in Figure 12: Digit display format for Calling Line Identification on page 203, the digit display at the phone receiving the call will display:

- If the CLID display is unavailable because the call was not routed on ISDN routes for the entire call, the trunk route access code and trunk route member number are displayed.
- For public networks, CLID displays the standard North American Numbering Plan 7- or 10-digit number, depending on the number dialed.
- For a private network over ESN, the CLID displays an "H" followed by xxxx yyyyy, where xxxx = a four digit Home Location Code (HLOC) and yyyy = a four digit DN.
- For a private network over CDP, the CLID displays a four digit LSC (Local Steering Code) followed by one of the following:
  - the extension's trailing digits (forming the CDP DN) when CDP is equipped
  - the calling phone's extension

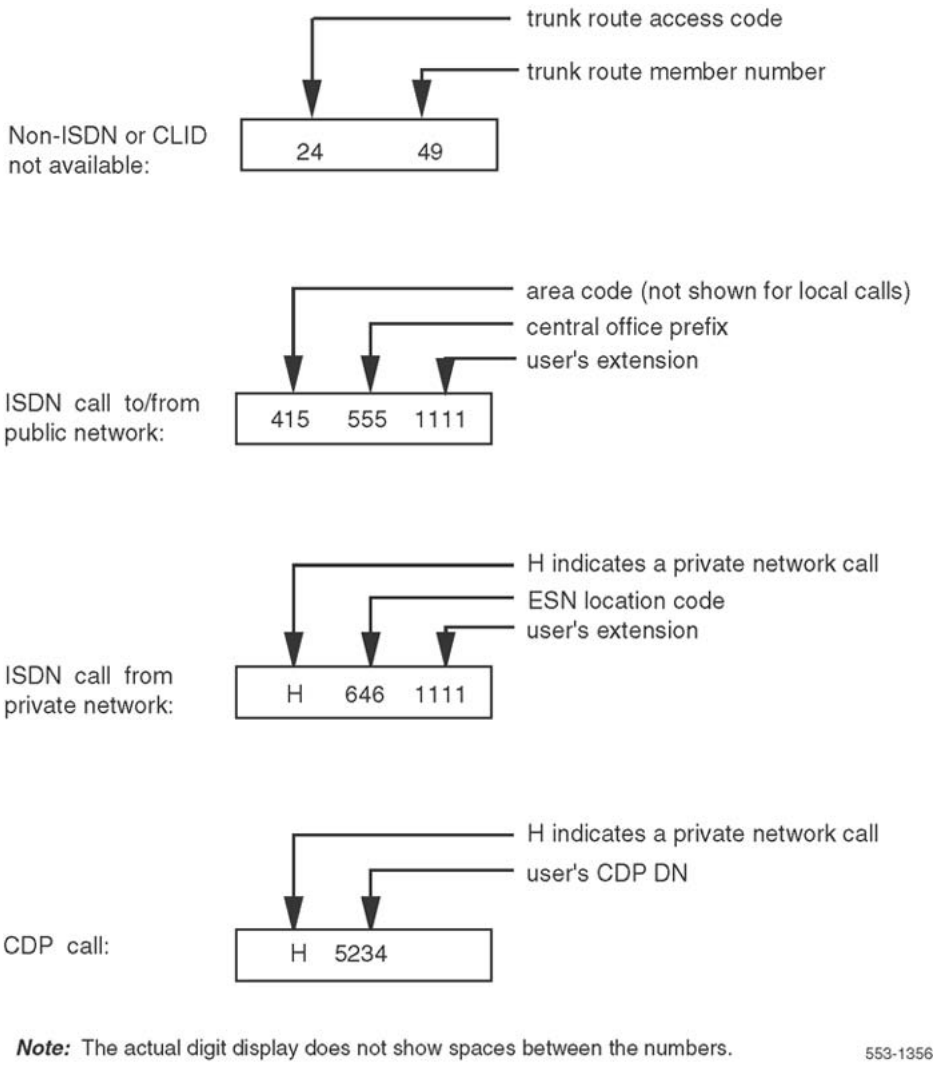


Figure 12: Digit display format for Calling Line Identification

Examples of CLID generation

The examples that follow show CLID generation using different combinations of CLID entries for DN keys, for Private ESN UDP/CDP calls and for Public Local and Public National calls.

Most of the examples use a North American Numbering Plan, since the construction of CLID entries in a CLID is the same for both a North American Numbering Plan and an International Numbering Plan. In some cases, however, examples are provided that pertain to calls using an International Numbering Plan.

Therefore, unless otherwise stated in the headings, the examples provided in the following sections pertain to calls made using a North American Numbering Plan.

The CLID construction for a CLID call type is as follows:

- Private ESN UDP Number - HLOC + DN
- Private ESN CDP Number - LSC + DN
- Public Local Number - HLCL + DN
- Public National Number - HNTN + HLCL + DN

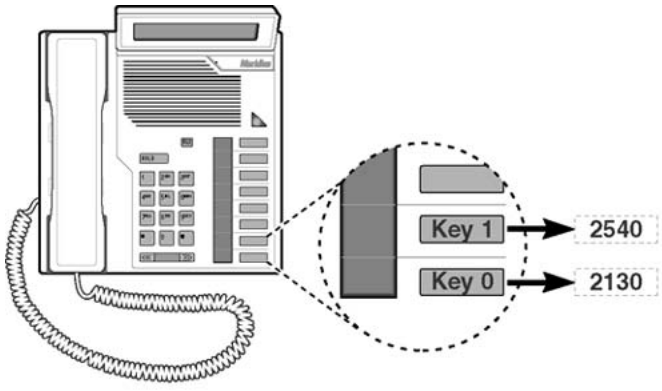
Please note that for calls made using the International Numbering Plan, the CLID for a Public Number is a National Number. For the examples of using the International Numbering Plan, the construction of a Public Local Number is for reference purposes only.

Example 1 - Phone has DID number on DN key 0

In this example, the calls are made on DN key 0, which has a DID number. Key 0 is configured against CLID table entry 0, and key 1 is configured against CLID table entry 1.

Table 84: Set has DID number on DN key 0

CLID entry	HNTN (code for the home national number)	HLCL (code for the home local number), or LDN	DIDN (use the DN as the DID?)	HLOC (home location code)	LSC (local steering code)
Entry 0	415	940	YES	646	5
Entry 1	415	9699170	NO	646	5
Entry 2			SRCH	646	5



553-7655.EPS

1. For a private ESN UDP call made from Key 0 (DN 2130) on the phone, the CLID would be built as follows:

   - HLOC from CLID entry 0 (646) plus the DN of the active key (2130).

- That is, the CLID that is sent would be 646 2130.

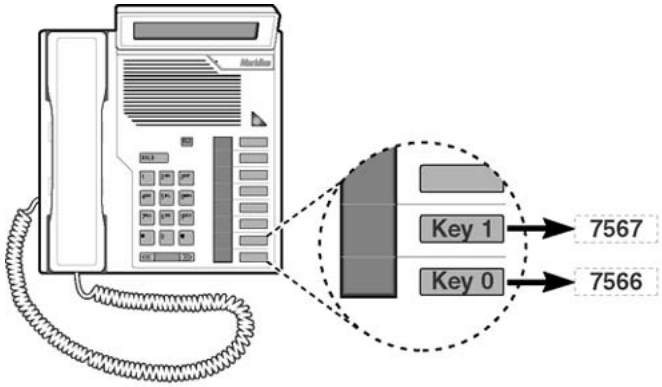
2. For a private ESN CDP call made from Key 0 (DN 2130) on the phone, the CLID would be built as follows:
   - LSC from CLID entry 0 (5) plus the DN of the active key (2130).
   - That is, the CLID that is sent would be 5 2130.
3. For a public local call made from Key 0 (DN 2130) on the phone, the CLID would be built as follows:
   - HLCL from CLID entry 0 (940) plus the DN of the active key (2130).
   - That is, the CLID that is sent would be 940 2130.
4. For a public national call made from Key 0 (DN 2130) on the phone, the CLID would be built as follows:
   - HNTN from CLID entry 0 (415) plus the HLCL from CLID entry 0 (940) plus the DN of the active key (2130).
   - That is, the CLID that is sent would be 415 940 2130.

Example 2 - Phone has DID number on DN key 0, International Numbering Plan.

In this example, the calls are made on DN key 0, which has a DID number. Key 0 is configured against CLID table entry 0, and key 1 is configured against CLID table entry 1.

Table 85: Set has DID number on DN key 0, for an International Numbering Plan.

CLID entry	HNTN (code for the home national number, equivalent to PFX1)	HLCL (code for the home local number, equivalent to PFX2), or LDN	DIDN (use the DN as the DID?)	HLOC (home location code)	LSC (local steering code)
Entry 0	2	6476	YES	578	6
Entry 1	2	64767676	NO	578	6
Entry 2			SRCH	578	6



553-7656.EPS

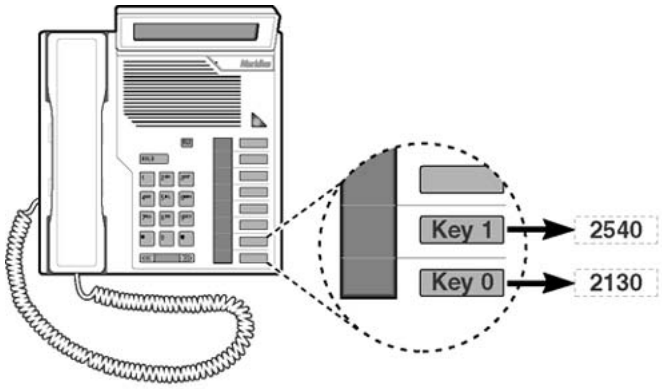
1. For a private ESN UDP call made from Key 0 (DN 7566) on the phone, the CLID would be built as follows:
   - HLOC from CLID entry 0 (578) plus the DN of the active key (7566).
   - That is, the CLID that is sent would be 578 7566.
2. For a private ESN CDP call made from Key 0 (DN 7566) on the phone, the CLID would be built as follows:
   - LSC from CLID entry 0 (6) plus the DN of the active key (7566).
   - That is, the CLID that is sent would be 6 7566.
3. For a public local call made from Key 0 (DN 7566) on the phone, the CLID would be built as follows:
   - HLCL from CLID entry 0 (6476) plus the DN of the active key (7566).
   - That is, the CLID that is sent would be 6476 7566.
4. For a public national call made from Key 0 (DN 7566) on the phone, the CLID would be built as follows:
   - HNTN from CLID entry 0 (2) plus the HLCL from CLID entry 0 (6476) plus the DN of the active key (7566).
   - That is, the CLID that is sent would be 2 6476 7566.

Example 3 - Phone does not have a DID number on DN key 1 (DIDN set to NO)

In this example, the calls are made on DN key 1, which does not have a DID number. Key 0 is configured against CLID table entry 0, and key 1 is configured against CLID table entry 1, which has the DIDN option set to NO. This means that the CLID is built using only the digits defined in the HLCL field of table entry 1 (the DN of key 1 would not be sent in the CLID after the HLCL digits.)

Table 86: Set does not have a DID number on DN key 1 (DIDN set to NO).

CLID entry	HNTN (code for the home national number)	HLCL (code for the home local number), or LDN	DIDN (use the DN as the DID?)	HLOC (home location code)	LSC (local steering code)
Entry 0	415	940	YES	646	5
Entry 1	415	9699170	NO	646	5
Entry 2			SRCH	646	5



553-7655.EPS

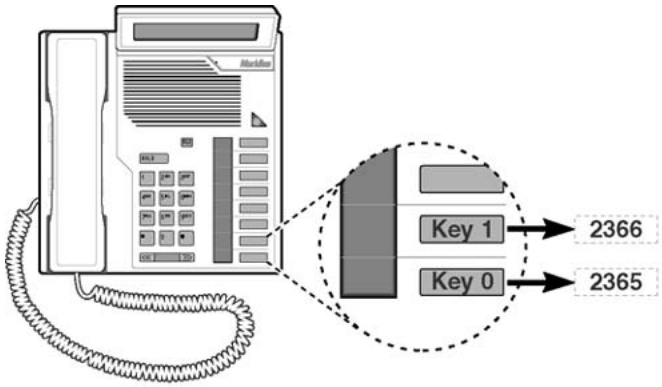
1. For a private ESN UDP call made from Key 1 (DN 2540) on the phone, the CLID would be built as follows:
    - HLOC from CLID entry 1 (646) plus the DN of the active key (2540).
    - That is, the CLID that is sent would be 646 2540.
2. For a private ESN CDP call made from Key 1 (DN 2540) on the phone, the CLID would be built as follows:
    - LSC from CLID entry 1 (5) plus the DN of the active key (2540).
    - That is, the CLID that is sent would be 5 2540.
3. For a public local call made from Key 1 (DN 2540) on the phone, the CLID would be built as follows:
    - HLCL from CLID entry 1 (9699170) - the DN of key 1 is not sent.
    - That is, the CLID that is sent would be 9699170.
4. For a public national call made from Key 1 (DN 2540) on the phone, the CLID would be built as follows:
    - HNTN from CLID entry 1 (415) plus the HLCL from CLID entry 1 (9699170).
    - That is, the CLID that is sent would be 415 9699170.

Example 4 - Phone does not have a DID number on DN key 1 (DIDN set to NO), International Numbering Plan.

In this example, the calls are made on DN key 1, which does not have a DID number. Key 0 is configured against CLID table entry 0, and key 1 is configured against CLID table entry 1, which has the DIDN option set to NO. This means that the CLID is built using only the digits defined in the HLCL field of table entry 1 (the DN of key 1 would not be sent in the CLID after the HLCL digits.)

Table 87: Set does not have a DID number on DN key 1 (DIDN set to NO).

CLID entry	HNTN (code for the home national number)	HLCL (code for the home local number), or LDN	DIDN (use the DN as the DID?)	HLOC (home location code)	LSC (local steering code)
Entry 0	89	3505	YES	581	6
Entry 1	89	35052020	NO	581	6
Entry 2			SRCH	581	6



553-7657.EPS

1. For a private ESN UDP call made from Key 1 (DN 2366) on the phone, the CLID would be built as follows:
   - HLOC from CLID entry 1 (581) plus the DN of the active key (2366).
   - That is, the CLID that is sent would be 581 2366.
2. For a private ESN CDP call made from Key 1 (DN 2366) on the phone, the CLID would be built as follows:
   - LSC from CLID entry 1 (6) plus the DN of the active key (2366).
   - That is, the CLID that is sent would be 6 2366.

3. For a public local call made from Key 1 (DN 2366) on the phone, the CLID would be built as follows:
    - HLCL from CLID entry 1 (35052020) - the DN of key 1 is not sent.
    - That is, the CLID that is sent would be 35052020.
4. For a public national call made from Key 1 (DN 2366) on the phone, the CLID would be built as follows:
    - HNTN from CLID entry 1 (89) plus the HLCL from CLID entry 1 (35052020).
    - That is, the CLID that is sent would be 89 35052020.

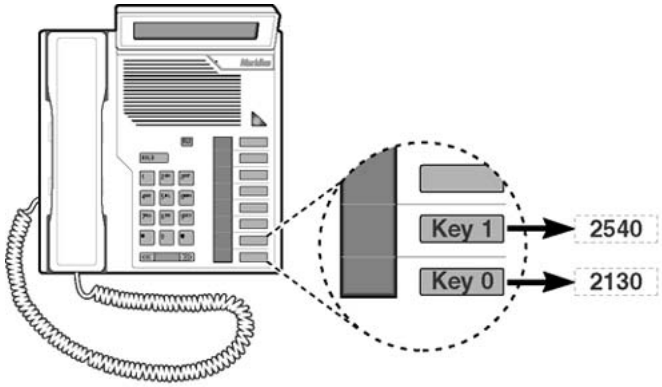
Example 5 - Phone does not have DID number on DN key 1, SRCH option is used (DIDN is set to SRCH)

In this example, the calls are made on DN key 1, which does not have a DID number. Key 0 is configured against CLID table entry 0, and key 1 is configured against CLID table entry 2, which has the DIDN option set to SRCH.

This means that the CLID assigned for key 1 is constructed based on a search starting from key 0. When a CLID entry is found, which has DIDN set to YES (that would be key 0), the HLCL for that key is used in the CLID for key 1. Therefore, the CLID that is sent for key 1 would contain the HLCL of key 0 plus the DN of key 0.

Table 88: Set does not have a DID number on DN key 1, SRCH is used (DIDN is set to SRCH).

CLID entry	HNTN (code for the home national number)	HLCL (code for the home local number), or LDN	DIDN (use the DN as the DID?)	HLOC (home location code)	LSC (local steering code)
Entry 0	415	940	YES	655	6
Entry 1	415	9699170	NO	655	6
Entry 2			SRCH	655	6



553-7655.EPS

1. For a private ESN UDP call made from Key 1 (DN 2540) on the phone, the CLID would be built as follows:
   - HLOC from CLID entry 2 (655) plus the DN of the active key (2540).
   - That is, the CLID that is sent would be 655 2540.
2. For a private ESN CDP call made from Key 1 (DN 2540) on the phone, the CLID would be built as follows:
   - LSC from CLID entry 2 (6) plus the DN of the active key (2540).
   - That is, the CLID that is sent would be 6 2540.
3. For a public local call made from Key 1 (DN 2540) on the phone, the CLID would be built as follows:
   - HLCL of CLID entry with DIDN set to YES (the HLCL of key 0, which is 940) plus the DN key 0 (2130.)
   - That is, the CLID that is sent would be 940 2130.
4. For a public national call made from Key 1 (DN 2540) on the phone, the CLID would be built as follows:
   - HNTN from CLID entry 0 (415) plus the HLCL from CLID entry 0 (940) plus the DN of key 0.
   - That is, the CLID that is sent would be 415 940 2130.

Example 6 -Set does not have DID number on DN key 1, SRCH option is used but does not find a DID

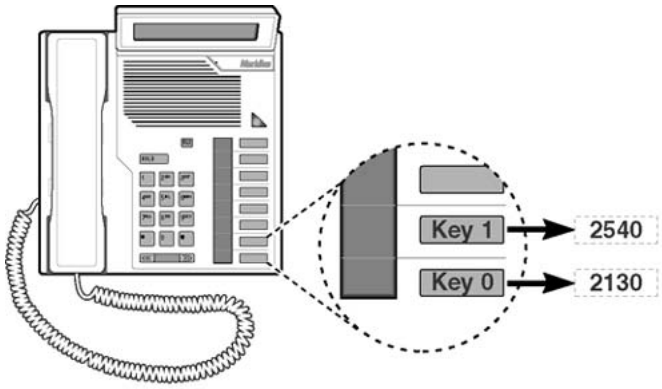
In this example, the calls are made on DN key 1, which does not have a DID number. Key 0 is configured against CLID table entry 1, and key 1 is configured against CLID table entry 2, which has the DIDN option set to SRCH. This means that the CLID assigned for key 1 is constructed based on a search starting from key 0.

Key 0 is configured against table entry 1. Since table entry 1 has DIDN set to NO, the search next takes place on key 1, which is configured against table entry 2. Since table entry 2 does not have DIDN set to YES either, the search fails to find a CLID entry which has DIDN set to

YES. This means that the CLID for key 1 is constructed using only the HLCL of the active DN (that is, the HLCL for key 1, which is the HLCL defined in table entry 2.)

Table 89: Set does not have a DID number on DN key 1, SRCH option fails to find a DID.

CLID entry	HNTN (code for the home national number)	HLCL (code for the home local number), or LDN	DIDN (use the DN as the DID?)	HLOC (home location code)	LSC (local steering code)
Entry 0	415	940	YES	646	5
Entry 1	415	9699170	NO	646	5
Entry 2	408	9885560	SRCH	655	6



553-7655.EPS

1. For a private ESN UDP call made from Key 1 (DN 2540) on the phone, the CLID would be built as follows:
   - HLOC from CLID entry 2 (655) plus the DN of the active key (key 1, 2540).
   - That is, the CLID that is sent would be 655 2540.
2. For a private ESN CDP call made from Key 1 (DN 2540) on the phone, the CLID would be built as follows:
   - LSC from CLID entry 2 (6) plus the DN of the active key (2540).
   - That is, the CLID that is sent would be 6 2540.
3. For a public local call made from Key 1 (DN 2540) on the phone, the CLID would be built as follows:
   - HLCL of CLID entry 2 (9885560) - since the search failed to find a DID number, only the HLCL of the active key (key 1, which is configured against table entry 2) is sent in the CLID. Since the DN of Key 1 (2540) is not a DID number, it is not sent in the CLID.

- That is, the CLID that is sent would be 9885560.

4. For a public national call made from Key 1 (DN 2540) on the phone, the CLID would be built as follows:
   - HNTN from CLID entry 2 (408) plus the HLCL from CLID entry 2 (9885560) - since the DN of Key 1 is not a DID number, the DN (2540) is not sent.
   - That is, the CLID that is sent would be 408 9885560.

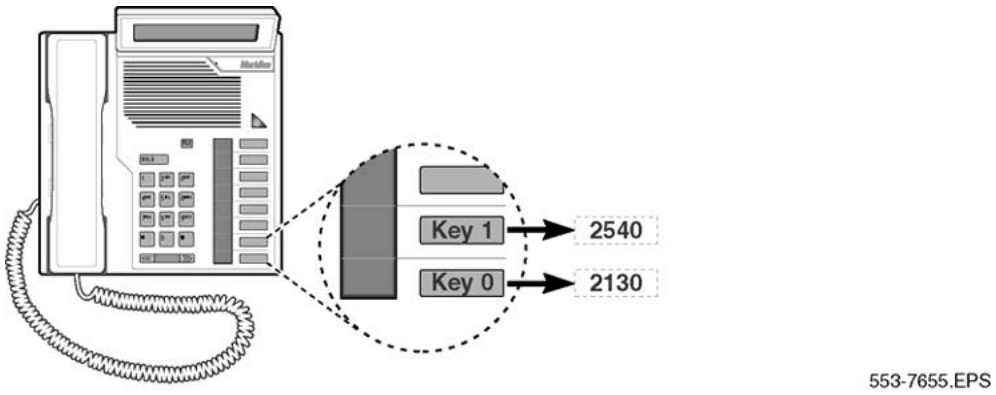
Example 7 - Phone with DN KEY prompt of D on key 1

In this example, the calls are made on DN key 1, which has the KEY prompt set to "D" in LD 11. This means that a search is done on the phone for a DN key, starting from key 0. The CLID of the found DN (key 0 in this case) key is used as the CLID of the active DN key; that is, the CLID for key 0 is used as the CLID for key 1.

Key 0 is configured against CLID table entry 0, and key 1 uses the CLID assigned to key 0.

Table 90: Set with DN KEY prompt of D on key 1.

CLID entry	HNTN (code for the home national number)	HLCL (code for the home local number), or LDN	DIDN (use the DN as the DID?)	HLOC (home location code)	LSC (local steering code)
Entry 0	415	940	YES	646	5
Entry 1	415	9699170	NO	646	5
Entry 2			SRCH	646	6



1. For a private ESN UDP call made from Key 1 (DN 2540) on the phone, the CLID would be built as follows:
   - HLOC from CLID entry 0 (646) plus the DN for key 0 (2130) - since the Key option has been set to 'D', a search is done from key 0 to find a DN key. The CLID for the found DN key is used as the CLID of active DN key - therefore,

CLID for key 0 (the found DN) is used in the CLID for key 1. This also applies to cases 2, 3 and 4 which follow.

- Therefore, the CLID that is sent would be 646 2130.

2. For a private ESN CDP call made from Key 1 (DN 2540) on the phone, the CLID would be built as follows:
   - LSC from CLID entry 0 (5) plus the DN of key 0 (2130).
   - That is, the CLID that is sent would be 5 2130.
3. For a public local call made from Key 1 (DN 2540) on the phone, the CLID would be built as follows:
   - HLCL of CLID entry 0 (940) plus the DN of key 0 (2130).
   - That is, the CLID that is sent would be 940 2130.
4. For a public national call made from Key 1 (DN 2540) on the phone, the CLID would be built as follows:
   - HNTN from CLID entry 0 (415) plus the HLCL from CLID entry 0 (940) plus the DN of key 0 (2130).
   - That is, the CLID that is sent would be 415 940 2130.

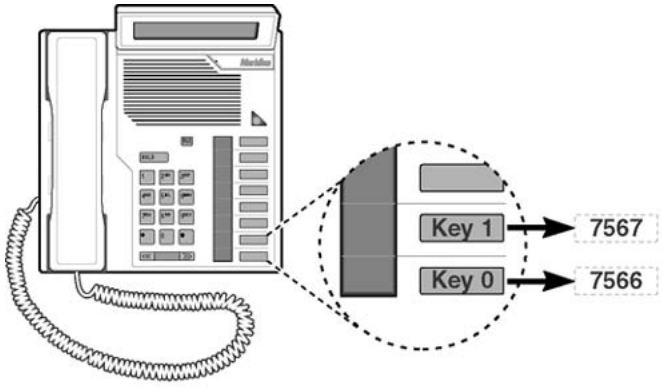
Example 8 - Phone with DN KEY prompt of D on key 1, International Numbering Plan

In this example, the calls are made on DN key 1, which has the KEY prompt set to "D" in LD 11. This means that a search is done on the phone for a DN key, starting from key 0. The CLID of the found DN (key 0 in this case) key is used as the CLID of the active DN key, that is, the CLID for key 0 is used as the CLID for key 1.

Key 0 is configured against CLID table entry 0, and key 1 uses the CLID assigned to key 0.

Table 91: Set with DN KEY prompt of D on key 1.

CLID entry	HNTN (code for the home national number)	HLCL (code for the home local number), or LDN	DIDN (use the DN as the DID?)	HLOC (home location code)	LSC (local steering code)
Entry 0	2	6476	YES	578	6
Entry 1	2	64767676	NO	578	6
Entry 2			SRCH	578	6



553-7656.EPS

1. For a private ESN UDP call made from Key 1 (DN 7567) on the phone, the CLID would be built as follows:
   - HLOC from CLID entry 0 (578) plus the DN for key 0 (7566) - since the Key option has been set to 'D', a search is done from key 0 to find a DN key. The CLID for the found DN key is used as the CLID of active DN key - therefore, CLID for key 0 (the found DN) is used in the CLID for key 1. This also applies to cases 2, 3, and 4 which follow.
   - Therefore, the CLID that is sent would be 578 7566.
2. For a private ESN CDP call made from Key 1 (DN 7567) on the phone, the CLID would be built as follows:
   - LSC from CLID entry 0 (6) plus the DN of key 0 (7566).
   - That is, the CLID that is sent would be 5 2130.
3. For a public local call made from Key 1 (DN 7567) on the phone, the CLID would be built as follows:
   - HLCL of CLID entry 0 (6476) plus the DN of key 0 (7566).
   - That is, the CLID that is sent would be 6476 7566.
4. For a public national call made from Key 1 (DN 7567) on the phone, the CLID would be built as follows:
   - HNTN from CLID entry 0 (2) plus the HLCL from CLID entry 0 (6476) plus the DN of key 0 (7566).
   - That is, the CLID that is sent would be 2 6476 7566.

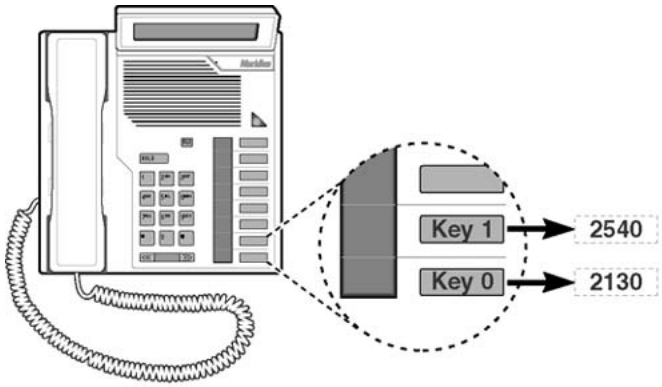
Example 9 - Phone with DN KEY prompt of D on key 0

In this example, the calls are made on DN key 0, which has the KEY prompt set to "D" in LD 11. This means that a search is done on the phone for a DN key, starting from key 0. The CLID of the found DN key (key 1 in this case) is then used as the CLID of the active DN key, that is, the CLID assigned to key 1 is used as the CLID for key 0.

Key 1 is configured against CLID table entry 0, and key 0 uses the CLID assigned to key 1.

Table 92: Set with DN KEY prompt of D on key 0.

CLID entry	HNTN (code for the home national number)	HLCL (code for the home local number), or LDN	DIDN (use the DN as the DID?)	HLOC (home location code)	LSC (local steering code)
Entry 0	415	940	YES	646	5
Entry 1	415	9699170	NO	646	5
Entry 2			SRCH	655	6



553-7655.EPS

1. For a private ESN UDP call made from Key 0 (DN 2130) on the phone, the CLID would be built as follows:

    - HLOC from CLID entry 0 (646) plus the DN for key 1 (2540) - since the Key option has been set to 'D', a search is done from key 0 to find a DN key. The CLID for the found DN key is used as the CLID of active DN key - therefore, CLID for key 1 (the found DN) is used in the CLID for key 0. This also applies to cases 2, 3, and 4 which follow.
    - Therefore, the CLID that is sent would be 646 2540.

2. For a private ESN CDP call made from Key 0 (DN 2130) on the phone, the CLID would be built as follows:

    - LSC from CLID entry 0 (5) plus the DN of key 1 (2540).
    - That is, the CLID that is sent would be 5 2540.

3. For a public local call made from Key 0 (DN 2130) on the phone, the CLID would be built as follows:

    - HLCL of CLID entry 0 (940) plus the DN of key 1 (2540).

- That is, the CLID that is sent would be 940 2540.

4. For a public national call made from Key 0 (DN 2130) on the phone, the CLID would be built as follows:
   - HNTN from CLID entry 0 (415) plus the HLCL from CLID entry 0 (940) plus the DN of key 1 (2540).
   - That is, the CLID that is sent would be 415 940 2540.

Operating parameters

In order for CLID to be properly delivered, a CLID entry must be defined in the CLID table for a customer. If a CLID entry or table is not defined, the active DN is sent as CLID.

The ISDN Calling Line Identification (CLID) Enhancements feature only pertains to calls that are made over ISDN routes.

The CLID that is sent from the system can be subject to any restrictions which can be imposed by the serving Telco.

This feature does not change the operation of CLID for attendant consoles. The codes for different call types are extracted from CLID entry 0 in LD 15, the customer data block. If the call type is private, the attendant DN is used. If the call type is public, LDN0 is used.

There is no system validation of digits entered in the CLID entry through a service change. Therefore, caution must be taken that any digits that are entered are valid and correct.

There is no cross-validation of the CLID entry associated with a phone against the CLID entry entered in LD 15. If a CLID entry is deleted from the CLID table and the CLID entry remains assigned to a DN key, the active DN is sent as CLID.

The maximum number of digits for a CLID in an ISDN message is 16.

The restriction remains of constructing a seven-digit local number or a 10-digit national number, for the North American Numbering Plan. Also, a seven-digit maximum is maintained for an ESN Uniform Dialing Plan number, if the Flexible Numbering Plan package is not equipped. The seven and 10 digit restriction for a public number does not apply to International ISDN interfaces.

The CLID is constructed using the information in the CLID table associated with the active DN key for the calling party in the Setup message and the connected party in the Connected and Notify messages. This feature does not change the Information Elements pertaining to any other ISDN network DNs, such as for Redirecting and Original Called Number.

For ISDN supplementary services, the ISDN CLID enhancements do not change how a Network DN is constructed. For a Local Number, the Network DN is formed using the Home Central Office Code (HNXX)/Prefix 2 (PFX2). For ESN Numbers, the Network DN is formed

using the Home Location Code (HLOC) for a Uniform Dialing Plan number or the Local Steering Code (LSC) for a Coordinated Dialing Plan number.

Feature interactions

Attendant Administration

Administration of a CLID entry, for a phone from an attendant console, is not supported.

Automatic Call Distribution

The ACD DN is sent as the CLID for a call made by an ACD agent using a DN key on a key other than Key 0. With the feature enhancement, the CLID is constructed using the CLID entry associated with the active DN key. The ACD agent ID is not designed to be sent as the CLID.

If an ACD agent has an active call on Key 0 and if a call transfer or conference is initiated by the ACD agent, the CLID entry associated with the ACD DN Key 0 is used as the CLID (for remote calls only.)

There is no CLID entry for an analog (500/2500-type) ACD DN. The CLID associated with the analog (500/2500-type) DN is used when a call transfer or conference is initiated by an ACD agent on an analog (500/2500-type) phone.

Call Detail Recording

The CLID in the CDR records, including the X records, contains the DN of the key from which the call is made, not the DN of Key 0.

Call Pickup Network Wide

The Private Integrated Services Network Exchange (PINX) DN in the customer data block is used for Call Pickup Network Wide. The network DN for the PINX DN is constructed using the existing Home Location Code (HLOC) or Local Steering Code (LSC) in LD 15. The DN of the originating party is constructed using the CLID associated with the active DN key. The DN of the originally called (ringing) party is constructed using the existing HLOC or LSC in LD 15.

For calls picked up from a secondary DN, the redirection DN in the Notify message is formatted with the CLID entry 0 of LD 15.

Calling Party Name Display

If a call transfer or conference is initiated on a multiple appearance DN programmed on a key other than Key 0, the Call Party Name Display associated with the DN of the active key is used, rather than the Call Party Name Display for Key 0.

CLID for an ISDN BRI phone

For an internal call terminating on an ISDN BRI phone, the calling phone's Dialed Digits Denied (DDGD)/Dialed Digits Allowed (DDGA) Class of Service is used to determine whether to send Calling Party Number to the terminating ISDN BRI phone for display purposes.

Calling Party Privacy Enhancement (CPPE)

The Calling Party Privacy Enhancement (CPPE) feature provides a route option to ignore the Calling Party Privacy Indicator for incoming calls received from all public Integrated Services Digital Network (ISDN) interfaces. When the Privacy Indicator Ignore (PII) prompt is YES in LD 16, the Calling Line Identification (CLID) Presentation Indicator and the Calling Party Name Display (CPND) Indicator override the blocking of the number and name details of the calling party. The name and number of the calling party appear on the receiving telephone.

The CPPE feature introduces a new route option AUXP for Auxiliary processor applications in LD 16 (RDB – Route Data Block). AUXP enhances the ability of the system to honor or ignore the Privacy Indicator for a Calling Party Privacy call, for each incoming route. If AUXP is 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, there will be no change to the CLID Presentation Indicator.

Connected Number

When a call is modified, such as by a call transfer, this feature enhancement will try to use the CLID entry associated with the active DN key if available, otherwise the connected number will be constructed using CLID entry 0.

EuroISDN Continuation for UK/Spain/Belgium/SN3

The EuroISDN Continuation feature allows Home National Numbers and Home Local Numbers to be configured on a route. When an ISDN call is made from a phone to a EuroISDN interface, the CLID constructed by EuroISDN, based on the outgoing route, takes precedence over the CLID constructed for the calling station phone.

EuroISDN Trunk - Network Side

The EuroISDN Trunk - Network Side connectivity supports all of the user side ISDN CLID enhancements.

Network Attendant Service

If Network Attendant Service is equipped, CLID entry 0 is used for incoming trunks.

Network Call Redirection

Network Call Redirection constructs Redirecting Number and Redirection Number. The feature enhancement does not change the construction of the Redirecting Number. However, the Redirection Number of the Notify message is constructed using the CLID entry 0 of LD 15.

Network Message Services

Message Waiting Indication with DMS

When a user leaves a voice message, from a multiple appearance DN, on a key other than Key 0 (such as Key 1), the caller's recorded number will be the multiple appearance DN on Key 1, rather than the primary DN of Key 0. This means that when the user returns the call, he/she will ring the DN of Key 1 on all the phones that have the appearance of the DN.

When a user retrieves messages using a multiple appearance DN key other than Key 0, the user now retrieves the messages on the other DN key.

The mail box number for a Network Message Service user is based on the original called number, in the form of a Coordinated Dialing Plan (CDP) or Uniform Dialing Plan (UDP) number. The construction of the original called number is not changed. If a user's DN is using a different Home Location Code or Local Steering Code for CLID, the Avaya CallPilot database adds the user's CLID as the mail box number. For example, if the first called number for a user is 6462300, a UDP number, and the user's CLID is 6472300, using CLID entry 'x', the user's mail box numbers are 6462300 and 6472300.

In general, by sending the DN associated with an active key to make a call to CallPilot, the secondary key's DN is included in the CallPilot User's definition (for either a local or remote user.)

Network Ring Again

The Network Ring Again feature remains operational when a user uses a multiple appearance DN on a key other than Key 0 to activate Network Ring Again, since Network Ring Again saves the Terminal Number of the phone that initiates Network Ring Again.

Remote Virtual Queuing

The Remote Virtual Queuing (RVQ) feature remains operational when a user uses a multiple appearance DN on a key other than Key 0 to activate RVQ.

Transaction Capabilities Application Part (TCAP)

The network DN for TCAP messages is constructed using the existing Home Location Code (HLOC), Local Steering Code (LSC) or Home Numbering Plan Area (HNPA)/Home Central Office Code (HNXX) prompts in LD 15. The feature enhancement does not change the construction of the TCAP messages.

Feature packaging

This feature requires the Integrated Services Digital Network (ISDN) package 145.

Feature implementation

Task summary list

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

1. Table 93: LD 15 - Construct the CLID entry for a customer. on page 221
2. Table 94: LD 21 - Print the CLID table for a customer. on page 222
3. Table 95: LD 10 - Define the CLID entry for analog (500/2500-type) phones. on page 223

4. Table 96: LD 11 - Define the CLID entry for DN keys for Meridian 1 Proprietary Phones. on page 223
5. Table 97: LD 27 - Define the CLID entry for ISDN BRI phones. on page 225

Table 93: LD 15 - Construct the CLID entry for a customer.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	NET	Networking data (if REQ = CHG only.)
CUST	0-99	Customer number as defined in LD 15.
ISDN	YES	Integrated Services Digital Network.
- PNI	1-327000	Private Network Identifier.
- CLID	(NO) YES	CLID option. YES = configure a CLID table for the customer. NO = (the default) do not configure a CLID table. In this case, the remaining prompts are not generated, and no CLID is sent for the customer.
- - SIZE	0-(256)-4000	The maximum number of CLID entries needed for a customer. If REQ = NEW, you can select the default value (256) by entering <CR> in response to this prompt. It is advised that you not define a size much larger than actually needed. This entry can be increased or decreased as required.
- - INTL	0-9999 X	Country code, for international number. Enter X to delete digits.
- - ENTRY	aaaa Xaaaa Xaaaa Xbbbb <CR>	aaaa = CLID entry to be configured. Xaaaa = CLID entry to be deleted. Xaaaa Xbbbb = CLID entries to be deleted. aaaa and bbbb must be a value between 0 and (SIZE-1). The ENTRY prompt is repeated until <CR> is entered as a response. If REQ = NEW, only one new entry can be created. The entry will be saved to system memory when the configuration for the entire overlay is completed. If REQ = CHG, as many entries as needed can be created, changed or deleted. The action for the entry will be saved to system memory after the CLID entry has been completely configured, that is, after the LSC prompt has been answered. If a new CLID entry is created, or an existing CLID entry is changed, the message "ENTRY aaaa SAVED" is displayed after the LSC prompt. If a CLID entry or CLID entries is/are

Prompt	Response	Description
		deleted, the message "ENTRY aaaa DELETED" or "ENTRIES aaaa-bbbb DELETED" is displayed after the LSC prompt.
- - - HNTN	0-999999 X	National code for home national number (1-6 digits). X = delete digits.
- - - HLCL	0-999..9 X	Local code for home local number or Listed Directory Number (1-12 digits). X = delete digits.
- - - DIDN		How to use the DN as a DID when constructing a CLID national or local number.
	(YES) NO SRCH	YES = The default. The CLID is constructed using the digits defined in HLCL followed by the DN of the active key. NO = Construct the CLID using the digits defined in HLCL. SRCH = Search on the phone, from key 0 - upwards, to find a CLID entry which has the DIDN set to YES. Use the found CLID to construct the local number.
- - - HLOC	0-9999999 X	Home location code (ESN), 1-7 digits. X = delete digits.
- - - LSC	0-9999999 X	Local steering code, 1-7 digits. X = delete digits.
ENTRY aaaa SAVED ENTRY aaaa DELETED ENTRIES aaaa-bbbb DELETED		Displayed message. Refer to Note 3 for the ENTRY prompt.

It is advised that after the CLID table is built in LD 15, generate a printout using LD 21 to verify the table entries defined in LD 15.

Table 94: LD 21 - Print the CLID table for a customer.

Prompt	Response	Description
REQ	PRT	Print the customer data block.
TYPE	CLID	CLID entry data.
CUST	0-99	Customer number as defined in LD 15.
SIZE	0-(256)-1000	The value that was defined for SIZE in LD 15 is printed automatically after the customer number has been entered.
RNGE	aaaa aaaa bbbb <CR>	CLID entry to be printed. Range of CLID entries to be printed (from aaaaa to bbbbb), aaaa and bbbb must be between 0 and the value defined for SIZE-1 in LD 15. Printing begins when <CR> is entered.

Prompt	Response	Description
INTL	0-9999	Country code, for international number, is printed.
ENTRY	aaaa	CLID entry number in CLID table is printed. If a range of entries is entered in the RNGE prompt, all of the entries in the specified range are printed in sequence.
- HNTN	0-999999	National code for home national number, 1-6 digits, is printed.
- HLCL	0-999..9	Local code for home local number or Listed Directory Number, 1-12 digits, is printed.
- DIDN	YES NO SRCH	The desired choice of how to use the DN as a DID, when constructing a CLID national or local number, will be printed.
- HLOC	0-9999999	Home location code (ESN), 1-7 digits, is printed
- LSC	0-9999999	Local steering code, 1-7 digits, is printed.

Table 95: LD 10 - Define the CLID entry for analog (500/2500-type) phones.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	500	500 phone.
TN	l s c u	Terminal Number for Large System and CS 1000E
DIG	xx yy	Dial Intercom group number and member number.
DN	xxxx (0)-N	DN and CLID entry. N = CLID SIZE-1 (SIZE defined in LD 15).

Note:

When assigning a CLID entry to an ACD phone, you cannot use the same position ID already on the phone. The phone must be first outted, or the ACD key must be nulled and then rebuilt with the table entry number.

Table 96: LD 11 - Define the CLID entry for DN keys for Meridian 1 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	l s c u	Terminal Number for Large System and CS 1000E.

Prompt	Response	Description
KEY		Phone function key assignments.
	xx MCN yyyy (0)-N/D xx MCR yyyy (0)-N/D xx PVN yyyy (0)-N/D xx PVR yyyy (0)-N/D xx SCN yyyy (0)-N/D xx SCR yyyy (0)-N/D	xx = key number. MCN = Multiple Call Non-ringing key. MCR = Multiple Call Ringing key. PVN = Private Line Non-Ringing key PVR = Private Line Ringing key SCN = Single Call Non-ringing key. SCR = Single Call Ringing key. yyyy = DN. (0)-N = CLID entry, with N = CLID SIZE-1 (SIZE defined in LD 15). D = CLID entry. Search for a CLID entry from key 0 upwards, to find a DN key. The found CLID is used as the CLID entry for the active DN key.
KEY	xx HOT D dd yyy...y zzzz m (0)-N/D	Two-way Hotline Direct key, where: xx = key number. dd = number of digits dialed. yyy...y = target number (terminating DN, maximum of 31 digits). zzzz = two-way hotline DN. m = one of the following Terminating Modes: • H = Hotline (default) • N = Non-ringing • R = Ringing • V = Voice (0)-N = CLID entry, with N = CLID SIZE-1 (SIZE defined in LD 15). D = CLID entry. Search for a CLID entry from key 0 upwards, to find a DN key. The found CLID is used as the CLID entry for the active DN key.
KEY	xx HOT L bbb zzzz (0)-N/D	Two-way Hotline List key, where: xx = key number. bbb = Hot Line List entry (0-999). zzzz = two-way hotline DN. (0)-N = CLID entry, with N = CLID SIZE-1 (SIZE defined in LD 15). D = CLID entry. Search for a CLID entry from key 0 upwards, to find a DN key. The found CLID is used as the CLID entry for the active DN key.
KEY	xx ACD aaaa 0-N/D bbbb	ACD key, where: xx = key number. aaaa = ACD DN or Message Center DN. 0-N = CLID entry, with N = CLID SIZE-1 (SIZE defined in LD 15). D = CLID entry. Search for a CLID entry from key 0 upwards, to find a DN key. The found CLID is used as the

Prompt	Response	Description
		CLID entry for the active DN key. bbbb = ACD agent's position ID.

Table 97: LD 27 - Define the CLID entry for ISDN BRI phones.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	TSP	Administer the Terminal Service Profile on the Digital Subscriber Loop.
...		
- SPID	xxxxxxx	Service Profile Identifier.
DN	xxxx (0)-N	xxxx = DN to be associated with the TSP. (0)-N = CLID entry, with N = CLID SIZE-1 (SIZE defined in LD 15).

Feature operation

No specific operating procedures are required to use this feature.

Chapter 22: ISDN PRI Central Office Connectivity

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

This feature allows the interconnection of the system with public exchanges in the international markets that support the following protocols:

- Asia Pacific, consisting of:
  - Australia (private or alternative carrier)
  - China
  - Hong Kong
  - India
  - Indonesia
  - Japan
  - Malaysia
  - New Zealand
  - Philippines
  - Singapore
  - Taiwan
  - Thailand
- Australia ETSI
- AXE-10 Australia (non-Asia Pacific ISDN Connectivity)
- AXE-10 (Sweden)
- EuroISDN
- Japan D70 (non-Asia Pacific ISDN Connectivity)
- NEAX-61 (New Zealand) (non-Asia Pacific ISDN Connectivity)
- Numeris VN3 (France)
- SwissNet (Switzerland)

- SYS-12 (Norway)
- 1TR6 (Germany)

These connectivities are described in the following sections.

Asia Pacific ISDN PRI Central Office Connectivity

Description

The Asia Pacific ISDN PRI Central Office Connectivity provides ISDN Primary Rate Interface (PRI) connectivity between the system and Public Exchange/Central Offices in the following Asia Pacific markets:

- Australia (private or alternative carrier)
- China
- Hong Kong
- India
- Indonesia
- Japan
- Malaysia
- New Zealand
- Philippines
- Singapore
- Taiwan
- Thailand

The Asia Pacific connectivities support the following ISDN features:

- Basic Call Service
- Back-up D-Channel, for Hong Kong
- Advice of Charge, for Japan (considered a basic service)
- Malicious Call Trace, for Australia
- Advice of Charge (AOC) at End of Call, for Australia
- Incoming Trunk Programmable CLID for analog trunks, for Australia. This feature is available for use in a private or alternative carrier network, as required in Australia.

- nB+D, for Japan (up to 215 B-channels/nine interfaces), for New Zealand (up to 120 B-channels/four interfaces), for Malaysia and (up to 120 B-channels/four interfaces),
- Calling Line Identification Presentation and Restriction (CLIP and CLIR)
- Connected Line Identification Presentation and Restriction (COLP and COLR), for India, Philippines, Taiwan, and Indonesia
- Circuit switched voice and data on the B-channel
- Direct Dialing Inward (DDI/DID), for Indonesia
- Overlap Sending (supported by all interfaces except Japan, Philippines)
- Overlap Receiving, for India, Indonesia, China, Malaysia, and Thailand
- COT, DID, DOD, and TIE trunk call types, as applicable
- 64 Kbit/s clear digital information
- Flexible Numbering Plan
- Sub-addressing (supported only when information is received from the Asia Pacific ISDN interfaces and passed through a tandem node)
- Channel Negotiation (for all countries except Singapore; see the note which follows)

Note:

As part of the Singapore enhancement, Channel Negotiation is not supported for Singapore. The CNEG option must be set to 1 (the default) in LD 17.

Interworking is provided with the following other ISDN or non-ISDN interfaces.

- existing ISDN CO connectivities (e.g., NUMERIS, Swiss Net 2)
- Taiwan R1 Modified Signaling
- private networks (for example, Meridian Customer Defined Network, and QSIG)
- other digital signaling systems (e.g., Digital Private Network Signaling System #1 (DPNSS1), and Digital Access Signaling System #2 (DASS2), and 1.5/2.0 Mbit DTI)
- analog signaling systems (Inter Register Multifrequency Compelled signaling (R2/MFC), Multifrequency Compelled Signaling for Socotel (MFE), dial pulse (DP), and Digitone (DTN)

Operating parameters

The Singapore interface complies with the updated national ISDN specification issued by the Telecommunication Authority of Singapore (TAS). This latest specification for Singapore requires that the Channel Negotiation option no longer be supported. Therefore, the CNEG option must be set to 1 (the default) in LD 17 for the Singapore interface type. Also pertaining to the Singapore interface, PROGRESS messages are supported only for BRI (not supported for PRI), and messages with Global Call Reference are supported only for PRI.

The system only supports point-to-point data link connections since PRI only allows a single-point terminal configuration on a user-to-network ISDN interface. BRI supports only point-to-point data link connections.

The ISDN PRI system to Asia Pacific connectivity feature does not support a backup D-channel, except for Hong Kong.

Network Attendant Service (NAS) is not supported across the Asia Pacific interface; however, incoming calls can be routed by NAS from another node.

In a Meridian Customer Defined Network (MCDN), receiving Calling Party Name Display (CPND) from an Asia Pacific ISDN and sending a CPND from an Asia Pacific ISDN are not supported.

Network Call Redirection, Network Call Forward, and Network Call Forward No Answer (MCDN Component) are not supported across an Asia Pacific ISDN interface.

For all of the Asia Pacific interfaces, User-to-User Signaling and Multiple Subscriber Number, Calling Party Name Display, and Call Diversion are not supported.

For all of the Asia Pacific interfaces, the SUSPEND, SUSPEND ACKNOWLEDGE, SUSPEND REJECT, RESUME, RESUME ACKNOWLEDGE, RESUME REJECT call rearrangement messages are not supported.

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

nB+D Primary Rate Access is supported only for Hong Kong, Japan, New Zealand, and Malaysia.

For the Japan interface, Overlap Signaling (Sending and Receiving) is not supported.

For the Hong Kong, Singapore, Australian, and New Zealand interfaces, Overlap Receiving is not supported.

For the Japan interface, Advice of Charge is not provided on BRI terminals.

The following features and capabilities are not supported across any Asia Pacific interface:

- PSTN Equal Access Service - North America Service
- Integrated Service Access (ISA)
- FEX and WATS call types
- Network Wide Ring Again (NRAG)
- Network Calling Party Name Display (NCPND)
- CLID Enhancements (Redirecting Number)
- Network Call Redirection, Network Call Forwarding, Network Call Forward No Answer
- Trunk Optimization/Anti-Tromboning
- ISDN Signaling Link

- RESTART message and INSERVICE message for BRI Trunk Access
- ESN Signaling on PRA (NCOS)
- Virtual Network Services (VNS)

The Malaysia interface does not support RESTART and RESTART ACKNOWLEDGE messages for BRIT.

Hardware requirements

For Large Systems, the following hardware is required:

- For Japan, Taiwan, and Hong Kong interfaces:
  - For Primary Rate Access, the QPC720 1.5 Mbit PRI with the NT6D80AA MSDL card, or the NT5D12AA dual port 1.5 DTI/PRI card with the required NTBK51AA/ NTBK51CA two-port Downloadable D-Channel daughterboard (DDCH).
  - For D-Channel processing, the NT6D80AA MSDL card, or the NTBK51AA/ NTBK51CA dual-port Downloadable D-Channel daughterboard (DDCH) to be used as an option to the MSDL with the NT5D12AA dual port 1.5 Mbit DTI/PRI.
  - For ISDN Basic Rate Access, the NT6D73AA MISP card and the NT6D70BA SILC card.
- For Australia, China, Indonesia, Malaysia, New Zealand, Singapore, Thailand, India, and the Philippines:
  - For Primary Rate Access, the NT8D72AA PRI2 card, or the NT5D97AD Dual PRI2/ DTI2 card.
  - For D-Channel processing, the NT6D80AA Multi-Serial Data Link (MSDL) card, or as an alternative to the MSDL, the NTBK51AA/NTBK51CA dual-port Downloadable D-Channel Daughterboard (DDCH) to be used with the NT5D97AD Dual DTI/PRI2 card.
  - For ISDN Basic Rate Access, the NT6D73AA MISP card and the NT6D70BA SILC card.

For CS 1000E, the following hardware is required:

- For Japan, Taiwan, and Hong Kong interfaces:
  - For Primary Rate Access, the 1.5 NTAK09AA (DTI/PRI) circuit card is required alongwith the NTBK51BA Downloadable D-Channel (DCHI) card.
  - For Basic Rate Access, the NTBK22XX MISP card and the NT6D70BA SILC card.
- For Australia, China, Indonesia, Malaysia, New Zealand, Singapore, Thailand, India, and the Philippines:
  - For Primary Rate Access, the NTBK50AA PRI2 card is required along with theNTBK51BA Downloadable D-Channel (DCHI) card

- For Basic Rate Access, the NTBK22XX MISP card and the NT6D70BA SILC card.

Feature interactions

Call Detail Recording

For a simple call, the total cost for the call is produced in the Call Detail Record.

For the Japan connectivity, if a call is redirected, the Call Detail Recording (CDR) records show all extensions associated with the particular call, however, it is not possible to show call costs associated with each extension. This is because the CDR reports which pertain to the Japan protocol (used for the Japan interface), charge information only at the point when the call is disconnected.

Calling Line Identification Enhancements

Prior to the CLID Enhancements feature, 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, for each phone. 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 can now support 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 a North American Numbering Plan, the system now supports multiple Home Central Office Codes (HNXXs), Home Numbering Plan Area (HNPA) codes, Home Location Codes (HLOCs) and Local Steering Codes (LSCs), on a DN or DN key basis.

For a North American Numbering Plan, each CLID entry can contain the following:

- a 1–6 digit national code for a home national number (HNTN)
- a 1–12 digit local code for a home local number (HLCL), or a one-12 digit Listed Directory Number for a switchboard
- a 1–7 digit Home Location Code (HLOC)
- a 1–7 digit Local Steering Code (LSC)

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

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

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

Another new capability, which applies to both the North American Numbering Plan and the International Numbering Plan, pertains to how the HLCL is constructed. A new prompt, DIDN (which signifies "use DN as a DID number") in 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)

Connected Line Identification Presentation and Restriction (COLP and COLR) are applicable only for Indonesia, India, Taiwan, and Philippines.

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

The same Class of Service is used to control both Connected Line Identification Restriction and Calling Line Identification Restriction (CLIR). Thus, if a user has presentation restricted

configured, their number is sent to the other party for both incoming and outgoing calls with the presentation flag set to restricted.

Electronic Switched Network (ESN)

Basic Alternate Route Selection (BARS)

BARS interacts with the Asia Pacific ISDN interface as follows:

- NPA and NXX cannot be used on the Asia Pacific ISDN interface, as the public network cannot accept them
- Special Numbers (SPNs) are converted to "unknown" type and plan, as the Electronic Switched Network (ESN) SPN is a North American concept

Network Alternate Route Selection (NARS)

All of the BARS concerns apply to NARS.

In addition, LOC codes must be used with care, since the networking features do not accept a Digit Manipulation Index (DMI) used to insert an Electronic Switched Network (ESN) access code. Therefore, the only supported usage has the trunks effectively being treated as though they are non-ISDN.

Coordinated Dialing Plan

A Coordinated Dialing Plan (CDP) call can access an Asia Pacific trunk. However, since neither the private plan nor the CDP numbering type is supported by the Asia Pacific ISDN, these are converted to a type and plan supported by the public network.

DPNSS Gateway

The Asia Pacific interface interacts with the DPNSS gateway in a manner consistent with other PSTN interfaces. The Information Elements (IEs) supported on the Asia Pacific interface, which have equivalents in DPNSS interwork are translated on an IE by IE basis. IEs which have no matching DPNSS element are discarded. The same is true for DPNSS to Asia Pacific. Supported IEs with DPNSS equivalents pass through with appropriate mapping.

Incoming Digit Conversion

The Asia Pacific interface supports Incoming Digit Conversion ISDN DID routes.

The Incoming Digit Conversion (IDC) feature provides the availability to convert the incoming digits from a DID route. Digits received as a called party number are converted if the IDC feature is activated on the route. Digit analysis is then performed on the converted digits by the system.

MCDN Call Redirection (Call Forward, Hunt, Call Forward No Answer)

These features operate in the same manner as when a call to a phone is redirected over any other Central Office trunk; however, notices of redirection or modification are not transferred.

nB+D

For the Malaysia and New Zealand connectivities, the channel identification scheme used must be changed from implicit to explicit.

Outpulsing of Asterisk and Octothorpe

The Outpulsing of Asterisk and Octothorpe is not supported for any Asia Pacific interface.

PRI Channel Negotiation

This interaction is applicable to the Japan interface only. When a D-channel has one or more secondary PRIs associated with it while using the Japan interface, channel negotiation requires that the PRI interface be identified (implicitly) in the channel interface IE for all B-channels not on the primary interface whether secondary loops are configured or not.

Trunk Optimization

Trunk Optimization is not supported across the Asia Pacific interface, but interworks transparently. In other words, trunk optimization will not be observed at the Asia Pacific ISDN. The call will appear to be static, not undergoing any modifications.

T309 Timer

The system supports the T309 timer on PRI trunks for all Asia Pacific interfaces.

Virtual Network Services (VNS)

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

Feature packaging

There are no new software packages required for this feature. However, the following packages are necessary in order to connect the system over an Asia Pacific ISDN PRI interface to a Central Office:

- Digit Display (DDSP) package 19
- 1.5 Mbit Digital Trunk Interface (PBXI) package 75
- Integrated Services Digital Network (ISDN) package 145
- For Japan, Taiwan and Hong Kong, 1.5 Mbit Primary Rate Access (PRA) package 146
- For Australia, China, Indonesia, Malaysia, New Zealand, Singapore, Thailand, India, and the Philippines, 2.0 Mbit Primary Rate Interface (PRI2) package 154, which has the following dependencies:
  - 2.0 Mbit Digital Trunk Interface (DTI2) package 129
  - Integrated Services Digital Network (ISDN) package 145
- International Primary Rate Access (IPRA) package 202, which has the following dependencies:
  - Digit Display (DDSP) package 19
  - Integrated Services Digital Network (ISDN) package 145
  - 2.0 Mbit Primary Rate Interface (PRI2) package 154
- Meridian 1 Extended Peripheral Equipment (XPE) package 203
- Multi-Purpose Serial Data Link (MSDL) package 222, which requires the Integrated Services Digital Network (ISDN) package 145
- New Format CDR (FCDR) package 234

For Basic Rate Interface trunk access the following packages are required instead of the PRI-related packages:

- Basic Rate Interface (BRI) package 216
- Basic Rate Interface Trunk (BRIT) package 233

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

The International nB+D (INBD) package 255 is required to configure nB+D PRI for the following interfaces only: D70, JAPN, HKNG, TCNZ, and MSIA.

The Overlap Signaling feature requires:

- Basic Routing (BRTE) package 14
- Network Class of Service (NCOS) package 32

- Basic Alternate Route Selection (BARS) package 57, Network Alternate Route Selection (NARS) package 58, and/or Coordinated Dialing Plan (CDP) package 59
- Flexible Numbering Plan (FNP) package 160
- Overlap Signaling (OVLP) package 184

Note:

This package should not be enabled for the Philippines

- Basic Queuing (BQUE) package 28

Feature implementation

Task summary list

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

1. Table 98: LD 15 - Configure the Customer Data Block for Asia Pacific with Calling Line Identification. on page 240
2. Table 99: LD 17 - Configure the Asia Pacific D-channel. on page 242
3. Table 100: LD 17 - Configure a Back-up D-Channel for Hong Kong. on page 245
4. Table 101: LD 17 - Configure nB+D for Hong Kong, Japan, New Zealand, and Malaysia. The International nB+D (INBD) package 255 must be enabled to configure nB+D for these interfaces. on page 246
5. Table 102: LD 17 - Configure the Bearer Capability for the Asia Pacific connectivity D-Channel. on page 248
6. Table 103: LD 16 - Configure Asia Pacific ISDN PRI Route Data Block. on page 248
7. Table 104: LD 14 - Configure Asia Pacific ISDN PRI trunks. on page 250
8. Table 105: LD 73 - Define the frame format and Grade of Service timers for a PRI/PRI2 loop. on page 251

Table 98: LD 15 - Configure the Customer Data Block for Asia Pacific with Calling Line Identification.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	NET	Networking data (if REQ = CHG only.)
CUST	0-99	Customer number as defined in LD 15.
OPT	a…a	Options.

Prompt	Response	Description
AC2	aaaa	Access Code 2 as defined in LD 86 (aaaa = NPA, NXX, INTL, SPN, or LOC).
CLID	YES	Configure CLID option for Asia Pacific.
- SIZE	0-(256)-4000	The maximum number of CLID entries needed for a customer. **Note:** Do not define a size much larger than actually needed. This entry can be increased or decreased as required.
- INTL	0-9999	Country code, for international number.
- ENTRY	aaaa Xaaaa Xaaaa Xbbbb	aaaa = CLID entry to be configured. Xaaaa = CLID entry to be deleted. Xaaaa Xbbbb = CLID entries to be deleted. aaaa and bbbb must be a value between 0 and (SIZE-1). The ENTRY prompt is repeated until <CR> is entered as a response. If REQ = CHG, as many entries as needed can be created, changed or deleted. The action for the entry is saved to system memory after the CLID entry has been completely configured; that is, after the LSC prompt has been answered. If a new CLID entry is created, or an existing CLID entry is changed, the message "ENTRY aaaa SAVED" is displayed after the LSC prompt. If a CLID entry or CLID entries is/are deleted, the message "ENTRY aaaa DELETED" or "ENTRIES aaaa-bbbb DELETED" is displayed after the LSC prompt.
- - HNTN	0-999999	National code for home national number (1-6 digits).
- - HLCL	0-999..9	Local code for home local number or Listed Directory Number (1-12 digits).
- - DIDN		How to use the DN as a DID when constructing a CLID national or local number.
	(YES) NO SRCH	YES = The CLID is constructed using the digits defined in HLCL followed by the DN of the active key (default). NO = Construct the CLID using the digits defined in HLCL. SRCH = Search on the phone, from key 0 - upwards, to find a CLID entry which has the DIDN set to YES. Use the found CLID to construct the local number.
- - HLOC	0-9999999	Home location code (ESN), 1-7 digits.
- - LSC	0-9999999	Local steering code, 1-7 digits.
ISDN	YES	Integrated Services Digital Network.
- PNI	1-327000	Private Network Identifier.

Prompt	Response	Description
ENTRY aaaa SAVED ENTRY aaaa DELETED ENTRIES aaaa-bbbb DELETED		Displayed message appears after each entry is input.

Table 99: LD 17 - Configure the Asia Pacific D-channel.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	CFN	Configuration data block.
- ADAN	NEW DCH xx	Add a D-channel on logical port 0-63 Large Systems.
- CTYP	MSDL	Card type where: MSDL = The NT6D80 Multi-purpose Serial Data Link or the NTBK51AA/NTBK51CA Downloadable D-Channel Daughterboard for Large Systems.
- GRP	0-4	Network group number for Option 81C systems.
- DNUM	0-15	Device number for I/O ports. All ports on the MSDL card share the same DNUM. The MSDL card address settings must match the DNUM value.
- CDNO	c	Card number for downloadable D-Channel Daughterboard for CS 1000E, where c = 1–4: chassis on superloop, 1–10: cabinet or MG 1010 on superloop.
- PORT	0-3 1	Port number on MSDL cards.
- DES	a…a	Designator. DES is used to identify the link and can be up to 16 alphanumeric characters: 0-9, and upper case A-Z. Characters "*" and "#" are not allowed.
- USR	PRI	This D-channel is used for Primary Rate only.
- IFC	APAC	Asia Pacific ISDN interface.
- - CNTY		Country associated with the APAC interface.
	AUST CHNA HKNG INDI INDO JAPN MSIA PHLP SING TAIW TCNZ	Australia. China. Hong Kong. India. Indonesia. Japan. Malaysia. Philippines. Singapore. Taiwan. New Zealand.

Prompt	Response	Description
	THAI	Thailand.
- - ISDN_MCNT	60-(300)-350	Layer 3 call control message count per 5 second time interval.
CLID	OPT4	OPT4 is the default for all Asia Pacific interface.
PROG	NCHG MALE MCON	Progress signal. NCHG = No Change. This is the default for all Asia Pacific interfaces except Singapore, Japan, and Australia. MALE = Alert message. MCON = Connect message. This is the default for the Australia interface. **Note:** The PROG prompt should not be configured for Japan and Singapore, since these countries do not support the Progress signal.
- DCHL	0-159	PRI/PRI2 loop number for D-channel.
	l	PRI/PRI2 loop number for Backup D-channel. For CS 1000E where l = 0, 4, 8, - 252: superloop number in multiples of 4.
- CNEG	(1)	Channel negotiation option. Channel is indicated and no alternative acceptable, exclusive.
	2	Channel is indicated and any alternative acceptable, preferred. **Note:** for the Singapore interface, CNEG must be set to 1.
- RLS	xx	Software release of the far end switch. If the far end has an incompatible release, it prevents the sending of application messages.
- RCAP		Remote capabilities, to configure the Connected Line ID Presentation supplementary service.
	(XCOL) COLP	To remove COLP. CLID Presentation supported. This prompt appears if CNTY = Indonesia, India, Taiwan, and the Philippines.
…		
- OVLR	YES	Allow Overlap Receiving. This prompt applies only if CNTY = THAI, MSIA, INDI, INDO, or CHNA.
- - DIDD	(0)-15	Number of leading digits to delete from DID trunks.
- OVLS	(NO) YES	(Do not) allow Overlap Sending.

Prompt	Response	Description
		Enter NO for Japan and Philippines. Enter YES for all other interfaces.
- - OVLT	(0)-8	Overlap Timer in seconds. This timer controls the interval between the sending of INFORMATION messages. "0," the default, means send immediately.
- TIMR	YES	Change programmable timers. Only supported for interfaces supporting one of the following timers.
- - T310	10-(10) (30)-60	Maximum time in seconds between an incoming CALL PROCEEDING message and the next incoming message. Not supported for Australia. Default values are as follows: CHNA = 30 seconds. TAIW = 30 seconds. PHLP = 10 seconds. HKNG = 10 seconds. INDI = 10 seconds. INDO = 10 seconds. JAPN = 10 seconds. MSIA = 10 seconds. SING = 10 seconds. TCNZ = 10 seconds. THAI = 10 seconds.
- - INC_T306	0-(2)-240	Variable timer for received DISCONNECT message on incoming calls allowing in-band tone to be heard when the network sends in-band tone. T306 is the duration of the network timer in seconds. The network will stop sending after T306 times out, so the maximum time will be T306. The value is stored in two-second increments, which are rounded up.
- - OUT_T306	0-(30)-240	Variable timer for received DISCONNECT message on outgoing calls allowing in-band tone to be heard when the network sends in-band tone. T306 is the duration of the network timer in seconds. The network will stop sending after T306 times out, so the maximum time will be T306. The value is stored in two-second increments, which are rounded up.
- LAPD	YES	YES = Allows the changing of the layer 2 timer. NO = Do not allow the changing of the layer 2 timer (default).
...		
PARM	YES	Change system parameters.
FCDR	NEW	Use new CDR format (recommended for Japan).
OCAC	YES	Support the Original Carrier Access Code format.

Prompt	Response	Description
		NO = Do not support the Original Carrier Access Code format (default).
MTRO	(MR) PPM	Message Registration (default). Periodic Pulse Metering.

Table 100: LD 17 - Configure a Back-up D-Channel for Hong Kong.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	ADAN	Action device and number.
- ADAN	NEW BDCH xx	Add a Back-up D-Channel on logical port 0-63.
- PDCH	xx	Primary D-Channel associated with the Backup D-Channel. The value must be the same as the one entered for the Primary D-Channel. xx = 0-63.
- CTYP	MSDL	Card type where: MSDL = The NT6D80 Multi-purpose Serial Data Link or the NTBK51AA/NTBK51CA Downloadable D-Channel Daughterboard.
- GRP	0-4	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.
- CDNO	c	Card number for downloadable D-Channel Daughterboard for CS 1000E. Where c = 1-4: chassis on superloop, 1-10: cabinet or MG 1010 on superloop.
- 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 BCHL prompt, which follows later (either 0 or 1 must be entered when specifying the loop number used by the Backup D-Channel).
	1	Port number of the NTBK51BA, for CS 1000E.
- RCVP	YES	Auto-recovery to primary D-Channel.

Prompt	Response	Description
- BCHL	0-254	PRI loop number for Backup D-channel.
…		
- USR	PRI	This D-Channel is used for Primary Rate only.
- IFC	APAC	Asia Pacific ISDN interface.
- - CNTY	HKNG	Hong Kong
…		
PROG	NCHG MALE MCON	Progress signal. NCHG = No Change. This is the default for all Asia Pacific interfaces except Australia (and Japan and Singapore, which do not support the Progress signal for PRI). MALE = Alert message. MCON = Connect message. This is the default for the Australia interface. **Note:** The PROG prompt should not be configured for Japan and Singapore, since these countries do not support the Progress signal.
- RCVP	YES	Auto-recovery to primary D-Channel.
- BCHL	0-159	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.
	1	PRI/PRI2 loop number for Backup D-channel. For CS 1000E where l = 0, 4, 8, - 252: superloop number in multiples of 4.

Table 101: LD 17 - Configure nB+D for Hong Kong, Japan, New Zealand, and Malaysia. The International nB+D (INBD) package 255 must be enabled to configure nB+D for these interfaces.

Prompt	Response	Description
REQ	ADD	Add new data.
	CHG	Change existing data.
TYPE	ADAN	Action device and number.
- ADAN	NEW DCH xx	Add a primary D-Channel on logical port 0-63, for Large Systems.
- CTYP	MSDL	Card type where: MSDL = the NT6D80 Multi-purpose Serial Data Link or the NTBK51AA/NTBK51CA Downloadable D-Channel Daughterboard for Large Systems.

Prompt	Response	Description
- GRP	0-4	Network group number for Option 81C systems.
- DNUM	0-15	Device number for I/O ports. All ports on the MSDL card share the same DNUM. The MSDL card address settings must match the DNUM value.
	1-10	The card number of the Downloadable D-Channel Daughterboard, for CS 1000E Media Gateway cabinet and MG 1010.
	1-4	The card number of the Downloadable D-Channel Daughterboard for MG 1000E.
- PORT	0-3	Port number on the NT6D80 MSDL card, if the MSDL is used for D-Channel handling, for Large Systems.
	0-1	Port number of the NTBK51AA/NTBK51CA, if the NTBK51AA/NTBK51CA is used for D-Channel handling on large systems 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 BCHL prompt, which follows later (either 0 or 1 must be entered when specifying the loop number used by the Backup D-Channel).
	1	Port number of the NTBK51BA, for CS 1000E.
- 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	APAC	Asia Pacific ISDN interface.
- - CNTY	HKNG MSIA JAPN TCNZ	Hong Kong. Malaysia. Japan. New Zealand.
- DCHL	LOOP ID 0-159 (0)-15	Primary PRI loop number and interface identifier for the D-Channel for Large Systems. **Note:** The INBD package 255 must be enabled to allow this range of entry. If the INBD package is not enabled, the allowable entry would be the PRI loop number (0-159 for Large Systems, 1-9 for CS 1000E). Also, the subsequent PRI prompts would not be generated.
	LOOP ID 1-9 (0)-15	Primary PRI loop number and interface identifier for the D-Channel (for CS 1000E).

Prompt	Response	Description
- PRI	LOOP ID 0-159 (0)-15	Secondary PRI loop number and interface identifier for Large Systems. Prompted if INBD package 255 is enabled. The values entered must be different than those entered for the loop number and interface identifier at the DCHL prompt. The PRI prompt is generated until <CR> is entered.
	LOOP ID 1-9 (0)-15	Secondary PRI loop number and interface identifier (for CS 1000E). Prompted if INBD package 255 is enabled. The values entered must be different than those entered for the loop number and interface identifier at the DCHL prompt.
- PRI	<CR>	

Table 102: LD 17 - Configure the Bearer Capability for the Asia Pacific connectivity D-Channel.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	PARM	Make changes to system parameters.
...		
- BCAP	a...a	Bearer Capability. a...a = SPEECH (the default) or 3.1 Khz data.
...		

Table 103: LD 16 - Configure Asia Pacific ISDN PRI Route Data Block.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	RDB	Route Data Block.
CUST	0-99	Customer number as defined in LD 15.
ROUT	0-511	Route number for Large System 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	PRI	PRI digital trunk type for TAIW, HKNG and JAPN.
	PRI2	PRI digital trunk type for AUST, CHNA, INDI, INDO, SING, THAI, TCNZ, MSIA, and PHLP.

Prompt	Response	Description
...		
- ISDN	YES	Integrated Services Digital Network.
- MODE	PRA	ISDN PRI route.
- IFC	APAC	Asia Pacific ISDN interface.
- - CNTY		Country associated with the APAC interface.
	AUST CHNA HKNG INDI INDO JAPN MSIA PHLP SING TAIW TCNZ THAI	Australia. China. Hong Kong. India. Indonesia. Japan. Malaysia. Philippines. Singapore. Taiwan. New Zealand. Thailand.
...		
CPFXS	NO	Customer-defined Prefixes option. If CPFXS = NO, when constructing the Calling or Connected Line Identification, the prefixes are retrieved from the Route Data Block via the HNTN and HLCL prompts which follow. Enter NO for APAC.
	(YES)	If CPFXS = YES, when constructing the Calling or Connected Line Identification, the prefixes are retrieved from the Customer Data Block via the HNTN and HLCL prompts in LD 15, as is currently done. This is the default response.
HNTN	0-9999	This prompt applies to APAC only if CPFXS = NO. Home National Number. This number is similar to the PFX1 number prompted in LD 15. It is added to this overlay so that this prefix can be configured on a route basis. As is the case with PFX1, the HNTN prefix can be from one-to-four digits long.
HLCL	0-9999	This prompt applies to APAC only if CPFXS = NO. Home Location Number. This number is similar to the PFX2 number prompted in LD 15. It is added to this overlay so that this prefix can be configured on a route basis. As is the case with PFX2, the HLCL prefix can be from one-to-four digits in length.
...		

Prompt	Response	Description
ICOG	IAO ICT OGT	Incoming and/or Outgoing trunk. IAO = The trunk is Incoming and Outgoing. ICT = The trunk is Incoming only. OGT = The trunk is Outgoing only.
...		
ACOD	x...x	The Access Code for the trunk route. The ACOD must not conflict with the existing numbering plan.
CDR	YES	Call Detail Recording if the AOC display is desired in the call record.
- - OTL	YES	CDR on outgoing toll calls.
- - OAN	YES	CDR on all answered outgoing calls.
...		
MR	ENDC	AOC at end of call.
RUCS	1	Route unit cost - value received is treated as charged.
RUCF	1 0	Route unit conversion factor - no conversion required.
...		
MCTS	YES	Enable Malicious Call Trace signaling for AUST.
- MCTM	(0)-30	Malicious Call Trace request timer is defined in seconds. This is the disconnection delay which is used. It overrides T306 for calls to/from Malicious Call Trace capable phones (for AUST).
- MTND	(NO) YES	Malicious Call Trace disconnect delay for tandem calls (for AUST).

Table 104: LD 14 - Configure Asia Pacific ISDN PRI trunks.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	TIE COT DID	TIE trunk data block. Central Office Trunk data block. Direct Inward Dialing trunk data block.
		Note: Must match TKTP defined in LD 16.
TN		Terminal number

Prompt	Response	Description
	l ch	Loop and channel for digital trunks, where: l = Previously defined PRI/PRI2 loops. ch = channel 1-24 for 1.5 Mbit DTI/PRI or 1-30 for 2.0 Mbit DTI/PRI
CUST	0-99	Customer number as defined in LD 15.
RTMB		Route number and Member Number
	0-511 1-4000	Range for Large System Range for CS 1000E system.
TGAR	0 - (1) - 31	Trunk Group Access Restriction. The default of 1 automatically blocks direct access.

Table 105: LD 73 - Define the frame format and Grade of Service timers for a PRI/PRI2 loop.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	PRI PRI2	PRI loop. PRI2 loop.
FEAT	LPTI	Set the timers for PRI2 loop.
LOOP	xxx	PRI/PRI2 loop number.
MFF	(AFF) CRC	Alternate frame format (default). CRC-4 frame format.
ACRC	(NO) YES	NO = No automatic reporting of CRC-4 errors during transmission. YES = Automatic reporting of CRC-4 errors during transmission.
ALRM	(REG) ALT	REG = Regular mode of firmware alarm version (default). ALT = Alternate mode of firmware alarm version.
RAIE	(NO) YES	NO = Disable RAIE Group II alarm state (default). YES = Enable RAIE Group II alarm state.
...		
PERS	0-(50)-256	Persistence timer for Group II problems (2 ms increments).
CLEA	0-(50)-256	Clearance timer for Group II problems (2 ms increments).

Feature operation

No specific operating procedures are required to use this feature.

Australia ETSI

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

Basic ISDN services:

- 2.0 Mbit 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. Therefore, the NT6D11 DCHI is not supported on Large Systems.

Overlap Receiving is not supported.

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

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 for each call.

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, for each phone. 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 ta?le (refer to the paragraph above for more details).

Also, the system can now support 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 now supports multiple Prefix 1 (PFX1) and Prefix 2 (PFX2) contents, and multiple Home Location Codes (HLOCs) and Local Steering Codes (LSCs), on a DN or DN key basis.

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

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

Another new capability pertains to how the HLCL is constructed. A new prompt, DIDN (which signifies "use DN as a DID number") in 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 Primary Rate Interface (PRI2) package 154
- Overlap Signaling (OVLP) package 184
- International Primary Rate Access (IPRA) package 202
- Multi-Purpose Serial Data Link (MSDL) package 222

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

For the Advice of Charge capability:

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

For the Malicious Call Trace capability:

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

For ISDN Basic Rate Interface Trunking connectivity:

- Basic Rate Interface (BRI) package 216
- Basic Rate Interface Trunk (BRIT) package 233

Feature implementation

Task summary list

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

1. Table 106: LD 17 - Configure a PRI2 loop for the ETSI Australian ISDN connectivity. on page 257
2. Table 107: LD 17 - Configure the D-channel for ETSI Australian ISDN connectivity. on page 257
3. Table 108: LD 16 - Configure the ETSI Australian ISDN PRI2 Route Data Block. on page 259
4. Table 109: LD 14 - Configure the Australia ETSI ISDN PRI2 trunks. on page 260
5. Table 110: LD 17 - Configure Advice of Charge for Australia ETSI. on page 260
6. Table 111: LD 15 - Allow Charge Display and CDR Charge. on page 261
7. Table 112: LD 10 - Assign meters to analog (500/2500-type) phones. on page 261
8. Table 113: LD 11 - Assign meters to digital proprietary phones. on page 261
9. Table 114: LD 16 - Allow Advice of Charge on the route configured for Australia ETSI. on page 261

Table 106: 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	0-159	PRI2 loop number.

Table 107: 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		

Prompt	Response	Description
	MSDL	Multi-purpose Serial Data Link card or Downloadable D-Channel Daughterboard for Large Systems.
- GRP	0-4	Network group number (Large Multi-group)
- DNUM	0-15	Device number for I/O ports for Large Systems. All ports on the MSDL card share the same DNUM. The MSDL card address settings must match the DNUM value.
	0-10	The card number of the Downloadable D-Channel Daughterboard, for CS 1000E Media Gateway cabinet and MG 1010
	0-4	The card number of the Downloadable D-Channel Daughterboard for MG 1000E.
- PORT	0-3 1	Port number on the MSDL card (Large Multi-group).
- 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	EURO	EuroISDN interface.
- - CNTY	EAUS	Australia ETSI.
- - DCHL	0-159	PRI2 loop number for D-channel for Large Systems.
- CNEG	(1) 2	Options for outgoing Channel Negotiation. Option 1: Channel is non-negotiable. Option 2: The Channel listed is preferred, but negotiable.
- RLS	xx	Software Release of the far-end switch.
- RCAP		Remote capabilities, prompted to configure the Connected Line ID Presentation supplementary service.
	(COLP) XCOL MCID XMCI	CLID Presentation supported. Remove COLP. Allow Malicious Call Trace. Remove Malicious Call Trace.
- 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.

Prompt	Response	Description
- - T310	(30)-100	Maximum time in seconds between an incoming CALL PROCEEDING message and the next incoming message.
- - INC_T306	0-(2)-30	Variable timer, in seconds, for received DISCONNECT message on incoming calls allowing in-band tone to be heard. The network will stop sending after this timer times out. The value is stored in two-second increments, which are rounded up.
- - OUT_T306	0-(30)-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 108: 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	0-99	Customer number as defined in LD 15.
ROUT	0-511	Route number for Large System 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.
...		
MCTS	YES	Enable MCT signaling.
...		

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

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	TIE COT DID	TIE trunk data block. Central Office Trunk data block. Direct Inward Dialing trunk data block.
		Note: Must match TKTP defined in LD 16.
TN		Terminal number
	l ch	Loop and channel for digital trunks Large Systems, where: Previously defined PRI2 loops. Channel 1-30
CUST	0-99	Customer number as defined in LD 15.
RTMB		Route number and Member Number
	0-511 1-4000	Range for Large System. Range for CS 1000E system.
TGAR	0 - (1) - 31	Trunk Group Access Restriction The default of 1 automatically blocks direct access.

Note:

The MR/PPM package 101 must be equipped on the system.

Table 110: 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.

Prompt	Response	Description
		The default is MR, for Message Registration.

Table 111: LD 15 - Allow Charge Display and CDR Charge.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	PPM	Periodic Pulse Data.
OPT	CHDA	Charge Display Allowed.
...		
UCST	(0)-9999	Unit cost for PPM.

Table 112: LD 10 - Assign meters to analog (500/2500-type) phones.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	500	Analog telephone.
TN	l s c u	Terminal Number for Large System and CS 1000E.

Table 113: LD 11 - Assign meters to digital 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	l s c u	Terminal Number for Large System and CS 1000E.

Table 114: 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	0-99	Customer number as defined in LD 15.
ROUT	0-511	Route number for Large System and CS 1000E system.

Prompt	Response	Description
TKTP	TIE COT DID	TIE trunk type. Central Office Trunk type. Direct Inward Dialing trunk type.
DTRK	YES	Digital trunk route.
- DGTP	PRI2	2.0 Mbit PRI digital trunk type.
ISDN	YES	Integrated Services Digital Network.
- MODE	PRA	ISDN PRI route.
- IFC	EURO	EuroISDN interface.
- - CNTY	EAUS	Australia ETSI.
CDR	YES	Include AOC information in the CDR ticket.
- OAL	YES	CDR on all answered outgoing calls.
- - OTL	YES	CDR on all outgoing toll calls.
MR	STAC DURC ENDC	Define AOC at call set-up. Define AOC during the call. Define AOC at end of call.
DSPD	(NO) YES	(Do not) display the charge during the call.
RUCS	0-9999	Route unit cost.
RURC	X Y	Route unit reference cost. Formula is $X*10^{(-Y)}$ where X = 0-9999, Y = 0-3. The default value for X is the value that is entered for RUCS.
RUCF	1 0	Route unit conversion factor. 0 = No conversion is required.
DSPT	0-(10)-60	Charge display timer.

Feature operation

No specific operating procedu?es are required to use this feature.

AXE-10 (Australia), non-Asia Pacific connectivity

Description

This feature allows ISDN PRI connectivity between the system and the AXE-10 for Australia, using switched-circuit mode connections. The design in based on the international 30B + D PRI configuration.

Basic ISDN PRI connectivity between the system and the AXE-10 will support:

- Basic Call Service
- Connected number delivery
- Calling Line Identification, both public and private
- Increased CDR accuracy
- COT, DID, DOD, and TIE call types
- 64 Kbit/s clear-bearer capability
- Channel Negotiation
- Overlap Sending (but not overlap receiving)
- Flexible Numbering Plan
- Advice of Charge at end of call

Operating parameters

The only connection that is supported is the switched circuit mode connection.

The supplementary service control messages not relating to a call are not supported.

The system does not send the High Layer Compatibility Information Element in the SETUP message to the AXE-10, nor does it send the Restart Request and Restart Global Call States.

Overlap Receiving is possible only if the AXE-10 conforms to the guidelines contained in CCITT's preliminary Q.931 section 5.0.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature requires the following packages:

- Digit Display (DDSP) package 19
- Integrated Services Digital Network (ISDN) package 145
- 2.0 Mbit Primary Rate Interface (PRI2) package 154
- Flexible Numbering Plan (FLEN) package 160
- Overlap Signaling (OVLP) package 184
- International Primary Rate Access (IPRA) package 202

Feature implementation

Task summary list

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

1. Table 115: LD 17 - Define system support for the AXE-10 Australia connectivity. on page 264
2. Table 116: LD 16 - Configure the routes for the AXE-10 Australia connectivity. on page 265

The following procedures assume that a D-channel has been configured. Refer to the Data Administration chapter overlay procedures.

Table 115: LD 17 - Define system support for the AXE-10 Australia connectivity.

Prompt	Response	Description
REQ	CHG	Change existing data block.
TYPE	CFN	Configuration record.
- ADAN	CHG DCH x	Change a D-channel on logical port 0-63 (For Large Systems).
...		

Prompt	Response	Description
- IFC	AXEA	Ericsson AXE-10 for Australia.
- CNEG	(1) 2	The required channel negotiation option, either exclusive (the default) or preferred.

Table 116: LD 16 - Configure the routes for the AXE-10 Australia connectivity.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	RDB	Route Data Block
CUST	0-99	Customer number as defined in LD 15.
ROUT	0-511	Route number for Large System and CS 1000E system.
IFC	AXEA	Ericcson AXE-10 for Australia.

Feature operation

No specific operating procedures are required to use this feature.

AXE-10 (Sweden), non-Asia Pacific connectivity

Description

This feature allows ISDN PRI connectivity between the system and the Ericsson AXE-10 for Sweden, using switched-circuit mode connections. The design in based on the international 30B + D PRI configuration.

Basic ISDN PRI connectivity between the system and the AXE-10 will support:

- Basic Call Service
- Connected number delivery
- Calling Line Identification, both public and private
- Increased CDR accuracy
- COT, DID, DOD, and TIE call types

- 64 Kbit/s clear-bearer capability
- Channel Negotiation
- Overlap Sending (but not overlap receiving)
- Flexible Numbering Plan
- Advice of Charge at end of call

Operating parameters

The only connection that is supported is the switched circuit mode connection.

The supplementary service control messages not relating to a call are not supported.

The system does not send the High Layer Compatibility Information Element in the SETUP message to the AXE-10, nor does it send the Restart Request and Restart Global Call States.

Overlap Receiving is possible only if the AXE-10 conforms to the guidelines contained in CCITT's preliminary Q.931 section 5.0.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature requires the following packages:

- Digit Display (DDSP) package 19
- Integrated Services Digital Network (ISDN) package 145
- 2.0 Mbit Primary Rate Interface (PRI2) package 154
- Flexible Numbering Plan (FLEN) package 160
- Overlap Signaling (OVLP) package 184
- International Primary Rate Access (IPRA) package 202

Feature implementation

Task summary list

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

1. Table 117: LD 17 - Define the system support for the AXE-10 Sweden connectivity. on page 267
2. Table 118: LD 16 - Configure the routes for the AXE-10 Australia connectivity. on page 267

The following procedures assume a D-channel has been configured.

Table 117: LD 17 - Define the system support for the AXE-10 Sweden connectivity.

Prompt	Response	Description
REQ	CHG	Change existing data block.
TYPE	CFN	Configuration record.
- ADAN	CHG DCH x	Change a D-channel on logical port 0-63.
- IFC	AXES	Ericsson AXE-10 for Sweden.
- CNEG	(1)2	The required channel negotiation option, either exclusive (the default) or preferred.

Table 118: LD 16 - Configure the routes for the AXE-10 Australia connectivity.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	RDB	Route Data Block
CUST	0-99	Customer number as defined in LD 15.
ROUT	0-511	Route number for Large System and CS 1000E system.
IFC	AXEA	Ericcson AXE-10 for Australia.

Feature operation

No specific operating procedures are required to use this feature.

NEAX-61 (New Zealand), non-Asia Pacific connectivity

Description

This feature provides an ISDN Primary Rate Interface (PRI) between the system and New Zealand NEAX-61. The design in based on the international 30B + D PRI configuration.

The system to New Zealand NEAX-61 ISDN PRI Connectivity supports the following:

- Basic Call Service
- Calling Line Identification (public and private)
- Overlap Sending
- COT, DID, DOD, and TIE trunk call types
- 64 Kbit/s clear bearer capability
- Channel Negotiation
- nB+D, up to 120 B-channels/four interfaces
- Public Switched Telephone Network (PSTN) three-party conferencing
- Flexible Numbering Plan
- Malicious Call Trace

Note that Electronic Switched Network (ESN) cannot be supported across the interface for private network applications. Coordinated Dialing Plan (CDP) and Network Automatic Route Selection/Basic Automatic Route Selection (NARS/BARS) can access the public network, but private network features are not supported. The NEAX-61 does not support the required numbering plans to allow CDP or UDP (location code dialing).

New Zealand Supplementary ISDN Services

Malicious Call Trace

This subscription supplementary service allows the system to inform the public network in New Zealand that the current call is malicious. The NEAX-61 automatically records al? calls to a number subscribing to the MCT feature. If an MCT is activated, the public network marks the call as being malicious.

This service is activated by ISDN messaging. It is a user-triggered feature, requiring the called party to press a key or dial codes.

PSTN Three Party Conference

General

This subscription supplementary service allows the New Zealand NEAX-61 to provide a Three Party Conference. Rather than having the system provide the conference capability and the two trunks, it allows a caller to conference in two external callers using one trunk. This supplementary feature is activated using the Trunk Hook Flash (THF) or Centrex Switch Hook Flash operation, and is user invoked.

To illustrate this operation, the PBX supported conference and PSTN supported conference is illustrated in Figure 13: PBX and PSTN supported conference on page 269.

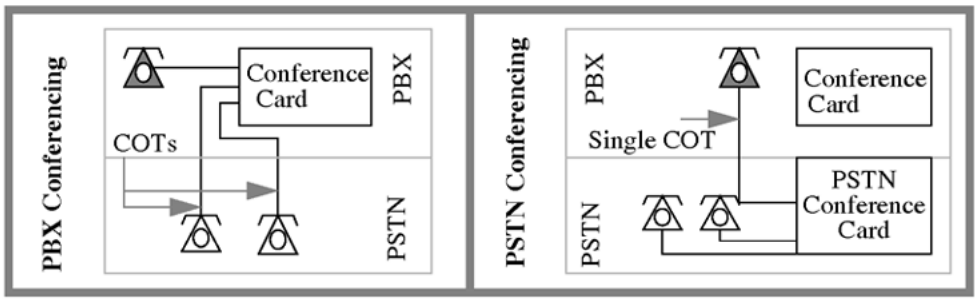


Figure 13: PBX and PSTN supported conference

This operation is a stand-alone feature as the network wide trunk hook flash is not supported. Note also that from the New Zealand NEAX-61 side, MCT and PSTN conferencing are mutually exclusive features.

Call Detail Recording for PSTN Three Party Conferencing

PSTN Three Party Conferencing does not have the capability of providing precise Call Detail Recording (CDR) records. The NEAX-61 does not send messages to indicate that one of the two external parties has released, or that the attempt to complete the conference through THF was unsuccessful, due to the third party not being established. Therefore, if party "A" on the system called party "B" in the public network, and later called party "C", no notice is given whether party "C" answered. As well, if either of "B" or "C" releases, the system will not be informed.

As a result, no attempt is made to make the CDR records match the normal "S" and "E" records of conference calls with two trunks. Instead of two "S" records at the start of the conference, a single "Y" record is issued to indicate a PSTN feature is invoked. This "Y" record includes

all digits dialed to access the third party. The final record, produced on call clearing, remains as an "N" record, and includes information from the original call.

Operating parameters

The following hardware is used to implement the interface:

- For Large Systems, the layer one function is provided by the circuit pack NT8D72 (2.0 Mbit PRI card for Large Systems). Layer two and three are provided by the Multi-purpose Serial Data Link (MSDL – NT6D80) for these systems. The NTRB53 Clock Controller will be used to provide clock synchronization.
- For CS 1000E, the layer one function is provided by the Downloadable D-channel Primary Rate motherboard, circuit pack NTBK50AA. Layer two and three are provided by the Downloadable D-channel daughterboard, circuit pack NTBK51AA/NTBK51CA. The Clock Controller daughterboard NTAK20BB is used to provide clock synchronization.

Feature interactions

Calling Line Identification

This feature is provided through the Calling Party Number. When the system is connected to NEAX-61, the feature's format is: ZZZZ + YYYY +DDDDDDD

The Customer Data block contains the prompts PFX1 and PFX2 for the responses "ZZZZ" and "YYYY" respectively. "DDDDDDD" is a system internal (set) DN up to seven digits. The prompts PFX1 and PFX2 must be defined by the technician so that the combination of PFX1, PFX2, and the originating DN provides a correct number for the terminating party to dial in order to reach the originating party. If no digits are configured for either of the prefixes, that part of the number will not be included in the Calling Party Number Information Element (IE).

Call Redirection (Call Forward, Hunt, CFNA) and Modification (Conference, Transfer)

All of the following actions apply to the call itself; a notice of redirection or modification (changes in CLID, for example.) is not transmitted. It is possible to Call ?orward a phone on a system to an external number over an NEAX-61 Primary Rate Interface (this operates the same as over any other Central Office trunk).

A phone on the system can Hunt or CFNA to an external number over an NEAX-61 Primary Rate Interface (this operates the same as over any other Central Office trunk).

It is possible to transfer or conference a call on the system to an external number over an NEAX-61 Primary Rate Interface (this operates the same as over any Central Office trunk).

Access restrictions can block some transfers from being completed, but this is consistent with other interfaces.

Outpulsing of Asterisk and Octothorpe (OPAO)

As the outpulsed digits can include the octothorpe "#" and asterisk "*", this feature is used to provide the function. The impact on other applications of OPAO is negligible.

Centrex Switchhook Flash

The Centrex Switchhook Flash feature provides an analog trunk line break to initiate public network features. This capability is expanded to include ISDN signaling to the New Zealand NEAX-61. As the ISDN equivalent is carried out by D-channel signaling, only the triggering of the flash by a phone is shared with the analog Centrex flash; signaling through ISDN messaging is independent.

Feature interactions Interworking within the Meridian Customer Defined Network

The following group of features are supported in part by the system to New Zealand NEAX-61 ISDN PRI Connectivity. These cannot be supported across the NEAX-61 interface, but will interwork with a call coming in from the PSTN. The portion of the function local to the Meridian Customer Defined Network (MCDN) is supported there.

BRI phones

The NEAX-61 interface will support calls from BRI phones within the MCDN. These will be able to call out to and receive calls from the Public Switched Telephone Network (PSTN). Operation will be as though the call was a tandem trunk call, with all IEs supported on the interface passed transparently.

BRI Trunks

BRI trunks are supported by the NEAX-61 interface.

Calling Party Name Display

This feature is offered within the MCDN. Names can be assigned to various items, including route access codes. An incoming call across this route will have the name stored against the route displayed. This will continue, however, CPND received from the NEAX-61 is not currently supported, and sending CPND to the NEAX-61 is also not supported at this time.

Calling Line Identification (CLID) Enhancements

MCDN Component CLID enhancements for a redirected call will be seen within the MCDN part of the call. If an incoming PSTN call is redirected, this will be displayed on the terminating device as applicable. If a call is redirected off the MCDN, information up to and including the redirection off of the network will be provided. This feature cannot be offered across the

NEAX-61 interface as it depends on an IE not supported by the NEAX-61 (the Redirecting Number IE).

DPNSS Gateway

The NEAX-61 interface interacts with the DPNSS gateway consistently as with other PSTN interfaces. The IEs supported on the NEAX-61 interface which have equivalents in DPNSS interwork on an IE-by-IE basis. IEs that have no matching DPNSS element are discarded. Supported IEs with DPNSS equivalents pass through with appropriate mapping.

Coordinated Dialing Plan (CDP)

A CDP call can access the trunk. However, since neither the private plan nor the CDP numbering type is supported by the NEAX-61, these get converted to unknown type and plan. This applies to both the called and calling number plan and type.

Since the normal use of CDP does not provide a number that can be dialed when calling the public network, the NEAX-61 will replace the CLID with the default number for the system. Normal use of steering codes with Distant Steering Codes (DSCs) and Trunk Steering Codes (TSCs) will be maintained. Use of a Digital Manipulation Index (DMI) to convert the number as required can be done.

Basic Automatic Route Selection (BARS)

This feature has no interactions with the NEAX-61 with the exceptions of: the NPA?and NXX cannot be used on this interface, as the public network cannot accept them; and SPNs get converted to "unknown numbering plan and type, as the ESN SPN is a North American concept.

Network Automatic Route Selection (NARS)

All BARS concerns exist for NARS as well. In addition, location codes should be used with care, since the networking features do not accept a Digit Manipulation Index (DMI) used to insert an ESN access code. Therefore, the only supported usage has the trunks effectively treated as though they were non-ISDN.

Network Attendant Service (NAS)

NAS is not supported across the interface except in a few simple ways. An incoming call from the NEAX-61 can be routed using NAS. NAS routing will allow a call to the attendants in night service to be rerouted across the NEAX-61, but no NAS function is possible. The call will be treated as though the call was placed directly, rather than through NAS. Thus, NAS attendant control, signaling, and recall and so forth are not possible.

Network Call Redirection, Network Call Forward, Network Call Forward No Answer - MCDN Component

These features cannot be offered to the NEAX-61 or across its interface. However, they are offered within the MCDN network, and the aspects applicable within the network still apply.

If an incoming call from the NEAX-61 is MCDN call redirected, all signaling information within the system network will be consistent with current operations, but redirection information will be discarded at the system to NEAX-61 interface. If a call originating within the MCDN is redirected to the PSTN, all redirection information up to and including the redirection to the public network will be provided through the network.

Trunk Optimization - MCDN Component

Trunk Optimization within the MCDN will be supported by the appropriate interfaces. It will not be supported across the NEAX-61 interface, but will interwork transparently. In other words, trunk optimization will not be observed at the NEAX-61 interface; the call will appear to be static, not undergoing any modifications.

Virtual Network Services (VNS)

VNS uses a dedicated, MCDN level of service DCH to control a number of temporary B-channels, seized and used on demand. The ISDN signaling requests a connection (SETUP), and a returned FACILITY has a DN that is to be recognized as a VNS virtual DN on the terminating switch. On this VNS DN being received, the original call is completed, since a B-channel has now been obtained.

An NEAX-61 trunk can act as a virtual B-channel.

Feature packaging

There is no new software package for this feature.

The following packages are required for customers who choose to have software that allows them to connect the system over ISDN PRI to the NEAX-61 (New Zealand) interface:

- Digit Display (DDSP) package 19 for Calling Line Identification on ISDN PRI
- Outpulsing of Asterisk "*" and Octothorpe "#" (OPAO) package 104
- Integrated Services Digital Network (ISDN) package 145
- 2.0 Mbit Primary Rate Access (PRI2) package 154
- International Primary Rate Access (IPRA) package 202
- New Format Call Detail Recording (FCDR) package 234

To carry out overlap sending to the PSTN, the following packages and their prerequisites are required:

- Overlap Signalling (OVLP) package 184
- One or more of:
  - Basic Alternate Route Selection (BARS) package 57
  - Network Alternate Route Selection (NARS) package 58 or
  - Coordinated Dialing Plan (CDP) package 59

If the user subscribes to the following services on the PSTN, the ISDN International Supplementary (ISDNS) package 161 is required. The following packages are required for each feature:

- Malicious Call Trace (MCT) – the Malicious Call Trace (MCT) package 107
- PRA Groups (nB+D) – the International nB+D (INBD) package 255 is required for the following interfaces only: D70, JAPN, HKNG, TCNZ, and MSIA.
- PSTN Three Party Conferencing – the Trunk Hook Flash (Centrex) (THF) package 157. For FFC access to this feature, the Flexible Feature Codes (FFC)?package 139, Controlled Class of Service (CCOS) package 81, and International Supplementary Services (SUPP) package 131 are required

Feature implementation

Task summary list

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

1. Table 119: LD 17 - Configure an ISDN interface supporting the Telecom New Zealand ISDN protocol. on page 274
2. Table 120: LD 16 - Configure a COT, DID/DOD, or TIE route using the Telecom New Zealand (TCNZ) protocol. on page 276

Table 119: LD 17 - Configure an ISDN interface supporting the Telecom New Zealand ISDN protocol.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	CFN	Configuration data block.
- ADAN	CHG DCH x	Change a D-channel on logical port 0-63 (For Large Systems).
- CTYP	MSDL	TCNZ requires the D-channel be defined on an MSDL.
- DNUM	0-15	Device number on the MSDL card for Large Systems.
- PORT	0-3	Port number on the MSDL.
- GRP	0-4	Network group number for Large Systems.
CDNO	c	Card number for downloadable D-Channel Daughterboard for CS 1000E. Where c = 1-4: chassis on superloop, 1-10: cabinet or MG 1010 on superloop.
- DES	aaaaaa	Alphanumeric designator.

Prompt	Response	Description
- DPNS	(NO) YES	Digital Private Network Signaling.
- USR	PRI	D-channel mode.This D-channel is used for Primary Rate only.
- IFC	TCNZ	Telecom New Zealand.
-DCHL	0-159 (0)-3	TCNZ loop and interface ID for Large Systems.
	0-255 (0)-3	TCNZ loop and interface ID for CS 1000E.
- PRI2	0-159 0-3	Secondary PRI2 loops for nB+D, plus sequence. Note that the D-channel is not necessarily on IFC ID 0; this is set by service change.
- OTBF	1-(32)-127	Number of output request buffers.
- CNEG	(1) 2	Options for outgoing Channel Negotiation. Option 1: Channel is non-negotiable. Option 2: The Channel listed is preferred, but negotiable.
- RCAP	aaa	Remote DCH capabilities, as applicable.
- OVLS	YES	Allow overlap sending.
- - OVLT	1-8, 0	Time the sending side has to wait between INFO messages. "0" means send immediately.
- TIMR	(NO) YES	Change protocol timer value.
- - T310	10-60	Timer used to determine how long the switch can wait for the response message when the QSIG outgoing call is in the outgoing call processing state.
- - INC_T306	0 (2)...T306	Variable timer for received DISCONNECT message on incoming calls, allowing in-band tone to be heard when sent by the network. The network will stop sending tone after T306 expires, so the maximum time will be T306. (T306 is the duration of the T306 Network timer in seconds) Entered in two-second increments. TCNZ interface: T306 = 30 sec.
- - OUT_T306	0 2...(T306) (T306 is the duration of the T306 Network timer in seconds)	Variable timer for received DISCONNECT message on outgoing calls, allowing in-band tone to be heard when sent by the network. The network will stop sending tone after T306 expires, so the maximum time will be T306. (T306 is the duration of the T306 Network timer in seconds) Entered in two-second increments. TCNZ interface: T306 = 30 sec.
- LAPD	(NO) YES	Change the Link Access Protocol for D-channel parameters.

Table 120: LD 16 - Configure a COT, DID/DOD, or TIE route using the Telecom New Zealand (TCNZ) protocol.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	RDB	Route Data Block
CUST	0-99	Customer number as defined in LD 15.
ROUT	0-511	Route number for Large System and CS 1000E system.
DGTP	PRI2	Type of digital route.
ISDN	YES	Default for a PR!2 route.
MODE	PRA	Primary Rate Access.
IFC	TCNZ	Telecom New Zealand (NEAX-61).
CTYP	UKWN INTL NPA NXX	Only these call types are supported for TCNZ interface TIE routes.
OPD	(YES) NO	Outpulsed digits in CDR.
NDP	INC 1-32 EXC 1-32	Number of printed digits. INC 1-32 = Output the first 1-32 digits. EXC 1-32 = Suppress the last 1-32 digits. If REQ = NEW, output all digits and suppress none, if REQ = CHG, leave unchanged.
CDRX	YES (NO)	X records produced on call transfer.
CDRY	(NO) YES	CDR Public Network Feature Invoke records will (will not) be generated when the feature is invoked. Default is NO.
OPA	(NO) YES	Generate CDR records for PPM pulses.
...		
MCTS	(NO) YES	Send a Malicious Call Trace request to the public network, if MCT is invoked in the private network on current switch.

Feature operation

No specific operating procedures are required to use thi? feature.

Numeris VN3 (France)

Description

This feature provides ISDN PRI connectivity between the system software and the ISDN access software developed by France Telecom (Numeris). The connectivity is based on International 30B+D ISDN Primary Rate Access development.

The following basic services are provided for this initial offering of the ISDN PRI system to Numeris connectivity:

- Basic Call Service
- Calling Line Identification (public and private)
- Confidentiality of Calling Line Identification
- COT, DID, DOD, TIE trunks call types
- 64 Kbit/s clear bearer capability
- Channel Negotiation
- Call Restriction
- Flexible Numbering Plan
- Advice of Charge at end of call, and during call

Operating parameters

The only connection that is supported is the switched circuit mode connection. The X.25 packet mode connection, through the B-channel, the packet mode temporary signaling connection, and the semi-permanent circuit-mode connection are not supported.

The NUMERIS supplementary service control messages not relating to a call are not supported.

The Calling Party Number Information Element can have two occurrences, with two different contents, in a SETUP message sent from the network to the user. For CLID display processing, the system interprets only the contents of the first Calling Number Information Element.

For direct trunk access code dialing, digit analysis is not performed for the enbloc dialing. A time-out rather than an "#" is used as an indication that dialing has been completed.

Overlap Receiving has not been implemented for NUMERIS.

The maximum delay to send an RAI signal upon detection of an LOS, LFAS, LMAS, and AIS is 100 milliseconds. The same applies to the termination of RAI transmission when the alarm condition clears up.

No incoming/outgoing calls can be initiated during an RAI, LFAS, LMAS, AIS, LOS, or ERR alarm condition.

Incoming/outgoing calls must be allowed within 100 milliseconds of cessation of an RAI, AIS, LOS, LMAS, or LFAS alarm condition.

Established calls must not be disconnected by layer 1 processing during an RAI, AIS, LOS, LMAS, LFAS, or ERR alarm condition.

An RAI signal must be sent if an ERR condition of 10-3 lasts more than four to five seconds.

The RAI transmission must cease after an ERR condition of 10-4 lasts for more than four to five seconds.

The PRI2 link must not be placed in out-of-service condition due to an RAI, LFAS, LMAS, AIS, LOS, or ERR alarm condition.

If the PRI2 pack is manually disabled, it must go into the out-of-service state.

Feature packaging

This feature requires the following packages:

- Digit Display (DDSP) package 19 (required for CLID)
- Integrated Services Digital Network (ISDN) package 145
- 2.0 Mbit Primary Rate Interface (PRI2) package 154
- International Primary Rate Access (IPRA) package 202

Feature interactions

There are no feature interactions associated with this feature.

Feature implementation

Task summary list

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

1. Table 121: LD 17 - Define system support for NUMERIS. on page 279
2. Table 122: LD 16 - Configure routes for NUMERIS. on page 279
3. Table 123: LD 73 - Define the alarm handler, disable or enable the Grade of Service processing in the software, and define the number of seconds before the firmware sends an ERR message. on page 279

The following procedures assume that a D-channel has been configured.

Table 121: LD 17 - Define system support for NUMERIS.

Prompt	Response	Description
REQ	CHG	Change existing data block.
TYPE	CFN	Configuration record.
- ADAN	CHG DCH x	Change a D-channel on logical port 0-63 (For Large Systems).
- IFC	NUME	Numeris for France.
- CNEG	(1) 2	The required channel negotiation option, either exclusive (the default) or preferred.

Table 122: LD 16 - Configure routes for NUMERIS.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	RDB	Route Data Block.
CUST	0-99	Customer number as defined in LD 15.
ROUT	0-511	Route number for Large System and CS 1000E system.
IFC	NUME	Numeris for France

For the following table for LD 73, define the alarm handler in the firmware, disable or enable the Grade of Service processing in the software, and define the number of seconds before the firmware sends an ERR message.

Table 123: LD 73 - Define the alarm handler, disable or enable the Grade of Service processing in the software, and define the number of seconds before the firmware sends an ERR message.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	PRI2	2.0 Mbit Primary Rate Interface data block.

Prompt	Response	Description
...		
ALRM	REG ALT	REG = Regular firmware alarm handler (RAI transmission is controlled by software). ALT = Alternate firmware alarm handler (Immediate transmission of RAI by firmware).
...		
NOOS	 (NO) YES	Grade of service feat. NO = Enables current grade of service feat. YES = Alternate grade of service feat.
...		
RATS	1-(10)-15	The number of consecutive seconds the firmware has to check and validate error rate condition.

Feature operation

No specific operating procedures are required to use this feature.

SwissNet 2 (Switzerland)

Description

This feature provides basic ISDN PRI connectivity between the system and a Central Office meeting SwissNet 2 specifications in Switzerland.

Following is the set of call services provided in the initial ISDN PRI offering for the system to SwissNet basic connectivity:

- Basic Call Service (3.1kHz, speech)
- Calling Line Identification (CLID) and Calling Line Restriction (CLIR)
- COT, DID, DOD, TIE call types
- 64 Kbit/s clear bearer capability
- Channel Negotiation
- Flexible Numbering Plan

- Overlap Sending
- Advice of Charge during call

Operating parameters

The only connection that is supported is the switched circuit mode connection. The X.25 packet mode connection, through the B-channel, the packet mode temporary signaling connection, and the semi-permanent circuit-mode connection are not supported.

For Channel Negotiation to operate correctly, customers on the system cannot share a PRI to the SwissNet interface.

For Calling Line Identification to work correctly across a network, the nodes should be connected using a Coordinated Dialing Plan, with DNs of the same fixed lengths on each node.

Overlap Receiving has not been implemented for the SwissNet interface.

ISDN PRA to SwissNet Connectivity is not supported on DPNSS1.

Feature interactions

Calling Line Identification

The CLID will have the following format for the system ISDN PRA to SwissNet Connectivity: zzzz+yyyy+ddddddd where zzzz and yyyy are entered in response to prompts PFX1 and PFX2 respectively, in LD 15, and ddddddd is an internal DN up to seven digits long. If no digits are configured for either PFX1 and PFX2, then the Calling Party Number IE will not contain the entire CLID number.

Call Detail Recording (CDR)

For calls made over an ISDN PRI to SwissNet interface equipped with the Advice of Charge at End of Call?feature, all Start and Transfer records will show "OVF99" in the overflow count field. This is done to indicate that there will be charge units to be shared among successive call controllers at the end of a call.

Virtual Network Services (VNS)

VNS is supported over an ISDN PRI to SwissNet interface.

Feature packaging

This feature requires the following packages:

- Call Detail Recording, Teletype Terminal (CTY) package 5
- Digit Display (DDSP) package 19 (required for CLID)
- PPM/Message Registration (MR) package 101
- International Supplementary Features (SUPP) package 131
- Integrated Services Digital Network (ISDN) package 145
- 2.0 Mbit Primary Rate Interface (PRI2) package 154
- ISDN Supplementary Features (ISDNS) package 161
- International Primary Rate Access (IPRA) package 202
- Multi-purpose Serial Data Link (MSDL) package 222 (if required)

Feature implementation

Task summary list

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

1. Table 124: LD 17 - Define the ISDN PRI to SwissNet connectivity. on page 282
2. Table 125: LD 16 - Define the ISDN PRI to SwissNet connectivity. on page 283

The following procedures assume that a D-channel has been configured.

Table 124: LD 17 - Define the ISDN PRI to SwissNet connectivity.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	CFN	Configuration Record.
- ADAN	CHG DCH x	Change a D-channel on logical port 0-63 (For Large Systems).
- IFC	SWIS	Interface type for SwissNet connectivity.
- CNEG	(1) 2	The required channel negotiation option, either exclusive (the default) or preferred.

Table 125: LD 16 - Define the ISDN PRI to SwissNet connectivity.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	RBD	Route Data Block.
CUST	0-99	Customer number as defined in LD 15.
ROUT	0-511	Route number for Large System and CS 1000E system.
IFC	SWIS	Interface type for SwissNet connectivity.

Feature operation

No specific operating procedures are required to use this feature.

SYS-12 (Norway)

Description

This feature allows ISDN PRI connectivity between the system and the Norwegian SYS-12, using switched-circuit mode connections. The ISDN PRI connectivity between the system and the SYS-12 supports:

- Basic call service
- Connected number delivery
- Calling Line Identification, both public and private
- Increased CDR accuracy
- COT, DID, DOD, and TIE call types
- 64 Kbit/s clear-bearer capability
- Channel Negotiation
- overlap sending (but not overlap receiving)
- Flexible Numbering Plan

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

For PRI connectivity, the following packages are required:

- Integrated Service Digital Network (ISDN) package 145
- 2.0 Mbit Primary Rate Access (PRI2) package 154
- 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 126: LD 17 - Define the ISDN PRI to SYS-12 connectivity. on page 284
2. Table 127: LD 16 - Define the ISDN PRI to SYS-12 connectivity. on page 285

The following procedures assume that a D-channel has been configured.

Table 126: LD 17 - Define the ISDN PRI to SYS-12 connectivity.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	CFN	Type of data block.
- ADAN	CHG DCH x	Change a D-channel on logical port 0-63 (For Large Systems).

Prompt	Response	Description
- IFC	SS12	Interface type for SYS-12 connectivity.
- CNEG	(1) 2	The required channel negotiation option, either exclusive (the default) or preferred.

Table 127: LD 16 - Define the ISDN PRI to SYS-12 connectivity.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	RDB	Route Data Block.
CUST	0-99	Customer number as defined in LD 15.
ROUT	0-511	Route number for Large System and CS 1000E system.
IFC	SS12	Interface type for SYS-12 connectivity.

Feature operation

No specific operating procedures are required to use this feature.

1TR6 (Germany)

Description

This feature will allow the interconnection of the system with the 1TR6 protocol. Basic connectivity between the system and the 1TR6 DCH supports:

- Basic Call Service
- Connected number delivery
- Calling Line Identification, both public and private
- Increased CDR accuracy
- COT, DID, DOD, and TIE call types
- 64 Kbit/s clear-bearer capability
- Channel Negotiation

- Overlap sending (but not overlap receiving)
- Flexible Numbering Plan
- Advice of Charge at end of call
- Call restrictions to tandem calls

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature requires the package Periodic Pulse Metering/Message Registration (MR) package 101.

For PRI connectivity the following packages are required:

- Integrated Service Digital Network (ISDN) package 14
- 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 128: LD 17 - Define the ISDN PRI to 1TR6 connectivity. on page 287
2. Table 129: LD 16 - Define the ISDN PRI to 1TR6 connectivity. on page 287

The following procedures assume that a D-channel has been configured.

Table 128: LD 17 - Define the ISDN PRI to 1TR6 connectivity.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	CFN	Configuration record.
- ADAN	CHG DCH xx	Change a D-channel on logical port 0-63 (For Large Systems).
- IFC	1TR6	Interface type for 1TR6 connectivity.
- CNEG	(1)2	The required channel negotiation option, either exclusive (the default) or preferred.

Table 129: LD 16 - Define the ISDN PRI to 1TR6 connectivity.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	RDB	Route Data Block.
CUST	0-99	Customer number as defined in LD 15.
ROUT	0-511	Route number for Large System and CS 1000E system.
IFC	1TR6	Interface type for 1TR6 connectivity.

Feature operation

No specific operating procedures are required to use this feature.

EuroISDN

Description

The original offering of the EuroISDN feature provided ISDN PRI connectivity between the system and Public Exchanges that comply to the European Telecom Standards Institute (ETSI) specification ETS 300 102 for the Layer 3. The interfaces provided by this feature also comply with the country-specific Application Documents for Austria, Denmark, Finland, Germany, Holland, Ireland, Italy, Norway, Portugal, Sweden, and Switzerland; other countries must comply with ETS 300 102.

The following capabilities were introduced for the system to EuroISDN connectivity:

- Basic Call Service
- Connected number delivery
- 64 Kbit/s clear-bearer capability
- Calling Line Identification Presentation
- Calling Line Identification Restriction, Connected Line Identification Presentation
- Connected Line Identification Restriction are provided for the above countries where Application Documents are available.
- Channel negotiation
- Advice of Charge (at call set-up, during call, and end of call) is supported in some countries
- interworking with other ISDN or non-ISDN interfaces, including MCDN, QSIG, R2 Multifrequency Compelled Signaling (R2 MFC), 2.0 Mbit Digital Trunk Interface (DTI2), DPNSS1 and DASS2 links, ETSI-NET3 compliant BRI phones, and analog trunk interfaces.

The EuroISDN Continuation feature expanded the scope of the original EuroISDN feature to include application support for Switzerland, Spain, Belgium and the United Kingdom.

In addition, the following functionalities are provided by the EuroISDN Continuation feature:

- Intercept treatment upon reception of an invalid or incomplete called party number
- Interception to an attendant for EuroISDN voice calls terminating on a data device
- The capability to listen to tones and announcements provided by the Central Office on call clearing (T306 supported)
- The capability of transferring outgoing EuroISDN calls after completion of dialing (only for Italy)
- Calling Line Identification and Connected Line Identification transparency to or from EuroISDN to or from a Basic Rate Interface (BRI) phone
- The configuration of Connected Line Presentation (COLP) for each D-channel (remote capability)
- 3.1 KHz audio bearer capability for outgoing fax calls, based on a Class of Service assigned to the analog (500/2500-type) phone
- The capability of defining the bearer as "Voice" or "3.1 KHz" on a system basis
- The addition and display of national or international prefix in front of the received Calling/ Connected Line Identification on incoming/outgoing EuroISDN calls
- Flexible national and local prefixes in addition to the Calling/Connected Line Identification on incoming/outgoing calls based on the route configuration
- User-to-User information transparency in call control messages

- Capability of mapping a PROGRESS message or Progress Indicator in a CALL PROCEEDING message into an ALERT or a CONNECT message for each configuration
- Capability of using different options to build th? Calling Line Identification (CLID)
- Called party number size increase to 31 octets

This development also provides a subset of the ETSI Generic functional protocol for the support of supplementary services. Implementation relies on the Generic Functional (GF) Transport platform introduced in the ISDN QSIG GF Transport feature. The ETSI GF subset provides two types of supplementary service control entities as follows:

- Bearer-related transport with a point-to-point transport mechanism

  This service is used to transport supplementary service protocol information in association with a basic call.

- Bearer-independent transport with a point-to-point (connection-oriented) mechanism

  This service is used to transport supplementary service protocol information, which is entirely independent of any existing basic call.

The ETSI GF provides a generic transport platform that supports ETSI-compliant supplementary services. The ETSI GF protocol is implemented on the ETSI, Swiss, German and Danish EuroISDN interfaces.

EuroISDN Continuation Phase III added France, Russia and the Ukraine to the list of countries having country-specific Application Documents compliance for EuroISDN.

In addition to providing all of the functionalities provided by the original introduction of EuroISDN and the EuroISDN Continuation, EuroISDN Continuation Phase III introduces the following new functionalities for all EuroISDN interfaces:

- optional sending of last forwarding DN as CLID
- Trunk Route Optimization - before Answer applied to incoming EuroISDN trunks
- Numbering Plan Identification (NPI) and Type of Number (TON) included in CDR tickets for EuroISDN calls

The EuroISDN ETS 300 403 Compliance Update feature has been introduced to provide ISDN Primary Rate Interface and Basic Rate Interface to Central Offices that comply with the ETS 300 403-1 European Telecom Standards Institute (ETSI) standard.

Note:

In order to support countries that have not yet upgraded to the ETS 300 403 standard, the system still interworks with Central Offices conforming to the ETS 300 102 standard.

So, when programming the D-Channel for PRI2 trunks (in LD 17) or a PRI2 route for ISDN trunks (in LD 16) for an ETS 300 403 interface, the following applies:

- If IFC = E403 and CNTY = ETSI (ETS 300 403 for the user side) or NET (ETS 300 403 for the network side), then the interface is fully compliant with ETS 300 403.
- If IFC = E403 and CNTY = any of the supported country entries except ETSI and NET, then the interface behaves like an ETS 300 102 extended version, that is, in addition to the existing ETS 300 102 capabilities, the bearer capability and High Layer Compatibility selection procedures (fall-back mechanism), and the basic telecommunication service identification are also implemented in order to take advantage of new teleservices being offered, such as 7kHz telephony and Videotelephony.

  A user can still configure an interface fully compliant with the ETS 300 102 standard if IFC = EURO and CNTY = any of the supported country values.

With the EuroISDN ETS 300 403 Compliance Update feature, all of the functionalities provided by the previous offerings of the EuroISDN features are supported, along with the following enhancements:

- Support for signalling procedures for bearer capability and High Layer Capability selections by, providing repeatable Bearer capability, High layer compatibility Information Elements (IEs) in the SETUP message. ALERT, CALL PROCEEDING, CONNECT, and PROGRESS messages can be used to confirm the selected Bearer Capability/High Layer compatibility for the called user.

  This enhancement provides high-quality bearer services or teleservices, with alternate bearer capability or high layer compatibility in case of fall-back.
- Support for Basic telecommunication service identification. Each basic telecommunication service has the required Bearer capability IE encoding and, if applicable, the required High Layer compatibility IE encoding defined for that service. The requested teleservice is identified by taking the presented Bearer?capability and High layer compatibility information elements in all combinations. If there is no valid combination, the presented Bearer Capability IE is considered in order to identify a bearer service.

Operating parameters

The EuroISDN feature does not support non-Downloadable D-Channel cards (an NT6D80 MSDL or NTBK51AA/NTBK51CA Downloadable D-Channel card is required).

Interworking with KD3 signaling is not supported.

Integrated Service Access (ISA) is not supported.

Advice of Charge for EuroISDN calls is supported with the Advice of Charge for EuroISDN feature.

The EuroISDN to CIS MF Shuttle gateway is supported, as provided by the CIS MF Shuttle feature.

The Auto Terminate Call feature is not supported.

The Call Back Queuing and Off-hook Queuing feature and the Flexible Hotline feature are not supported with Overlap Signaling.

End-to-End Signaling is supported on all outgoing EuroISDN routes as soon as the CALL PROCEEDING message with a Progress Indicator is received.

Special dial tones after dialed numbers are not supported for incoming calls.

For Connected Line Identification Presentation (COLP) supplementary service, sending or restricting connected line identification is not supported for each call on non-ISDN BRI phones.

For the Optional Sending of Last Forwarding DN as CLID functionality, if more than one ACD redirection takes place when an ACD phone is involved, the DN of the called phone is sent as the CLID rather than the DN of the last forwarding phone.

Operating parameters pertaining to Call Control Procedures:

- If more than one Channel Identification IE is received in a SETUP message, only the first one is used by the system. If it is not available, the call is processed according to the channel negotiation configuration. The call is released if no negotiation is allowed. The transfer of a unanswered EuroISDN call to a remote ringing phone requires disconnect supervision on the TIE trunk. If the disconnect supervision is not available, after the external user hangs up, the trunk could be locked out. It is the craftsman's responsibility to ensure that the trunks used for this type of call actually have disconnect supervision.

Operating parameters pertaining to ETS 300 102:

- The user-to-user compatibility checking, by the means of the Low Layer compatibility IE and/or the High Layer compatibility IE, is not supported. The LLC IE and the HLC IE are tandemed by the system, but this information is not used to perform any checking on a system node.
- Transit network selection is not supported. This IE is normally used by the user to identify a selected transit network in the SET-UP message. As no European country specifies the coding to use this IE, this service is not supported by the system. As a result, this IE is never sent by the system.
- Extension for symmetric call operation is not supported. This is normally used to implement a private network application.
- Network specific facility selection procedures are not supported.
- D-Channel backup procedures are not supported by EuroISDN.
- Message segmentation procedures are not supported. These are normally used to split messages that are too long.

- Low Layer Information coding principle is not checked by the system. No LLC is generated by the system, but this information is tandemed if received (from an ISDN BRI phone for example).
- Low layer compatibility negotiation procedures are not supported.
- The USER INFORMATION message is not implemented in the system software.

Operating parameters pertaining to ETSI GF:

- ETSI GF gateways to and from other signalling systems, s?ch as DPNSS1, QSIG, and MCDN, are not supported.
- The following ETSI GF procedures are not implemented:
  - control of supplementary services using the separate message approach (HOLD/ RETRIEVE).
  - bearer-related broadcast transport mechanism (multipoint configuration).
  - bearer-independent point-to-point connectionless transport mechanism.
  - bearer-independent broadcast transport mechanism.
  - generic notification procedures.
  - network-side channel reservation function.
  - generic procedures for supplementary service management.
  - generic status request procedure.
  - support of the Extended facility information element.
- DN address translation requires the association with a customer number. For an ETSI basic call establishment, the customer number association is found through the B-Channel identified in the channel ID IE. For DN address translation that is not associated with a basic call, the customer number association needs to be determined through other methods.

  For a BRI trunk DSL interface, there is a customer number association with the D-Channel. For a PRI interface, a prompt is added for an ETSI D-Channel configuration to create a customer number association with a given D-Channel. This implies that bearer independent messages on a primary rate D-Channel are associated with a single customer as configured in LD 17. For example, in a multi-customer configuration, if every customer on the switch wishes to use the bearer-independent transport service over ETSI PRI interfaces, then each customer requires a separate D-Channel.
- The Facility Information Element (FIE) is a repeatable IE, and its length is application dependent. However, due to system capacity considerations, such as call register usage and real time usage, there are two types of limitations enforced by the ETSI GF transport platform (actually by the ISDN GF transport platform provided by the ISDN QSIG GF development):
  - the system ETSI GF platform supports up to a maximum of eight ROSE components in one message. The eight components can be included in one Facility Information Element (FIE), or multiple FIEs. In addition, the inclusion of components in a

message is also limited by the "available message length". The available message length is the difference between the maximum message length (260 octets), and the maximum message length taken up by other mandatory and optional IEs supported in the given message.

- when a supplementary service requests the ISDN ETSI GF transport to send a component which exceed the available message length or the number of components supported, the supplementary service will be notified.

Operating parameters pertaining to ETS 300 403:

- The operating parameters pertaining to ETS 300 102 are applicable to ETS 300 403, except for the user-to-user compatibility checking, which is partially supported in some cases. For instance, as in the case of the EuroISDN 7 kHz/Videotelephony teleservices.
- Codeset 4, which is reserved for use by the ISO/IEC standards, has been added to the ETS 300 403 standard. This codeset is currently not used by the system. Therefore, it is not supported.
- The fall-back capability for multirate services is not supported.

Feature interactions

Call Completion Supplementary Service

Call Completion interacts with Call Transfer Over EuroISDN SN3. Call Completion Free Notification can only be presented to the Call Completion originating station and cannot be transferred to any other station. However, once the Call Completion call is established, it behaves as a normal call with respect to Call Transfer.

Call Detail Recording

The Optional Sending of Last Forwarding DN as CLID functionality interacts with the CDR feature. The CLID field in the CDR ticket for the redirected outgoing EuroISDN call is modified according to the CLID information in the SETUP message.

CLID Enhancements

The Optional Sending of Last Forwarding DN as CLID functionality interacts with the ISDN CLID Enhancements feature, in that the CLID entry programmed against the prime DN of the forwarding phone at a gateway node ?s used to build the outgoing CLID.

Incoming Digit Conversion Enhancement

The Incoming Digit Conversion Enhancement (IDC) feature converts incoming digits from a DID route. This feature is supported on the incoming EuroISDN DID routes. Digits received as a called party number are converted if the IDC feature is activated on the route. Digit analysis is then performed on the converted digits by the system.

Incoming Trunk Programmable CLID

The Optional Sending of Last Forwarding DN as CLID functionality interacts with the Incoming Trunk Programmable CLID feature. This feature provides the capability of assigning a billing number to incoming trunks.

If the redirection does not occur at the gateway node, the incoming redirecting number information is used as the outgoing CLID on the standard ISDN connectivity DID or CO trunk, even though the Incoming Trunk Programmable CLID feature is configured to send a billing number as CLID information.

Intercept Treatment on Invalid or Partial Dialing

There are three features that are involved in the treatment of calls with dialling irregularities. These are the Partial Dial Timing (PRDL), the Vacant Number Routing (VNR), and the Intercept features.

The Intercept feature allows calls with dialling irregularities to be routed to a Recorded Announcement, an attendant, or to receive overflow or busy tone. Separate treatments can be specified for DID and TIE trunks, for CDP/UDP calls, or for non CDP/ non UDP calls.

The PRDL feature allows to route non-ISDN DID incoming calls to the attendant after a configurable amount of time has expired, if the received digits cannot lead to the completion of the routing.

The VNR feature allows a call to a vacant number to be routed to another node, where the call is either treated as a vacant number, terminated, or given intercept treatment.

In order to provide as much flexibility as possible in the treatment of calls from a EuroISDN interface with dialling irregularities, the PRDL feature has been enhanced. Once the PRDL feature is activated, instead of automatically routing the call to an Attendant, the call will be given the intercept treatment provided by the VNR and Intercept features.

For the PRDL feature, when applied to EuroISDN trunks, the EOD timer used as the PRDL timer must be at least two seconds shorter than the ISDN interdigit timer (T302 = 15 seconds) used for the VNR and Intercept treatments. The call is rejected if T302 times out first. The VNR feature uses another timer which is the NIT timer. Its value is configurable between 3 - 8 seconds for each customer.

Every time a digit is received, if the number received is an invalid number, the VNR, and then potentially the Intercept treatment feature, are activated.

A number is known to be invalid under the following conditions:

- Enbloc receiving is configured and an incomplete or invalid number has been received.
- Overlap receiving is configured, and:
  - an invalid number has been received
  - an incomplete number with a Sending complete IE has been received
- If the number received is not completed and not yet invalid, one of the following timers is started:
  - EOD timer (if PRDL = YES)
  - NIT timer for UDP/CDP calls (If PRDL = NO)
  - T302 timer

Interworking Notification

Direct Interworking

When "direct interworking" occurs, the EuroISDN trunk terminates directly on the analog trunk. As a result, the CALL PROCEEDING message is sent to indicate that no further called party information is required. No further call control message is initiated by the analog trunk until answer supervision is provided. If the trunk is a supervised trunk, the resulting CONNECT message will permit the Public Switched Telephone Network (PSTN) to start charging the caller. However, an unsupervised trunk can also be called. In this case, the CONNECT message is sent upon the expiry of the pseudo answer supervision timer.

The destination non-ISDN trunk can be busy instead of ringing; therefore, it is not effective to provide an ALERTING message, since the PSTN provides the caller with ringback. However, the PSTN will be running a T310 timer. Expiry of this timer causes call clearing, and a message must be provided.

The following three alternatives can be configured in LD 16 by responding to the PROG prompt to stop the network T310 timer:

1. A PROGRESS message is sent to the network after the CALL PROCEEDING message.
2. An ALERT message is sent to the network after the CALL PROCEEDING message.
3. A CONNECT message is sent to the network after the CALL PROCEEDING message.

Tandem Interworking

When tandem interworking occurs, the EuroISDN trunk terminates on an ISDN link terminating on an analog trunk. When the tandem SETUP message is sent, the CALL PROCEEDING message is sent to the PSTN to indicate that no further called party information is required.

The tandem ISDN link returns a PROGRESS message, indicating interworking with a non-ISDN trunk.

Since the PSTN does not implement Annex N, proposed by ETSI, it runs a T310 timer. A message must be provided to stop the T310 timer.

The following three alternatives can be configured in PROG prompt LD 16 to stop the network T310 timer:

1. A PROGRESS message is sent to the network after the CALL PROCEEDING message.
2. An ALERT message is sent to the network after the CALL PROCEEDING message.
3. A CONNECT message is sent to the network after the CALL PROCEEDING message.

When interworking with an analog trunk without answer supervised signaling, the system should be configured to send a CONNECT message rather than an ALERT message. An ALERT or CONNECT message stops the receipt of digits, which implies that all digits have been received.

Networking Features

Some networking features currently exist on more than one ISDN interface implemented on the system. These features are listed in Table 130: Networking features supported by more than one ISDN interface on page 296. Any networking feature that does not appear in the table is only supported on one ISDN interface and is rejected by the gateway when the service is requested. That is the case for all Meridian Customer Defined Network (MCDN) features that are not supported over the EuroISDN or QSIG interface.

Table 130: Networking features supported by more than one ISDN interface

	Euro-ISDN	MCDN	QSIG	ETSI BRI phones	DPNSS 1	MFC	DTI2	Analog
Calling Line ID	Yes	Yes	Yes	Yes	Yes	No	No	No
Connected Number	Yes	Yes	Yes	No	Yes	No	No	No
Transit Counter	No	Yes	Yes	No	Yes	No	No	No

Transfer of Unanswered Call

The EuroISDN Continuation Phase III feature supports the Transfer of an Unanswered Call feature. This allows a station (an attendant or phone) to transfer an unanswered outgoing EuroISDN call, after dialing has been completed and before the transferred-to phone answers

(while the phone is ringing). The transferred-to phone can be local, or can be remote if the link is an MCDN link equipped with Network Attendant Services (NAS), or if the link is a QSIG link equipped with Slow Answer Recall.

Note:

The trunk type for the EuroISDN link must be either CO, DID, TIE, FEX, or WATTS for this functionality to work. Otherwise, the transferred-to phone must answer before the transfer can be done.

The transfer of an unanswered EuroISDN call interacts with the CDR feature. When the originating party completes the call transfer after the transferred-to phone answers, a CDR Start (S) is generated when the transferred-to phone answers. When one side releases the call, a CDR End (E) ticket is generated. If the originating party completes the transfer while the transferred-to phone is ringing, an S ticket will be generated only when the transferred-to phone answers.

Virtual Network Services

A EuroISDN link can be used as a B-Channel for the Virtual Network Services feature.

Feature packaging

The EuroISDN interface is packaged under the EuroISDN (EURO) package 261.

EuroISDN Continuation is included in EuroISDN (EURO) package 261. The following packages are also required:

- Digit Display (DDSP) package 19
- Integrated Services Digital Network (ISDN) package 145
- 2.0 Mbit Primary Rate Interface (PRI2) package 154
- Integrated Services Digital Network Supplementary Features (ISDN INTL SUP) package 161
- Overlap Signaling (OVLP) package 184
- International ISDN PRI (IPRA) package 202
- Multi-purpose Serial Data Link (MSDL) package 222

For Basic Rate Interface (BRI), the following software packages are required:

- Digit Display (DDSP) package 19
- Integrated Services Digital Network (ISDN) package 145
- Integrated Services Digital Network Supplementary Features (ISDN INTL SUP) package 161

- International ISDN PRI (IPRA) package 202
- Meridian 1 Extended Peripheral Equipment (XPE) package 203
- Basic Rate Interface (BRI) package 216
- Integrated Services Digital Network Basic Rate Interface Trunk Access (BRIT) package 233

For the Uniform Dialing Plan (UDP), the following packages are also required:

- Basic Routing (BRTE) package 14
- Digit Display (DDSP) package 19
- Network Call of Service (NCOS) package 32
- Network Alternate Route Selection (NARS) package 58

For the Coordinated Dialing Plan (CDP), package 59 is required. Flexible Numbering Plan (FNP) package 160 is optional.

The following packages are required for EuroISDN to interact with Multifrequency Compelled Signaling (MFC), Multifrequency Compelled Signaling for Socotel (MFE), Direct Inward Dialing (DID), Digital Access Signaling System #2 (DASS2), and Digital Private Network Signaling System #1 (DPNSS1):

- Universal ISDN Gateway (UIGW) package 283
- Enhanced DPNSS1 Gateway (DPNSS189I) package 284

The following packages are required for the Numbering Plan Identification (NPI) and Type of Number (TON) fields in CDR:

- Call Detail Recording (CDR) package 4
- Call Detail Recording on Teletype Terminal (CDR_TTY) package 5
- Calling Line Identification in Call Detail Recording (CLID_CDR_PKG) package 118
- New Format for Call Detail Recording (CDR_NEW) package 234

Feature implementation

Task summary list

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

1. Table 131: LD 15 - Configure the Optional Sending of Forwarding CLID functionality (in response to the OCLI prompt). on page 299
2. Table 132: LD 17 - Create a new D-channel for the EuroISDN interface. on page 299

3. Table 133: LD 16 - Configure new DID routes using the same responses to the IFC and CNTY prompts as in LD 17. on page 301
4. Table 134: LD 17 - Configure the NPI and TON fields in the CDR tickets. on page 304 These fields, which are additional to the CLID, apply only for EuroISDN calls. Entering YES in response to the CLID prompt configures these fields automatically.

Table 131: LD 15 - Configure the Optional Sending of Forwarding CLID functionality (in response to the OCLI prompt).

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	NET	Networking data.
CUST	0-99	Customer number as defined in LD 15.
ISDN	YES	Integrated Services Digital Network.
PNI	(0)-32700	Private Network Identifier.
SATD	0-(1)-5	Satellite Delays.
OCLI	(NO)	NO = No manipulation is done on outgoing CLID for calls forwarded over EuroISDN links.
	EXT	EXT = The last forwarding DN is sent as CLID information for incoming calls over a standard ISDN (EuroISDN, APAC, NI-2) DID or CO link to a gateway system node, and redirected back over the standard ISDN (EuroISDN, APAC, NI-2) DID or CO link.
	ALL	ALL = The same as for EXT. Moreover, the last forwarding DN is sent as CLID information for internal calls from a local phone or over ISDN TIE trunks, except for DPNSS1 calls which are redirected from a gateway node over a standard ISDN (EuroISDN, APAC, NI-2) DID or CO link. If the redirection does not occur at the gateway node, the redirecting information number, if present in the incoming SETUP message received at the gateway node from the private network, is sent as CLID.

Table 132: LD 17 - Create a new D-channel for the EuroISDN interface.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	ADAN	Action Device and Number.
- ADAN	NEW DCH x	New D-channel at port number x.
- CTYP	MSDL	Card type, where MSDL = Multipurpose Serial Data Link
- GRP	0-4	Network Group Number.

Prompt	Response	Description
- DNUM	0-15	Device number for I/O ports.
- PORT	0-3	Port number on MSDL card.
- USR	PRI	D-channel for ISDN PRI only. Precede with X to remove.
- IFC		DCH interface type.
	EURO E403	EURO = EuroISDN E403 = EuroISDN Interface conforming to the ETS 300 403, or the extended ETS 300 102 version for country-specific interfaces. Refer to Description on page 287.
- - CNTY		Enter country pertaining to EuroISDN interface.
	AUS DEN (ETSI) FIN GER ITA NOR POR SWE EIR DUT SWI BEL ESP UK FRA CIS EAUS	Austria Denmark ETS 300-102 basic protocol Finland Germany Italy Norway Portugal Sweden Ireland Holland Switzerland Belgium Spain United Kingdom France Commonwealth of Independent States (Russia and the Ukraine). Australia ETSI. Enter country pertaining to the E403 interface. If either ETSI or NET are entered, the interface is fully compliant with ETS 300 403. See the Description on page 287.
	ETSI NET	ETS 300 403 for the user side. ETS 300 403 for the network side. If any of the countries listed for the EURO IFC are entered, the interface functions with the extended ETS 300 102 capabilities. See Description on page 287.
PINX_CUST	xx	The customer number used for the DN address.
...		
DCHL	0-159	The PRI loop number for the D-Channel.
- - CNEG	(1) 2	Channel Negotiation option.

Prompt	Response	Description
		(1) = Channel is indicated and no alternative is acceptable. This is the default value for all EuroISDN Interfaces except FRA. 2 = Channel is indicated and any alternative is acceptable. This is the default value for the FRA interface.
- RCAP	COLP XCOL	Enter COLP to support Connected Line Identification Presentation as a remote capability. This is the default value for the EAUS, ESIG, ISIG, NI2, and EURO interfaces. Enter XCOL to remove Connect Line Identification Presentation as a remote capability. This is the default value for the AUS, EIR, DEUT, ESP, BEL, and FRA interfaces.

Table 133: LD 16 - Configure new DID routes using the same responses to the IFC and CNTY prompts as in LD 17.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	RDB	Route Data Block.
CUST	0-99	Customer number as defined in LD 15.
ROUT	0-511	Route number for Large System and CS 1000E system.
TKTP	DID COT	Trunk type. DID = Direct Inward Dialing COT = Central Office.
...		
DTRK	YES	Digital trunk route.
- DGTP	PRI2 BRI	Digital trunk type for route.
...		
- IFC		DCH interface type.
	EURO E403	EURO = EuroISDN E403 = EuroISDN Interface conforming to the ETS 300 403, or the extended ETS 300 102 version for country-specific interfaces. Refer to Description on page 287.
- - CNTY		Enter country pertaining to EuroISDN interface.
	AUS DEN (ETSI)	Austria Denmark ETS 300-102 basic protocol

Prompt	Response	Description
	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).
	EAUS	Australia ETSI. Enter country pertaining to the E403 interface. If either ETSI or NET are entered, the interface is fully compliant with ETS 300 403. See Description on page 287.
	ETSI	ETS 300 403 for the user side.
	NET	ETS 300 403 for the network side. If any of the countries listed for the EURO IFC are entered, the interface functions with the extended ETS 300 102 capabilities. See Description on page 287.
ICOG	IAO	Incoming and Outgoing trunk.
	OGT	Outgoing trunk only.
	ICT	Incoming trunk only.
ACOD	x..x	Access code for this trunk route.
- CLID	OPT0	Prefix = 0, for North American dialing plan. This is the default value for ESIG and ISIG interfaces.
	OPT1	Prefix = 1, for international PFXs in CLID, any numbering type supported. This is the default value for all EuroISDN interfaces.
	OPT2	Prefix = 2, for international PFXs in CLID, CCITT numbering type supported: UKWN, INTL, NPA, and NXX. Default value for CO/DID routes for the Telecom New Zealand interface.
	OPT3	Prefix = 3, for international PFXs in CLID, only NXX number type supported. Default value for TIE routes for the Telecom New Zealand interface.
	OPT4	For international COs, if the call originates from a CO trunk type, add nothing. Otherwise, add PFX1 and PFX2. This is the default value for the Hong Kong, Singapore, and Thailand interfaces.

Prompt	Response	Description
	OPT5	This is the same as OPT4, except it supports a maximum of 10 digits in the CLID. This is the default value for the Austrian interface.
PROG		Progress Signal.
	NCHG	Send PROGRESS (default value for all interfaces, except Austria).
	MALE	Send ALERT after CALL PROCEEDING.
	MCON	Send CONNECT after CALL PROCEEDING (this is the default value for the Austrian interface).
...		
- RCAP	COLP XCOL	Enter COLP to support Connected Line Identification Presentation as a remote capability. This is the default value for the ESIG, ISIG, NI2, and EURO interfaces. Enter XCOL to remove Connect Line Identification Presentation as a remote capability. This is the default value for the AUS, EIR, DEUT, ESP, BEL, and FRA interfaces.
...		
- CPFXS	(YES) NO	Customer-defined Prefixes option. This prompt is added as one of the sub-prompts of ISDN = YES. It is added at the end of the ISDN sub-prompts. If CPFXS = YES, when constructing the Calling or Connected Line Identification, the prefixes are retrieved from the Customer Data Block via the PFX1 and PFX2 prompts in LD 15, as is currently done. This is the default response. If CPFXS = NO, when constructing the Calling or Connected Line Identification, the prefixes are retrieved from the Route Data Block via the HNTN and HLCL prompts in LD 16.
- - HNTN	0-9999	Home National Number. This number is similar to the PFX1 number prompted in LD 15. It is added to this overlay so that this prefix can be configured on a route basis as required in some countries (e.g., Italy). As is the case with PFX1, the HNTN prefix can be from one-to-four digits long. This prompt is displayed only if CPFXS = NO. If only a <CR> is entered, this prompt keeps its previous configuration. If no value was configured previously, no value will be configured. Enter X to delete the digits.
HLCL	0-9999	Home Location Number. This number is similar to PFX2 number prompted in LD 15. It is added to this overlay so that this prefix can be configured on a route basis as

Prompt	Response	Description
		required in some countries (e.g., Italy). As is the case with PFX2, the HLCL prefix can be from one-to-four digits long. This prompt is displayed only if CPFXS = NO. If only a <CR> is entered, this prompt keeps its previous configuration. If no value was configured previously, no value will be configured. Enter X to delete the digits.
ADDP	(NO)	If ADDP = NO, the Calling or Connected Party Number displayed is not modified.
	YES	If ADDP = YES, the prefixes 0 (national) or 00 (international) are added to the Calling Party Number if the Type of Number (TON) is public.

Table 134: LD 17 - Configure the NPI and TON fields in the CDR tickets.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	PARM	Change system parameters.
...		
- FCDR	NEW	Format for Call Detail Recording. Enter NEW for new format.
...		
- CLID	(NO) YES	Enter YES to include the TON and NPI fields in the CDR ticket, in addition to the CLID.

Feature operation

No specific operating procedures are required to use this feature.

Japan D70, non-Asia Pacific connectivity

Description

This feature provides ISDN PRI connectivity between the system and the INS1500 D70 for Japan. The design in based on the 23 B+D PRI configuration.

The system to Japan D70 ISDN PRI connectivity supports the following:

- Basic Call Service
- Calling Line Identification (public and private)
- Overlap Sending
- COT, DID, DOD, and TIE trunk call types
- 64 Kbit/s clear bearer capability
- Channel Negotiation
- nB+D, up to 120 B-channels/four interfaces
- Public Switched Telephone Network (PSTN) three-party conferencing
- Flexible Numbering Plan
- Malicious Call Trace

Operating parameters

For the ISDN layer 1 interface, the circuit pack QPC720 (Digital Trunk Interface) is used for the system to D70 connectivity.

The QPC757E version of the D-channel Handler (DCH) circuit pack provides the Layer 2 functions and incoming Layer 3 preprocessing for the system to D70 connectivity.

The NTAK09AA (DTI/PRI) circuit pack is used for Option 11 systems, along with the NTAK93 D-channel Interface (DCHI) and NTAK20 Clock Controller.

A Multi-purpose Serial Data Link (MSDL) can be used in place of the DCH circuit pack (NT6D80AA).

Feature interactions

PRI Channel Negotiation

When a D-channel has one or more secondary PRIs associated with it while using the D70 interface, channel negotiation requires that the PRI interface be explicitly defined for all B-channels not on the primary interface.

Feature packaging

This feature requires the following packages:

- Digit Display (DDSP) package 19 for CLID
- 1.5 Mbit Digital Trunk Interface (PBXI) package 75
- Integrated Services Digital Network (ISDN) package 145
- 1.5 Mbit Primary Rate Access (PRA) package 146
- International Primary Rate Access (IPRA) package 202
- Multi-purpose Serial Data Link (MSDL) package 222 for use of the MSDL card in place of the DCH card

Feature implementation

Task summary list

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

1. Table 135: LD 17 - Define the ISDN PRI to D70 connectivity. on page 306
2. Table 136: LD 16 - Define the ISDN PRI to D70 connectivity. on page 307

Table 135: LD 17 - Define the ISDN PRI to D70 connectivity.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	CFN	Configuration record.
- ADAN	CHG DCH x	Change a D-channel on logical port 0-63 (For Large Systems).

Prompt	Response	Description
- IFC	D70	Interface type for Japan D70 connectivity.
- CNEG	(1) 2	The required channel negotiation option, either exclusive (the default) or preferred.

Table 136: LD 16 - Define the ISDN PRI to D70 connectivity.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	RDB	Route Data Block.
CUST	0-99	Customer number as defined in LD 15.
ROUT	0-511	Route number for Large System and CS 1000E system.
IFC	D70	Interface type for Japan D70 connectivity.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 23: ISDN QSIG

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

The European Computer Manufacturer's Association (ECMA) has defined an ISDN protocol that specifies the Layer 3 signaling requirement for support of circuit switched call control at the "Q" reference point between Private Telecommunication Network Exchanges (PTNXs) connected within a Private Telecommunication Network (PTN). This protocol has been adopted by the European Telecommunication Standards Institute (ETSI) and the International Standards Organization (ISO). Most of the major European PTNX manufacturers will be supporting ISDN connectivity based on this standard.

QSIG is oriented towards signaling and services that occur between two switches. For example, two PBXs, or a PBX and a Centrex switch could exchange signaling for services across a "Q" reference point.

The QSIG interface supports the following services:

- Call Establishment and Tear Down
- ETSI or ISO version of basic call
- 64 Kbit/s clear data
- Overlap Sending/Receiving
- Channel Negotiation
- Calling and Connected Parties Information (CLIP/COLP)
- Calling and Connected Parties Restriction (CLIR/COLR)
- Flexible Numbering Plan
- TIE call types
- Transit Count information transmitted when ISDN Call Connection Limitation (ICCL) is present

The QSIG interface is supported on PRI, BRI, and ISDN Signaling Links (ISLs). The QSIG interface does not have any transit capability for supplementary services.

QSIG development underscores Avaya's commitment to global standards. QSIG will provide a greater interworking of voice, image, video, and data services in multi-vendor environments. Users will also be able to combine ISDN PRI with ISDN BRI to deliver enhanced services through end-to-end ISDN networks.

Operating parameters

There are no operating parameters specified for this feature.

Feature interactions

Networking Features

Some networking features currently exist on more than one ISDN interface implemented on the system. These features are listed in Table 137: Networking features that exist on more than one ISDN interface implemented on the system on page 311. Any networking feature that

does not appear in the table is only supported on one ISDN interface and is rejected by the gateway when the service is requested. That is the case for all Meridian Customer Defined Network (MCDN) features that are not supported over the QSIG interface.

Table 137: Networking features that exist on more than one ISDN interface implemented on the system

	Euro - ISDN	MCDN	QSIG	BRI phones	DPNSS	MFC	MFE	DTI2	Analog	1TR6	NUMERIS	AXE10
Calling Line ID	Y	Y	Y	Y	Y	Y	N	N	N	Y	Y	Y
Connected Number	Y	Y	Y	N	Y	N	N	N	N	Y	N	N
Transit Counter	N	Y	Y	N	Y	N	N	N	N	N	N	N
Call Charge	Y	N	N	N	N	N	N	Y	Y	Y	Y	N

Trunk Route Optimization

Trunk Route Optimization (TRO) is supported within the system network only. When a redirecting node sends a message to the originating node and the TRO request is accepted, the new call can go through the QSIG interface as a normal basic call. However, TRO signaling does not operate on the QSIG interface.

Network Call Redirection

The existing Network Call Redirection limitation on unsupported interfaces applies to the QSIG interface. When a call is terminated on the system and network call redirection is active, the feature can still be operated but the Original Called number and the Redirection number Information Elements (IEs) which are used by the Network Call Redirection feature will not be sent on by the QSIG interface.

ISDN Signaling Link (ISL)

The existing ISL operation is supported on the QSIG interface on the PRI/PRI2 interface only.

Network Attendant Service (NAS)

The interaction with NAS is as though the call is going to a route without NAS equipped. The only information exchanged between NAS and the QSIG interface is the transit count. Other information contained in the Progress IE will be handled by a future development.

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

Call Forward, Break-In and Hunt Internal or External Network Wide uses the Network Attendant Service equivalent information that is transported on protocols such as Party Category and Progress Indicator for QSIG.

Virtual Network Services (VNS)

VNS provides ISDN features when no MCDN link is available between two switches. This can be done when analog lines are available or when ISDN links that provide no supplementary services are used. The existing link is used as a B-channel, and a separate D-channel handles the MCDN signaling between the two end switches.

A QSIG link can be used as a B-channel for the VNS feature over a private network. All VNS services are then supported as normal; the QSIG link is only used as a speech bearer.

Calling Party Privacy Enhancement

The ISDN QSIG interface supports the Calling Party Privacy Enhancement (CPPE) feature in CS 1000 Release 6.0 and later.

The Calling Party Privacy Enhancement (CPPE) feature provides a route option to ignore the Calling Party Privacy Indicator for incoming calls received from all public Integrated Services Digital Network (ISDN) interfaces. When the Privacy Indicator Ignore (PII) prompt is YES in LD 16, the Calling Line Identification (CLID) Presentation Indicator and the Calling Party Name Display (CPND) Indicator override the blocking of the number and name details of the calling party. The name and number of the calling party appear on the receiving telephone.

The CPPE feature introduces a new route option AUXP for Auxiliary processor applications in LD 16 (RDB – Route Data Block). AUXP enhances the ability of the system to honor or ignore the Privacy Indicator for a Calling Party Privacy call, for each incoming route. If AUXP is 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, there will be no change to the CLID Presentation Indicator.

The AUXP route option applies to the CLID Presentation Indicator in the Calling Number IE and the CPND Indicator in the Display IE in the incoming SETUP messages only.

Feature packaging

A new QSIG Interface (QSIG) package 263 must be provisioned to activate this feature.

The following software packages are also required for QSIG:

- Digit Display (DDSP) package 19
- Integrated Services Digital Network (ISDN) package 145
- Primary Rate Access (PRA) package 146
- 2.0 Mbit Primary Rate Access (PRI2) package 154
- Overlap Signaling (OVLP) package 184
- International Primary Rate Access (IPRA) package 202
- Calling Party Privacy (CPP) package 301

Feature implementation

Task summary list

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

- Table 138: LD 17 - Assign the configuration record. on page 313
- Table 139: LD 16 - Define a Route Data Block. on page 314
- Table 140: LD 97 - Configure the Extended Peripheral Equipment (XPE). on page 314
  This overlay defines the Extended Peripheral Equipment (XPE) configuration. Data relating to the software downloading for the MSDL card remains in the overlay. Hence, this overlay must be modified to accept the parameters required for downloading the PRI application data files. The parameters specify conditional or forced downloading.

Table 138: LD 17 - Assign the configuration record.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	CFN	Configuration Record.

Prompt	Response	Description
...		
IFC	ISIG ESIG	Interface ID for ISO QSIG. Interface ID for ETSI QSIG.
TIMR	(NO) YES	NO = skip timer prompt. YES = change timer value.
T310	10-(30)-60	10–60 seconds (1-second increments). 30 seconds is the default value.

Table 139: LD 16 - Define a Route Data Block.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	RDB	Route Data Block.
...		
DGTP	PRI BRI	Digital route type.
IFC	ISIG ESIG	New DCH interface ID.
PII	YES	Enable the Privacy Indicator Ignore (PII) feature
AUXP	YES	Enable the Auxiliary processor application (AUXP) feature **Note:** The AUXP prompt is automatically set to YES when the PII prompt is YES, and can be modified only when PII is NO.

Table 140: LD 97 - Configure the Extended Peripheral Equipment (XPE).

Prompt	Response	Description
REQ	CHG	Change.
TYPE	SYSM	System parameters for MSDL/MISP cards.
...		
FDLC		Peripheral Software Download Option.
	P1 P2	This will be set to the application and its data files, such as BRIE, and PRIE. Specifies conditional (C) or forced downloading (F).

Feature operation

There are no operating procedures specified for this feature.

Chapter 24: ISDN QSIG Alternate Routing

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

The ISDN QSIG Alternate Routing feature provides a solution to calls encountering congestion due to high traffic situations within a QSIG network. The QSIG interface protocols, European Telecommunication Standard Institute (ETSI) and International Standards Organization (ISO) are supported by this feature.

This feature uses the routing capability of Network Alternate Route Selection (NARS) to re-route a congested call. For each QSIG call translated at a system node, NARS selects one route from up to 512 routes to complete the call. These routes are programmed in a route list.

Each route in the list is called an entry. There can be up to 64 entries in each route list. Any combination of trunks (such as public exchange, TIE) can be specified in a route list.

QSIG Alternate Routing can be configured for each of the 512 different routes.

Congestion occurs when all trunks of a route are busy. With the introduction of the QSIG Alternate Routing feature, each entry of a route list on one node can be configured to take an alternate entry (route) from the route list of that node (Private or Public Exchange), if congestion is encountered.

Using LD 86, the Electronic Switched Network (ESN) administration overlay, the option defined for the Step Back On Congestion (SBOC) prompt determines the type of alternate routing available to calls over a particular route. These options are as follows:

- NRR (no alternate routing is performed. The call receives congestion treatment).
- RRO (reroute if a call encounters congestion at the originating node. If congestion is encountered at a transit node, the call drops back to the originating node, so that the originating node decides if re-routing is required. The drop-back functionality, which is part of the ISDN Drop Back Busy feature, is also configured in LD 86, using the IDBB prompt).
- RRA (reroute the call at any node, whether congestion is encountered at the originating or transit node).

QSIG Alternate Routing is triggered at the controlling node when a Call Clearing message (DISCONNECT or a REJECT) is received, and the cause value is to activate QSIG Alternate Routing. The cause values are defined using LD 86 (see Table 141: LD 86 - Configure the QSIG Alternate Routing options. on page 323).

Transit node operation

Consider the following calling scenario for a transit node operation (refer to Figure 14: Transit node operation on page 319). An attempt is being made to establish a call over a QSIG link, from originating node A to terminating node C, through transit node B. All the trunks pertaining to the call attempt at node C are busy. Node C sends congestion message information, along with a supported QSIG Alternate Routing cause value, back to node B.

At node B, the SBOC option, as defined in LD 86, is checked to determine the routing treatment. If SBOC = RRA, the next free alternate route is tried. In our example, the first free alternate route that is found is between node B and node D. Node D can either be a Private or Public Exchange. If the Network Class of Service access checks are passed, the direct leg between the congested node (node C) is released, and an indirect new leg is created to node C, through node D. The call would then be established from node A, to node B, to node D, to node C.

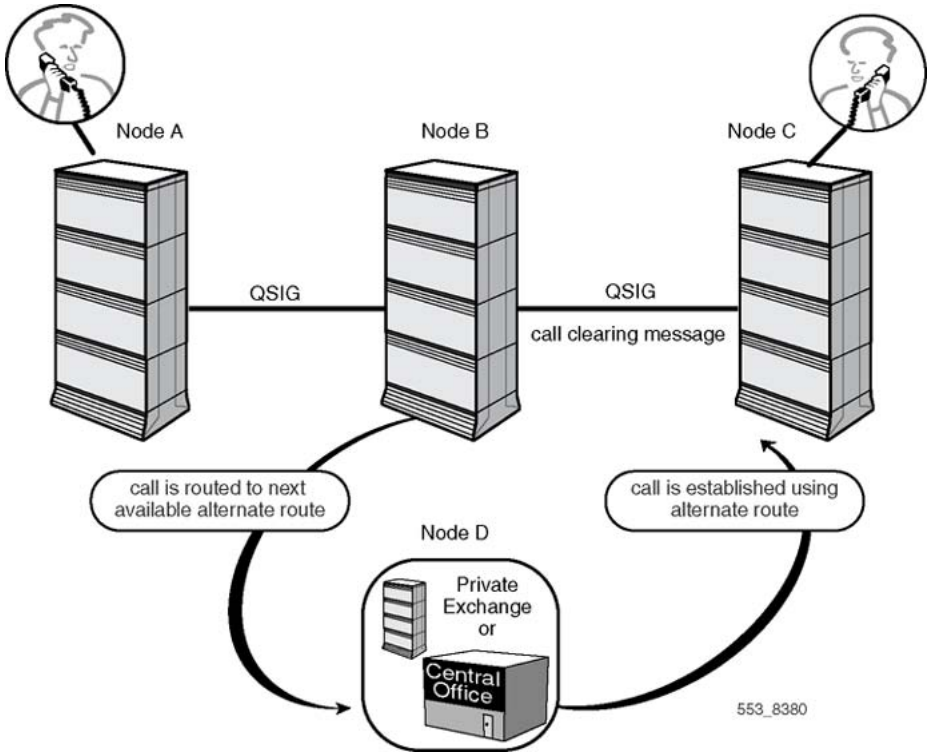


Figure 14: Transit node operation

An attempt is made to find a free alternate route until all of the alternate routes, as defined in LD 86 (refer to the ROUT prompt in Table 141: LD 86 - Configure the QSIG Alternate Routing options. on page 323), are tried. If no free alternate route is available, the congestion message information is sent from node B to the originating node A, where the QSIG Alternate Routing functionality is activated. If no alternate routes are found, the call receives network blocking treatment, as defined by prompt NBLK in LD 15, as part of the initial customer configuration.

If SBOC = RRO, the congestion information is passed back from transit node B to the originating node A. At node A, QSIG Alternate Routing is activated in an attempt to find an alternate route to set up the call from node A, to node B, to node D, to node C.

If SBOC = NRR, no alternate routing is performed. The call receives network blocking treatment, depending on the congestion cause value and the type of treatment defined by prompt NBLK in LD 15.

Originating node operation

Consider the following calling scenario for an originating node operation (refer to). An attempt is being made to establish a call over a QSIG link, from originating node A to terminating node B. All the trunks All the trunks pertaining to the call attempt at node B are busy, so node B sends congestion message information, along with a supported QSIG Alternate Routing cause value, back to node A. Figure 15: Originating node operation on page 320

At node A, the SBOC option, as defined in LD 86, is checked to determine the routing treatment. If SBOC = RRA or RRO, the next free alternate route is tried. In our example, the first free alternate route that is found is between node A and node C. Node C can either be a Private or Public Exchange. If the Network Class of Service access checks are passed, the direct leg between the congested node (node B) is released, and a new indirect leg is created to node B, through node C. The call would then be established between node A, to node B, to node C.

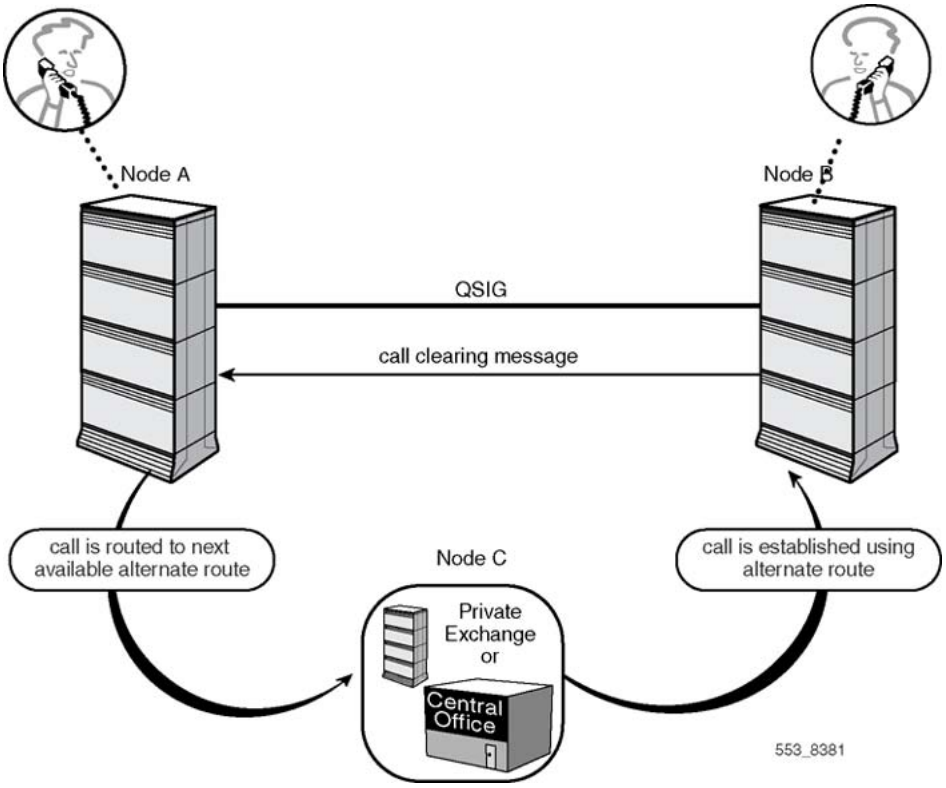


Figure 15: Originating node operation

An attempt to find a free alternate is made until all of the alternate routes, as defined in LD 86 (refer to the ROUT prompt), are tried. If no alternate routes are found, the call receives network blocking treatment, as defined by prompt NBLK in LD 15, as part of the initial customer configuration.

If SBOC = NRR, no alternate routing is performed. The call receives network blocking treatment, depending on the congestion cause value and the type of treatment defined by prompt NBLK in LD 15.

Operating parameters

There are two forms of QSIG Call Transfer - QSIG Call Transfer by the "rerouting" method, and QSIG Call Transfer by the "join" method. QSIG Alternate Routing and the QSIG Call Transfer by the "re-routing" method are mutually exclusive.

Optional Information Elements (IEs) can be lost at a rerouting node. Only the following optional Information Element (IEs) are tandemmed in the SETUP of the alternate call, if they are present in the congested call:

- Calling number
- Called and calling subaddress
- High layer compatibility
- Low layer compatibility

On a QSIG network, optional IEs (like progress IEs) carried in the SETUP messages are lost when the alternate routing occurs at a transit node. Mandatory IE are always tandemmed transparently.

GF facility information is lost at a transit node. If QSIG Alternate Routing is performed, Facility IEs contained in the SETUP message are lost, so services like QSIG Name Display, QSIG Call Diversion (rerouting and by join method) are not delivered to the end-user. QSIG Alternate Routing is exclusive with QSIG Call Transfer, if the rerouting method is used. This does not have a major impact, since QSIG Call Transfer by rerouting is optional, whereas QSIG Call Transfer by join is mandatory. It is therefore advised that if a third-party PBX is part of a QSIG network, QSIG Call Transfer by join method is used.

Missing progress IEs due to QSIG Alternate Routing activation can create a situation where a call fails involving an ISDN BRI phone. This can occur when a Meridian proprietary phone calls an ISDN BRI phone over a QSIG link, and QSIG Alternate Routing is triggered. In this case the Progress Indicator Number 3 "Originator is not ISDN" is lost.

Route Access codes are not supported. A Coordinated Dialing Plan (CDP) or a Uniform Dialing Plan (UDP) is required for the QSIG Alternate Routing feature.

Feature interactions

Drop Back Busy and Off-Hook Queuing

QSIG Alternate Routing takes precedence over ISDN Drop Back Busy, if both are configured on an MCDN-to-QSIG gateway node.

Intercept treatment

If QSIG Alternate Routing fails to find an alternate route for a call encountering congestion at a transit node, intercept treatment is not applied at the transit node. The call is dropped back to the originating node, with the appropriate congestion IE information.

Overlap Sending

QSIG Alternate Routing is supported over both the enbloc and overlap signaling methods.

QSIG Name Display

When QSIG Alternate Routing is triggered at a tandem node, no calling name display information is sent in the call SETUP message for the new call over the alternate route. However, the calling name display information is sent from an originating node or at an MCDN-to-QSIG gateway node, if it is available.

QSIG Path Replacement

QSIG Alternate Routing cannot be applied to a QSIG Path Replacement call set-up. Therefore, QSIG Path Replacement is rejected if congestion is encountered.

Virtual Network Services

QSIG Alternate Routing can function over all QSIG-based bearer interfaces supporting Virtual Network Services.

MCDN End to End Transparency

When QSIG Alternate Routing is triggered at a tandem node, the Facility IE messages containing the Network Attendant Services (NAS) and Network Attendant ACD (NACD) information are not included in the SETUP message for the new call over the alternate route.

Feature packaging

This feature requires the following packages:

- Basic Routing (BRTE) package 14
- Network Class of Service (NCOS) package 32

One of the following:

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

Feature implementation

Note:

QSIG Alternate Routing can be configured for each of the 512 different routes.

Table 141: LD 86 - Configure the QSIG Alternate Routing options.

Prompt	Response	Description
REQ	CHG	Change existing data.
CUST	0-99	Customer number as defined in LD 15.
FEAT	RLB	Configure the Route List as a feature.
MXLC	0-999 0-16000	Maximum number of NARS Location Codes. Maximum number of Location Codes (with the ESN Location Code Expansion feature enabled and with the FNP package 160)
RLI	xxx	Route List Index to be accessed. xxx = 0-127 if a Coordinated Dialing Plan is used. xxx = 0-255 if NARS is configured. xxx = 0-1999 if the Flexible Numbering Plan is configured.

Prompt	Response	Description
ENTR	0-63	Entry number for the NARS route list.
ROUT	0-511	Route number for Large System and CS 1000E system.
SBOC		Step Back On Congestion option.
	(NRR)	No re-routing.
	RRO	Re-route if congestion is encountered at the originating node; if congestion is encountered at a transit node, drop-back to the originating node, so that the originating node decides if re-routing is needed (the IDBB prompt in this LD 86 must be set to DBA or DBI).
	RRA	Re-route whether congestion is encountered at the originating or tandem node.
- COPT		Conditions that can cause QSIG Alternate Routing. These values are contained in signaling messages that trigger alternate routing. The COPT prompt is generated only if SBOC = RRO or RRA.
	(1)	QSIG Alternate Routing is supported due to the following causes: — Cause 34, "No Channel or Circuit Available" — Cause 38, "Network Out of Order" — Cause 42, "Congestion"
	2	QSIG Alternate Routing is supported due to the following causes: — Cause 27, "Destination is Out of Service" — Cause 34, "No Channel/Circuit Available" — Cause 38, "Network Out of Order" — Cause 42, "Congestion"
IDBB		ISDN Drop Back Busy options.
	(DBD) DBA DBI	Deny Drop Back Busy. Allow Drop Back Busy on, if all route phones are busy. Allow Drop Back Busy, if all Initial route phones are busy.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 25: ISDN QSIG Basic Call

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

Integrated Services Digital Network (ISDN) Q Reference Signaling Point (QSIG) Basic Call complies with the ISDN protocol that specifies the Layer 3 signaling requirement for support of circuit switched call control at the "Q" reference point between Private Telecommunication Network Exchanges (PTNXs) connected within a Private Telecommunication Network (PTN).

This protocol has been adopted by the European Telecommunication Standards Institute (ETSI) and the International Standards Institute (ISO). Most of the major European PTNX

manufacturers will be supporting ISDN connectivity based on this standard. This applies to both ISDN PRI and ISDN BRIT (trunk application.)

QSIG is oriented toward signaling and services that occur between two "peer-to-peer" switches across a "Q" reference point. For example, two PBXs, a PBX and a Centrex switch or two Centrex switches could exchange signaling and services across a "Q" reference point.

The ISDN QSIG Basic Call feature introduces the following basic call services on ISDN 1.5 Mbit PRI on QSIG connectivities (these services were previously available only on ISDN 2.0 Mbit PRI connectivities):

- Basic Call Service
- 64 K clear bearer capability
- TIE call types
- Connected number delivery (COLP/COLR)
- Channel Negotiation
- Flexible Numbering Plan
- Enbloc dialing
- Overlap sending
- Overlap receiving
- Calling Line Identification Presentation and Restriction (CLIP/CLIR)
- Loop avoidance (transit count) on ETSI QISG, not on ISO QSIG
- Party Category on ETSI QISG, not on ISO QSIG.

Operating parameters

Meridian Customer Defined Network (MCDN) features cannot be interworked between an MCDN and QSIG interface, with the exception of basic call, Calling Line Identification Presentation/Restriction and Calling and Connected Line Presentation/Restriction.

The system accepts up to three Progress Information Elements (IE) in one message on the QSIG interface.

The maximum length of an ISDN message is limited to 260 bytes.

Only circuit mode connection is supported.

nB+D is not supported for QSIG.

Feature interactions

Backup D-Channel

Backup D-Channel is not supported on the QSIG interface.

ISDN Signaling Link

ISL on QSIG is not supported on ISDN BRI trunking.

Network Attendant Services

ISDN QSIG Basic Call interacts with Network Attendant Services (NAS) as if the call is going to a route without NAS being equipped.

Network Automatic Call Distribution

Network Automatic Call Distribution signaling is not supported on the QSIG interface. Network ACD operation on QSIG is the same as the existing Network ACD treatment for unsupported interfaces.

Network Call Redirection

When a call is terminated on the system and Network Call Redirection (NCR) is active, the QSIG Basic Call can still operate; however, the original called number and redirection number IE that are used by NCR will not be sent on the QSIG interface.

Network Call Trace

Network Call Trace is not supported on the QSIG interface.

Network Calling Party Name Display

Network Calling Party Name Display (NCPND) is supported within the Meridian Customer Defined Network only. When QSIG is involved in the call setup, the existing NCPND operation on unsupported interfaces applies on QSIG.

Network Message Service

Network Message Service (NMS) is only supported on the Meridian Customer Defined Network. NMS operation on QSIG is the same as the existing treatment for unsupported interfaces.

Network Ring Again

Network Ring Again signaling is supported within the Meridian Customer Defined Network only. Network Ring Again requests which go through the QSIG interface will not be supported.

Remote Virtual Queuing

Remote Virtual Queuing (RVQ) does not operate on the QSIG interface. The existing RVQ operation on unsupported interfaces applies on the QSIG interface.

Trunk Route Optimization

Trunk Route Optimization (TRO) is supported within an MCDN network only. When a redirecting node sends a message to the originating node and the TRO request is accepted, the new call will go through the QSIG interface just as a normal basic call. However, TRO signaling will not be able to operate on the QSIG interface.

Virtual Network Service

A QSIG link can be used as a B-channel for the Virtual Network Service (VNS) over a private network. All VNS services are supported as normal. QSIG is only used as a speech bearer.

Feature packaging

This feature requires the following packages:

- Digit Display (DDSP) package 19
- Integrated Services Digital Network (ISDN) package 145
- Primary Rate Access (PRI) package 146
- 2.0 Mbit Primary Rate Access (PRI2) package 154
- Overlap Signaling (OVLP) package 184
- International Primary Rate Access (IPRA) package 202
- ISDN QSIG Basic Call (QSIG) is package 263

Feature implementation

Task summary list

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

1. Table 142: LD 17 - Assign Configuration Record. on page 329
2. Table 143: LD 16 - Define the Route Data Block. on page 330
3. Table 144: LD 97 - System Configuration. on page 330

Table 142: LD 17 - Assign Configuration Record.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	ADAN	Action Device and Number.
- ADAN	CHG DCH x	Change input/output device. x = 0-63
...		
- - IFC	ISIG ESIG	Interface ID for ISO QSIG download table. Interface ID for ETSI QSIG download table.
...		
TIMR	(NO) YES	NO = skip timer prompt. YES = change timer value.

Prompt	Response	Description
T310	110 - (120)	The timer is used to define how long the system can wait for the response message when the QSIG outgoing call is in an outgoing call proceeding state.

Table 143: LD 16 - Define the Route Data Block.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	RDB	Route Data Block.
DTRK	YES	Digital Trunk Route.
...		
DGTP	PRI PRI2 BRI	1.5 Mbit PRI. 2.0 Mbit PRI. Basic Rate Interface.
IFC	ISIG ESIG	New DCH interface identification as entered in LD 17.

Table 144: LD 97 - System Configuration.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	SYSM	System parameters for Multi-purpose Serial Data Link (MSDL) or Multi-purpose ISDN Signaling Processor (MISP).
...		
FDLC	 p1 p2 p3 p4	Fast download control parameters. p1 = Application will be set to PRIE or BRIE. p2 = Specifies conditional (C) or forced downloading (F). p3 = Specifies current (C), latest (L) and specified (S) version. p4 = xx (version number if p3 = S).

Feature operation

No specific operating procedures are required to use this feature.

Chapter 26: ISDN QSIG-BC and QSIG-GF Compliance Update

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

Two European QSIG interfaces are available. One version (called the ISDN QSIG-BC and QSIG-GF Compliance Update) is compliant with latest editions of the standards. The other version is compliant with older versions of the standards.

The ISDN QSIG-BC and QSIG-GF Compliance Update feature is compliant with the following European standards documents for QSIG:

- ETS 300-172, fourth edition (1997)
- ETS 300-239, second edition

Initially, support for the ISDN QSIG interface and the basic call capability was introduced. This version of the interface was based on the ETS 300-172, first edition (1990) document.

Later, the QSIG-GF Transport interface was introduced, in accordance with the ETS 300-239 (1993) standard document. The basic call functionality was upgraded to be compliant with the second edition of the ETS 300-172 document.

QSIG Basic Call Compliance Update

Comprehension required IEs

When an incoming Information Element (IE) coded "comprehension required" is not recognized, the system treats the message the same way it treats messages with a mandatory information element missing. A message is sent back with cause 96 "mandatory information element missing." The messages returned are shown in Table 145: Unrecognized comprehension required IE treatment on page 332.

Table 145: Unrecognized comprehension required IE treatment

Incoming message	Message returned
SETUP	RELEASE COMPLETE with cause 96
DISCONNECT	RELEASE with cause 96
RELEASE	RELEASE COMPLETE with cause 96
RELEASE COMPLETE	No message is returned (see note 1)
Other	STATUS with cause 96

Note:

The message is treated as if it is received with cause 31 "Normal, unspecified."

Transit counter

The transit counter IE is defined as a codeset 4 IE instead of codeset 5. The transit counter functionality is considered to be a supplementary service.

Restart procedure

When a RESTART message with a restart IE coded "Channel indicated" is received with no Channel ID IE, the message is treated as if a mandatory IE is missing. A STATUS message is returned with cause 96, (mandatory IE missing).

Progress indicator

Progress indicators 15, 16, 17 and 18, defined in the European Computer Manufacturer's Association (ECMA) coding are not defined for this interface.

QSIG GF Compliance Update

Facility IE coding

Figure 16: Facility IE layout on page 333 illustrates the layout of the Facility IE.

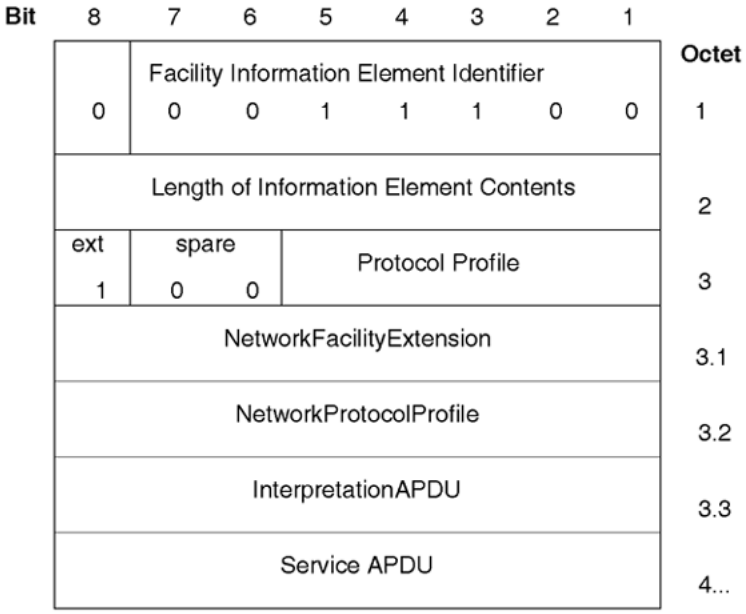


Figure 16: Facility IE layout

Protocol profile coding

Table 146: Protocol Profile Coding on page 334 illustrates three different examples of coding in the Protocol Profile.

Table 146: Protocol Profile Coding

Bits					
5	4	3	2	1	
1	0	0	0	1	ROSE
1	0	0	1	1	ACSE
1	1	1	1	1	Networking Extension
All other values are reserved.					

On transmission, the system codes the Protocol Profile as "Networking Extension."

On reception, the system supports Protocol Profile coded as "ROSE" or "Networking Extension."

Network Protocol Profile coding

The Network Protocol Profile is used to specify the Service APDU coding type. Since the system only supports ROSE (the default coding type), the Network Protocol Profile is always omitted.

NFE and interpretation coding

No change is introduced by this update in the coding of the NFE and of the interpretation APDU.

Service APDU coding

The Service APDU coding is service dependent. It is not affected by this update.

Operating parameters

The earlier version of QSIG is still available, to be used in networks where supplementary services are involved.

The ISDN QSIG-BC and QSIG-GF Compliance Update feature is not available on PRI 1.5 Mbit links. These links are usually configured with ISO QSIG GF.

The following optional parts of the protocol are not supported by the ISDN QSIG-BC and QSIG-GF Compliance Update feature:

- Message segmentation
- Multi-rate
- Indication of the channel with a map in the Channel ID IE

There are two choices of PRI2 hardware:

- PRI2 card (NT8D72) and MSDL card (NT6D80)
- Dual port DTI/PRI2 card and either the Downloadable D-channel Daughterboard card (NTBK51) or the MSDL card (NT6D80)

These systems require the NTRB53 or the QPC471 Clock Controller card.

The hardware required for CS 1000E is:

- PRI2 card (NTBK50)
- Downloadable D-channel Daughterboard card (NTBK51)
- Clock Controller (NTAK20)

The hardware required for ISDN BRI is as follows:

- SILC card (NT6D70)
- MISP card (NT6D73) for Large Systems
- MISP card (NTBK22) for CS 1000E

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature requires the following packages:

- ISDN Signaling (ISDN) package 145
- Primary Rate Access (PRA) package 146
- 2 Mbit/s PRI (PRI2) package 154
- International PRA (IPRA) package 202
- Multi-Purpose Serial Data Link (MSDL) package 222
- QSIG Interface (QSIG) package 263
- QSIG GF Transport (QSIGGF) package 305

The following software packages are required for ISL QSIG:

- ISDN Signaling (ISDN) package 145
- ISDN Signaling Link (ISL) package 147

- Multi-Purpose Serial Data Link (MSDL) package 222
- QSIG Interface (QSIG) package 263
- QSIG GF Transport (QSIGGF) package 305

The following software packages are required for ISDN BRIT QSIG:

- ISDN Signaling (ISDN) package 145
- ISDN Basic Rate Access (BRI) package 216
- Multi-Purpose Serial Data Link (MSDL) package 222
- ISDN BRI Trunk Access (BRIT) package 233
- QSIG Interface (QSIG) package 263
- QSIG GF Transport (QSIGGF) package 305

Feature implementation

Task summary list

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

1. Table 147: LD 17 - Configure the updated interface type for the D-channel. on page 336
2. Table 148: LD 16 - Configure the updated interface type for the trunk route. on page 337

Table 147: LD 17 - Configure the updated interface type for the D-channel.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	CFN	Configuration Record.
ADAN	CHG DCH x	Change D-Channel number.
...		
- IFC	ESGF ISGF	ETSI QSIG interface with GF platform. ISO QSIG interface with GF platform.

Table 148: LD 16 - Configure the updated interface type for the trunk route.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	RDB	Route Data Block.
CUST	0-99	Customer number as defined in LD 15.
ROUT	0-511	Route number for Large System and CS 1000E system.
DTRK	YES	Digital Trunk Type.
- DGTP	PRI BRI	PRI trunk. BRI trunk.
- IFC	ESGF ISGF	ETSI QSIG interface with GF platform. ISO QSIG interface with GF platform.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 27: ISDN QSIG Call Diversion Notification

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

ISDN QSIG Call Diversion Notification allows the system to provide notification to the originating calling party and forwarded-to (diverted) party when a call is forwarded over an ISDN QSIG private network. With this feature, call diversion is invoked by the following triggers:

- Call Forwarding Busy (CFB)—the called phone is in use
- Call Forwarding No Reply (CFNR)—the called phone does not answer after a given number of rings
- Call Forwarding Unconditional (CFU)—the called phone is forwarded as soon as the phone is reached

With this feature, notification can be provided to the calling party and the forwarded-to (diverted) party that call diversion has occurred. When provisioned, call diversion notification is displayed on the calling party's phone and on the forwarded-to (diverted) party's phone according to the existing terminal display configuration.

Diversion notification provided to the calling and forwarded-to (diverted) party depends on the subscription options (Class of Service) configured on the called party's phone. The called party can configure two diversion options. One Class of Service option is applicable to the calling party and the other option is applicable to the forwarded-to (diverted) party. These new Classes of Service determine if call diversion notification is provided to the calling party and the forwarded-to party and what type of diversion information these parties receive.

Figure 17: Call Diversion Environment on page 341 shows an example of a Call Diversion environment.

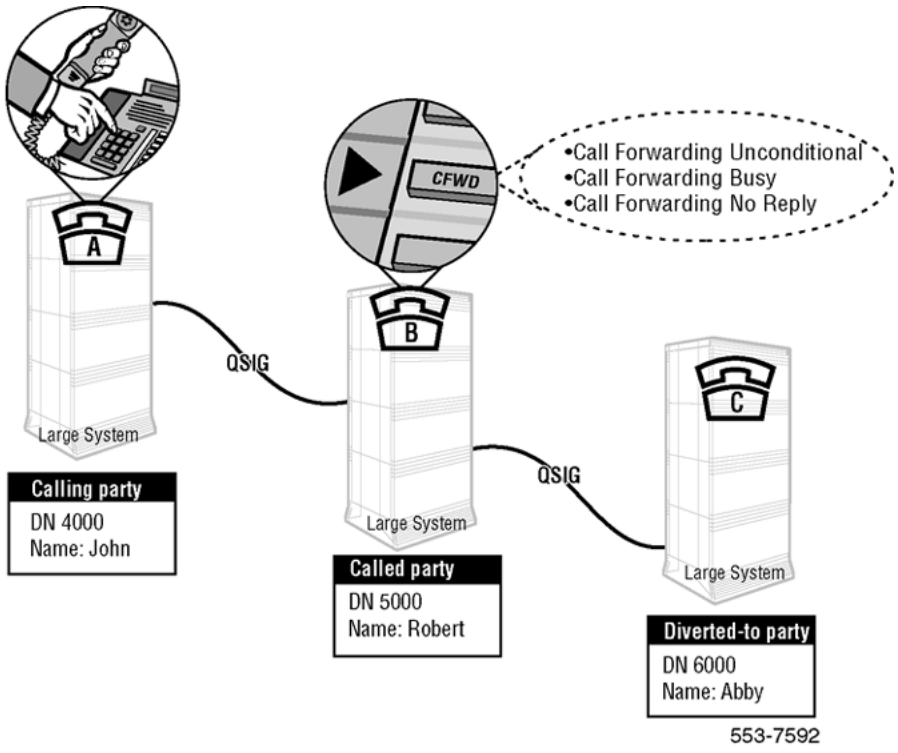
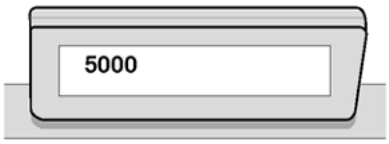
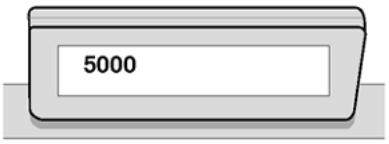


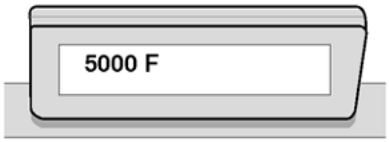
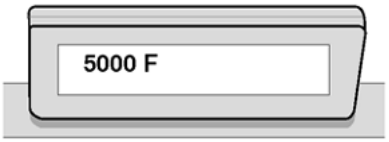
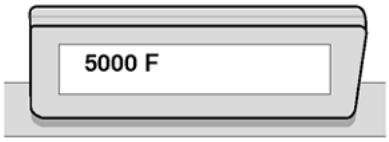
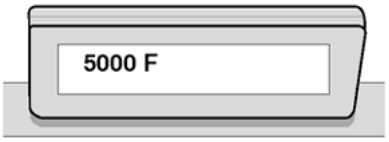
Figure 17: Call Diversion Environment

Calling Party's Notification

The calling party that originates the call receives notification of call diversion depending on the configuration of the called party's phone. As shown in Table 149: Calling Party's Notification on page 341, the called party configures one of the three possible Classes of Service.

Table 149: Calling Party's Notification

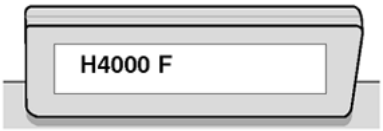
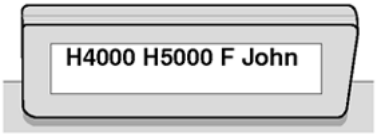
Called Party's Diversion Subscription Options (Class of Service)	Calling Party's Display when call is forwarded.	Forwarded-to Party's Display
CLS = DNO1 No diversion notification provided to calling party.		

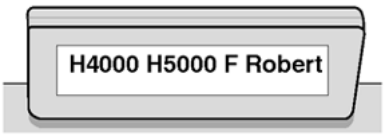
Called Party's Diversion Subscription Options (Class of Service)	Calling Party's Display when call is forwarded.	Forwarded-to Party's Display
CLS = DNO2 Diversion notification provided without forwarded-to (diverted) party's number and name.		
CLS = DNO3 (default) Diversion notification provided with forwarded-to (diverted) party's number and name when available.		

Called Party's Notification

The forwarded-to (diverted) party receives notification of call diversion depending on the configuration of the called party's phone. As shown in Table 150: Forwarded-to Party's Notification on page 342, the called party configures one of the two possible Classes of Service.

Table 150: Forwarded-to Party's Notification

Called Party's Diversion Subscription Options (Class of Service)	Forwarded-to Party's Display.	Forwarded-to Party's Class of Service
DNDN = No Called party's number and name are not released to the forwarded-to (diverted) party.		DNDD = Dialed Name Display Denied (default)
DNDY = Yes (default) Called party's number and name are released to forwarded-to (diverted) party.		DNDD = Dialed Name Display Denied (default)

Called Party's Diversion Subscription Options (Class of Service)	Forwarded-to Party's Display.	Forwarded-to Party's Class of Service
DNDY = Yes (default) Called party's number and name are released to forwarded-to (diverted) party.		DNDA = Dialed Name Display Allowed

On the forwarded-to (diverted) party's phone, the Dialed Name Display Allowed/Denied (DNDA/DNDD) Class of Service allows the forwarded-to (diverted) party to choose the name displayed after call diversion. If the forwarded-to (diverted) party configures a Dialed Name Display Allowed (DNDA) Class of Service their display shows one of the following:

- the original called party's name when available, or
- if the original called party's name is not available, the redirecting name when available, or
- if neither of the two previous names are available, then nothing is displayed.

If the forwarded-to (diverted) user has configured a Dialed Name Display Denied (DNDD) Class of Service then the calling party's name is displayed if available. Otherwise, nothing is displayed.

Operating parameters

A system user cannot remotely activate or deactivate QSIG Call Diversion Notification from either a system or on another third party Private Branch Exchange (PBX).

According to the current operation of the Call Forward features, the system does not provide any indication, such an LED, on the called party's phone that call diversion has been activated.

The validity of the forwarding number is not supported. If an invalid forwarding directory number is dialed no verification is provided by the system.

The calling party is notified each time a call redirection occurs, provided this information is supplied by the network. If previous call diversion notification is provided, then it is replaced by last received. If call redirection occurs and no call redirection information is provided by the network, then any notification information that has previously been provided remains unchanged.

When a call redirection occurs no call diversion information is provided by the network. However, if a previous notification has been provided then the call diversion information remains unchanged.

The maximum allowable size of QSIG name information is 50 characters. With the Call Party Name Display feature, only a 27-character string length is supported. Accordingly, names processed by the system can be truncated.

Across a network, it is not possible to prevent reciprocal Call Forward from occurring. However, the capabilities of the Flexible Orbiting Prevention Timer and Total Redirection Count features can be used.

QSIG Call Diversion Notification requires Coordinated Dialing Plan (CDP) or Uniform Dialing Plan (UDP). Digit manipulation is not supported.

Interworkings with Meridian Customer Defined Network (MCDN) call redirection and QSIG Call Diversion Notification is not supported.

Interaction with Meridian Mail and this feature is not supported.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

QSIG Call Diversion Notification is ISDN QSIG Supplementary Service - Call Completion (QSIG-SS) package 316, which has the following dependencies:

- Digit Display (DDSP) package 19
- Network Alternate Route Selection (NARS) package 58 (for Uniform Dialing Plan), or Coordinated Dialing Plan (CDP) package 59
- Calling Party Name Display (CPND) package 95
- QSIG Interface (QSIG) package 263
- ISDN QSIG-GF Transport (QSIG-GF) package 305

ISDN PRI QSIG Interface requires:

- Primary Rate Access (PRI) package 146
- 2.0 Mbit Primary Rate Interface (PRI2) package 154
- International Primary Rate Access (IPRA) package 202
- Multipurpose Serial Data Link (MSDL) package 222

ISDN Basic Rate Interface Trunk Access (BRIT) QSIG Interface requires:

- Basic Rate Interface (BRI) package 216
- ISDN Basic Rate Interface Trunk Access (BRIT) package 233

Feature implementation

Task summary list

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

1. Table 152: LD 17 - Configure or remove remote capabilities D-channel for QSIG Call Diversion Notification. on page 347
2. Table 153: LD 16 - Configure Remote Capability for a route with QSIG Generic Functional Interface. on page 348
3. Table 154: LD 10 - Configure QSIG Diversion Notification on analog (500/2500-type) phones. on page 348
4. Table 155: LD 11 - Configure QSIG Diversion Notification on Meridian 1 proprietary phones. on page 349
5. Table 156: LD 27 - Configure Basic Rate Interface phones. on page 350
6. Table 157: LD 95 - Modify Call Party Name Display data block. on page 351
7. Table 158: LD 95 - Define Name for phones. on page 351
8. Table 159: LD 11 - Configure the name that is to be displayed on the Forwarded-to (diverted) Party's phone. on page 352

The QSIG Call Diversion Notification feature requires the configuration of one of the following call redirection features: Call Forward All Calls, Call Forward/Hunt Override Via Flexible Feature Codes, Call Forward Remote (Attendant and Networkwide), Call Forward No Answer/ Flexible Call Forward No Answer, Call Forward No Answer Second Level, Call Forward Internal Calls and User Selectable Call Redirection. The implementation of these features is not modified.

Remote Capability Meanings

Table 152: LD 17 - Configure or remove remote capabilities D-channel for QSIG Call Diversion Notification. on page 347 indicates the remote capabilities meanings for both D-channel and BRI routes. When using Remote Capablility Meanings consider the following:

- Only nodes subject to be Originating, Served, Diverted or Rerouting nodes with respect to QSIG Call Diversion Notification need to have diversion remote capability configured. Transmit nodes pass the information transparently.
- When choosing the Operation Coding Choice, the interface type should be considered. When the QSIG interface used is ISO (IFC ISGF), operations are usually coded with Integer Values. For ETSI interfaces (IFC ESGF), the operation coding depends on the standard version. The latest ones are likely to use Integer Values, whereas some older ones can use Object Identifiers.

Only one remote capability allows the QSIG Diversion configuration on a D-channel/BRI route. This remote capability gathers the three following possibilities for the D-channel/BRI route:

1. Coding of operations is sent to the remote switch, which can be coded either as Object Identifier or as Integer Value. If coded as Object Identifier, the remote capability ends with an 'O', whereas for Integer Value, the remote capability ends with an 'I'. This means that remote capabilities explained below in 2 and 3 are defined twice.
2. Sending of QSIG Diversion Notification Information to the remote switch: this information is sent only if the remote capability is of first or third type, i.e. DV1x or DV3x, where the x is either 'I' or 'O' as explained in 1.
3. Treating Rerouting requests received from the remote switch: a rerouting request is only processed if the remote capability is of second or third type, i.e. DV2x or DV3x, where x is either 'I' or 'O' as explained above in 1.

Table 151: Remote Capability Meanings

Remote capability	Meaning for Operation Coding	Meaning for Notification Information	Meaning for Rerouting request
None of the following remote capabilities.	Not applicable (nothing sent)	Not sent	Not processed when received
DV1O	Sent coded as Object Identifier	Sent	Not processed when received
DV1I	Sent coded as Integer Value		
DV2O	Sent coded as Object Identifier	Not Sent	Processed when received

Remote capability	Meaning for Operation Coding	Meaning for Notification Information	Meaning for Rerouting request
DV2I	Sent coded as Integer Value		
DV3O	Sent coded as Object Identifier	Sent	Processed when received
DV3I	Sent coded as Integer Value		

Table 152: LD 17 - Configure or remove remote capabilities D-channel for QSIG Call Diversion Notification.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	ADAN	Action device and number.
ADAN	CHG DCH x	Configure new remote capabilities on D-channel x.
...		
IFC	xxxx	QSIG interface type where xxxx is: ESGF = interface ID for ETSI QSIG with GF capability. ISGF = Interface ID for ISO with GF capability.
...		
RLS	xx	Software release.
...		
RCAP		QSIG SS Call Diversion Notification remote capability. Configure sending of QSIG Diversion Notification Information, treatment of Rerouting request and coding of operations. If coded as Object Identifier, the remote capability ends with 'O", whereas for Integer Value, the remote capability ends with 'I'. Only one remote capability is allowed. Refer to Table 1 for further information.
	DV1I DV1O DV2I DV2O DV3I DV3O	Diversion information is sent to remote switch. Rerouting requests from remote switch are processed. Diversion information is sent to remote switch. Rerouting requests from remote switch are processed. Diversion information is sent to remote switch. Rerouting requests from remote switch are processed. Precede with 'X' to remove capability.

Table 153: LD 16 - Configure Remote Capability for a route with QSIG Generic Functional Interface.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	RDB	Route Data Block.
CUST	0-99	Customer number as defined in LD 15.
ROUT	0-511	Route number for Large System and CS 1000E system.
IFC	xxxx	QSIG interface type where xxxx is: ESGF = interface ID for ETSI QSIG with GF capability. ISGF = Interface ID for ISO with GF capability.
RCAP		QSIG SS Call Diversion Notification remote capability. Configure sending of QSIG Diversion Notification Information, treatment of Rerouting request and coding of operations. If coded as Object Identifier, the remote capability ends with 'O", whereas for Integer Value, the remote capability ends with 'I'. Only one remote capability is allowed. Refer to Table 1 for further information.
	DV1I DV1O DV2I DV2O DV3I DV3O	Diversion information is sent to remote switch. Rerouting requests from remote switch are processed. Diversion information is sent to remote switch. Rerouting requests from remote switch are processed. Diversion information is sent to remote switch. Rerouting requests from remote switch are processed. Precede with 'X' to remove capability.

Table 154: LD 10 - Configure QSIG Diversion Notification on analog (500/2500-type) phones.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	500	Type of phone.
...		
TN	l s c u	Terminal Number for Large System and CS 1000E.
DN	x...x yyy	Directory Number and Calling Line Identification entry

Prompt	Response	Description
		(Range is (0) - value entered for SIZE prompt in LD 15 minus one).
- CPND	aaa	Calling Party Name Display where: NEW = Add data block. CHG = Change existing data block. OUT = Remove existing data block.
- -CPND_LANG	aaa	Calling Party Name Display Language where: (ROM) = Roman. KAT = Katakana.
- - NAME	aaaa bbbb	Calling Party Name Display Name where: First name Last name.
CLS	(DNO3) DNO1 DNO2	QSIG Call Diversion Notification options for calling party. Notification with diverted-to party's number and name. No Notification Notification without diverted-to party's number and name.
	(DNDY) DNDN	QSIG Call Diversion Notification options concerning forwarded- to party. (DNDY)=Notification with Called party's number and name. DNDN=Notification without Called party's number and name.
	CNDA CFXA CFTA DDGA FNA HTA NAMA	Call Party Name Display Allowed Call Forward to External DN Allowed. Call Forward by Call Type Allowed. Directory Number Display on other phone Allowed. Call Forward No Answer Allowed. Hunting Allowed. Name Display on other phone Allowed or (Denied).

Table 155: LD 11 - Configure QSIG Diversion Notification on Meridian 1 proprietary phones.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	xxxx	Type of phone.
...		
TN	l s c u	Terminal Number for Large System and CS 1000E.
FDN	x...x	Flexible Call Forward No Answer DN for an Internal Directory Number.

Prompt	Response	Description
CLS	(DNO3) DNO1 DNO2	QSIG Call Diversion Notification options for calling party. (DNO3)=Notification with diverted-to party's number and name. DNO1=No notification. DNO2=Notification without diverted-to party's number and name.
	(DNDY) DNDN	QSIG Call Diversion Notification options concerning forwarded- to party. (DNDY)=Notification with Called party's number and name. DNDN=Notification without Called party's number and name.
	CNDA CFXA CFTA DDGA FNA HTA NAMA	Call Party Name Display Allowed Call Forward to External DN Allowed. Call Forward by Call Type Allowed. Directory Number Display on other phone Allowed. Call Forward No Answer Allowed. Hunting Allowed. Name Display on other phone Allowed or (Denied).
...		
RCO	(0) - 2	Ringing Cycle Option for Call Forward No Answer.
EFD	x...x	Call Forward No Answer Directory Number for external calls
...		
- CPND	aaa	Calling Party Name Display where: NEW = Add data block. CHG = Change existing data block. OUT = Remove existing data block.
- -CPND_LANG	aaa	Calling Party Name Display Language where: (ROM) = Roman. KAT = Katakana.
- - NAME	aaaa bbbb	Calling Party Name Display Name where: First name Last name.

Table 156: LD 27 - Configure Basic Rate Interface phones.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	TSP	Terminal Service Profile.
DSL	l s c dsl	Digital Subscriber Loop for Large System and CS 1000E system.

Prompt	Response	Description
USID	x	User Service Identifier.
DN	xxx...x	Directory Number associated with TSP (1 to 7 digits).
- CTI	VCE DTA	Call type where VCE = circuit switched voice DTA = circuit switched data.
FEAT	CFXA	Call Forward Allowed to external Directory Number.
FEAT	xxxx	QSIG Call Diversion Notification for calling party where xxxx: DNO1 = no notification DNO2 = notification without forwarded-to (diverted) party's number and name (DNO3) = notification with forwarded-to (diverted) party's number and name when available.
FEAT	xxxx	QSIG Call Diversion Notification for forwarded-to (diverted) party where xxxx: DNDN = no notification of called party's number and name notification (DNDY) = notification with called party's number and name when available (default).
SSRV_ETSI	aaa mmm nnn	ETSI Supplementary Service where: VCFW = Voice Call Forward and DCFW = Data Call Forward.

Table 157: LD 95 - Modify Call Party Name Display data block.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	CPND	Call Party Name Display data block.
CUST	0-99	Customer number as defined in LD 15.
RESN	YES	Display Redirection reason allowed. (NO) = Display Redirection reason denied (default).
- CFWD	aaaa	Mnemonic for Call Forward All Call display. (F) = default
- CFNA	aaaa	Mnemonic for Call Forward No Answer display (N) = default.
- HUNT	aaaa	Mnemonic for Call Forward Busy display. (B) = default

Table 158: LD 95 - Define Name for phones.

Prompt	Response	Description
REQ	NEW	Add new data.
TYPE	NAME	Name.
CUST	0-99	Customer number as defined in LD 15.

Prompt	Response	Description
DN	x..x	Directory Number.
- NAME	a..a	Calling Party Name Display in ASCII characters.

Table 159: LD 11 - Configure the name that is to be displayed on the Forwarded-to (diverted) Party's phone.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	a...a	Telephone type. Type **?** for a list of possible responses.
TN	l s c u	Terminal Number for Large System and CS 1000E.
CLS	DNDA	Dialed Name Display Allowed. (DNDD)= Dialed Name Display Denied (default).

Note:

LD 81 can be used to print counting or listing queries pertaining to the new Classes of Service, by entering CNT or LST against the REQ prompt, and the appropriate COS against the FEAT prompt.

Note:

LD 83 can be used to print the new Classes of Service in the TN blocks, by entering TNB against the REQ prompt, and the appropriate COS against the Class of Service prompt.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 28: ISDN QSIG Call Diversion Notification Enhancements

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

The ISDN QSIG Call Diversion Notification feature allows the system to provide notification to the originating calling party and forwarded-to (diverted) party when a call is forwarded over an ISDN QSIG private network. With this feature, call diversion is invoked by the following triggers:

- Call Forwarding Busy (CFB)—the called phone is in use
- Call Forwarding No Reply (CFNR)—the called phone does not answer after a given number of rings
- Call Forwarding Unconditional (CFU)—the called phone is forwarded as soon as the phone is reached

With this feature, notification can be provided to the calling party and the forwarded-to (diverted) party that call diversion has occurred. When provisioned, call diversion notification is displayed on the calling party's phone and on the forwarded-to (diverted) party's phone according to the existing terminal display configuration.

Diversion notification provided to the calling and forwarded-to (diverted) party depends on the subscription options (Class of Service) configured on the called party's phone. The called party can configure two diversion options. One Class of Service option is applicable to the calling party and the other option is applicable to the forwarded-to (diverted) party. These Classes of Service determine if call diversion notification is provided to the calling party and the forwarded-to party and what type of diversion information these parties receive.

The QSIG Call Diversion Notification service is provided across an ISDN QSIG network using the QSIG Generic Functional (GF) protocol transport platform.

With this enhancement, the ISDN QSIG Call Diversion Notification Enhancements feature allows the QSIG Call Diversion Notification functionality to be also supported over a QSIG/Meridian Customer Defined Network (MCDN) and a QSIG/Digital Private Signaling System No.1 (DPNSS1) network, as well as a QSIG network. According to the existing implementation, the QSIG Call Diversion Notification Enhancements feature is supported on ISDN Primary Rate Interface and ISDN Basic Rate Interface Trunking (BRIT) interfaces.

The notification of the originating phone and the diverted-to phone now depends on the type of signaling protocol that is in play (QSIG, DPNSS1, or MCDN), at what stage of the call establishment that the diversion takes place, and the user's subscription option.

Operating parameters

There are no operating parameters associated with this feature.

Determination of the rerouting node

The determination of the rerouting node depends on the type of diversion that is configured for the node, either Call Diversion by Rerouting or Call Diversion by Forward Switching.

If the Forward Switching method is implemented (call diversion is generated by the system or another third-party PBX also using the Forward Switching method), the rerouting node is the served node. If the Call Diversion by Rerouting method is implemented (a third-party switch using the Rerouting method generates a diversion), the rerouting node is the originating node.

The Forward Switching method is the one presently implemented on the system. It performs the diversion by joining together the first connection from the originating user's node to the served user's node, and the second new connection from the served user's node to the diverted-to user's node. In fact, the rerouting node is the served user's node.

Notifications rules

This section describes Originating and Diverted-to user's notification rules, for single as well as multiple diversions in a pure QSIG environment.

Single Diversion

Originating and Diverted-to user's Notifications are detailed for the following scenario: "Station A calls Station B. Station B has activated CFU, CFB or CFNR to Station C. Station C answers".

Originating users notification rules

Diversion Reason Notification rules:

- This reason is displayed or not on the originating user's phone (as soon as Diversion Notification information is received from the served node) according to the served user's Subscription option (see Table 160: Originating users notification versus served users Subscription option on page 356).

Diverted-to Number Notification rules:

- The diverted-to number (received in the Diversion Notification information delivered by the served node) is displayed on the originating user's phone:
  - if the received presentation information (received in the Diversion Notification information issued from the diverted-to node) indicates that presentation is allowed.

- if the served user's subscription option (received within Diversion Notification information from the served node) allows it (see Table 160: Originating users notification versus served users Subscription option on page 356).

Diverted-to Name Notification rules:

- The diverted-to name, when available (as optionally received as part of the Diversion Notification information delivered by the diverted-to node), is displayed on the originating user's phone:
  - if the intrinsic name presentation (received in the Diversion Notification information issued from the diverted-to node) indicates that presentation is allowed.
  - if the served user's subscription option (received within Diversion Notification information from the served node) allows it (see Table 160: Originating users notification versus served users Subscription option on page 356).

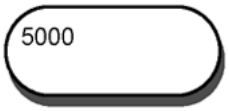
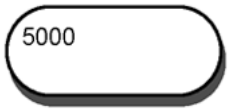
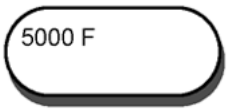
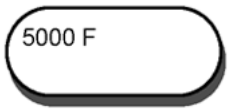
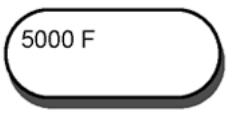
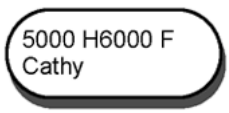
  Note that DNDA/DNDD functionality has no impact on the originating user's notification.

When both originating and served users are on the same node, the existing system treatment is still applicable, which is that the served user's subscription options have no impact on the originating user notification.

When both originating and diverted-to users are on the same system node, and diversion is performed by the Rerouting method, then the served user's subscription options are effective for the originating user notification only before the diverted-to user answers, but no name is provided. As soon as the diverted-to user answers, then the originating user receives full notification (reason, diverted-to user's number and name), according to the served user's subscription option.

Table 160: Originating users notification versus served users Subscription option on page 356 sums up originating user's Notification according rules, according to the served user's Subscription options.

Table 160: Originating users notification versus served users Subscription option

	Originating user's display	
Served user's Subscription option: "Calling user receives notification that the call has been diverted"	after receipt of served user's diversion notification information	after receipt of diverted-to user's diversion notification information
No		
Yes without diverted-to number and name		
Yes with diverted-to number and name when available		

Diverted-to users notification rules

Diversion Reason Notification rules:

- The reason is displayed on the diverted-to user's phone (as soon as Diversion Notification Information is received from the served node).

Served Number Notification rules:

- The served user's number (the originally-called number) is displayed or not on the diverted-to user's phone, according to the served user's Subscription option (see Table 161: Diverted-to users notification versus served users Subscription option and diverted-to user class of service for single diversion on page 358). If the served user's Subscription option is to not release the calling party name/number to the diverted-to user, the served user's number can be displayed with dashes (if received with a restricted presentation) or not displayed (if no served user's number is received).

Served Name Notification rules:

- The functionality DNDA/DNDD allows the diverted-to user to choose the name to be displayed after diversion has taken place:
  - if the diverted-to phone has a DNDA Class of Service, then the diverted-to user's phone displays one of the following: - the original called name when available (as optionally received from the served user's node, depending on the previous served users's Subscription Option), or - if the original called name is not available, the redirecting name when available (as optionally received from the served user's node, depending on the served user's Subscription Option), or - nothing if none of the two previous names is available.
  - if the diverted-to phone has a DNDD class of service, then the calling user's name is displayed if available, otherwise nothing is displayed.

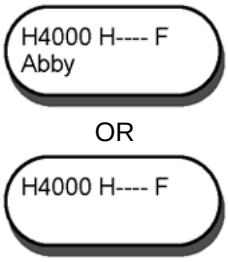
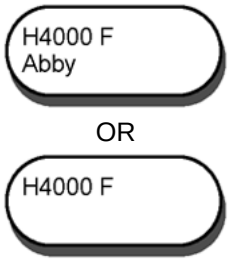
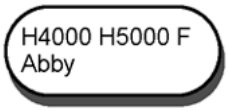
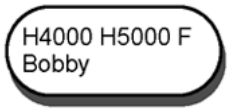
When both served and diverted-to users are on the same node, the present system treatment applies, that is, the served user's subscription options have no impact on the diverted-to user notification.

When both the originating and diverted-to users are on the same system node and diversion is performed by the Rerouting method, then served user's subscription options have no impact on the diverted-to user notification. In this case, DNDA functionality is not supported.

Table 161: Diverted-to users notification versus served users Subscription option and diverted-to user class of service for single diversion on page 358 sums up the diverted-to user's Notification rules, according to the served user's Subscription options and the diverted-to user's class of service DNDA/DNDD.

Note that, in the case presented in the example, the Originating user name is Abby, with a DN of 4000. The Served user name is Bobby, with a DN of 5000. The Diverted-to user name is Cathy, with a DN of 6000.

Table 161: Diverted-to users notification versus served users Subscription option and diverted-to user class of service for single diversion

Served user's Subscription option: "Served user releases the number/ name to the diverted-to user"	Diverted-to user's display (after receipt of served user's diversion notification information) depending on the diverted-to user class of service:	
	DNDD	DNDA
No		
Yes		

Multiple diversions

Originating and Diverted-to user's Notifications are detailed for the following scenario: "Station A calls Station B. Station B has activated CFU, CFB or CFNR to Station C. Station C has activated diversion to Station D, which itself has activated diversion to Station E. Station E has activated diversion to Station F. Station F answers".

Originating users notification rules

Diversion Reason Notification rules:

- The same rules apply according to the single diversion case (refer to Single Diversion on page 355). The last diversion reason will replace the previous one.

Diverted-to Number Notification rules

The diverted-to number (as received in the last Diversion Notification information message issued from the last served node) is presented to the calling user if all of the following conditions apply:

- all previously Diversion Notifications information messages received at the originating node contain a subscription option with a value of "Yes with diverted-to number/name", and
- any previously Diversion Notification information message issued from the diverted-to node contain a presentation indicator that allows presentation.

The last diverted-to number will replace the previous one.

Diverted-to Name Notification rules

The diverted-to name, when available (as optionally received within the Diversion Notification information message delivered by the diverted-to node), is presented to the calling user if all of the following conditions apply:

- all previously Diversion Notifications information messages received at the originating node include a subscription option with a value of "Yes with diverted-to number/name", and
- the intrinsic name presentation is allowed.

The last diverted-to name will replace the previous one. Note that DNDA/DNDD functionality has no impact on the originating user's notification.

Diverted-to users notification rules

Diversion Reason Notification rules:

- The same rules apply according to the single diversion case (Single Diversion on page 355).

Served Number Notification rules:

- Either the originally-called number (as soon as a Diversion Notification information message is received from the last served node) is displayed or not on the diverted-to user's phone, according to the first served user's Subscription option.

Served Name Notification rules:

The same rules, including the DNDA/DNDD functionality, apply as for the single diversion case.

Table 162: Diverted-to users notification versus served users Subscription option and diverted-to user class of service for multiple diversions on page 360 sums up a multiple diversion case considering the different parties' Subscription option, and the diverted-to user class of service.

Note:

In the example, the Originating user name is Abby, with a DN of 4000. The first Served user name is Bobby, with a DN of 5001. The second Served user name is Billy, with a DN of 5002. The Diverted-to user name is Cathy, with a DN of 6000.

Table 162: Diverted-to users notification versus served users Subscription option and diverted-to user class of service for multiple diversions

Served users' Subscription option: "Served user releases the number/name to the diverted-to user"		Diverted-to user's display (after receipt of diversion notification information) depending on the diverted-to user Class of Service:	
Bobby	**Billy**	**DNDD**	**DNDA**
No	No	H4000 H---- F Abby OR H4000 H---- F	H4000 F Abby OR H4000 F
Yes	No	H4000 H5001 F Abby	H4000 H5001 F Bobby
No	Yes	H4000 H---- F Abby OR H4000 H---- F Billy	H4000 F Abby OR H4000 F Billy
Yes	Yes	H4000 H5001 F Abby	H4000 H5001 F Bobby

Feature interactions

Call Forward by Call Type

This feature allows to perform CFNA or Hunt to a different DN (FDN/EFD or HUNT/EHT) whether the call is internal or external.

This feature is supported by QSIG Diversion. The definition of an Ôinternal call' is not modified by the introduction of the QSIG Call Diversion Enhancements feature. An ISDN trunk call using

private numbering is considered internal (note that no attempt is made to determine the real originating party with QSIG, only the type of numbering plan for the QSIG call is used).

Call Waiting Redirection

The Call Waiting Redirection (CWTR) feature allows unanswered calls given Call Waiting treatment to receive call redirection through Call Forward No Answer (CFNA) for the active phone. An unanswered call receiving Call Waiting treatment is redirected to the active phone's CFNA DN after the expiration of the customer data block defined CFNA timer selected for that phone. This redirected call is treated as an unanswered call by the CFNA DN (the call receives messaging services such as Avaya CallPilot, Voice Mail, and Message Centers).

The QSIG Call Diversion Notification feature handles this feature as a usual CFNA.

Call Forward Subscription Option

The behavior of the Call Forward Subscription Option is not changed. The QSIG Call Diversion Notification Subscription Options are defined as Classes of Service, but the active one will always be the Served User's Subscription Option, independently of the OPT overlay configuration (CFO/CFF).

Phantom TN

A Phantom TN cannot be remotely activated, deactivated or interrogated through QSIG.

When a Phantom TN is Call Forwarded, the QSIG Diversion Notification is handled as if the Phantom TN was a normal DN.

Call Forward to Trunk Access Code

The CFTA prompt in LD 15, the Customer Data Block, is used to allow or disallow a Call Forward Trunk Access Code to be assigned to Call Forward All Calls DID calls.

Since QSIG Call Diversion is only supported on CDP or UDP dialing plans, it cannot be activated by CFTA.

Flexible Orbiting Prevention Timer

The Flexible Orbiting Prevention Time is supported on QSIG Diversion. However, since it limits the number of diversions that can be performed by a phone in a given time, it is recommended that the Total Redirection Count capability be used instead.

Call Forward/Hunt Override

This feature is not supported on QSIG Diversion. A system user can neither originate nor receive a call over a QSIG link through Call Forward/Hunt Override. At gateways between QSIG and MCDN, this information is lost.

Access Restrictions/Trunk Group Access Restrictions

Access restrictions are checked on the served node. This means that QSIG Diversion is not performed if the served user is not able to access the route to the diverted-to node.

Dialing plans

ISDN QSIG GF transport platform supports both the ISO QSIG GF specification and the ETSI QSIG GF specification. The same numbering plans and types of numbers supported by QSIG basic call are also supported by ISDN QSIG GF transport for any DN address translation required. They are summarized below:

E.164 Numbering Plan:

- Unknown
- International Number
- National Number
- Subscriber Number

Private Numbering Plan:

- Unknown
- ESN LOC (level 1 regional number)
- CDP (local number)

Unknown Numbering Plan:

- Same as for QSIG basic call. An Unknown Numbering Plan is treated as a Private numbering plan, with an Ôunknown' type of number.

Meridian Link

Present Call Indication

A Present Call Indication message contents an IE called "Call Type", which contains Diversion information about the incoming call. In case of a QSIG Diverted incoming call, this field is updated.

Unsolicited Status Message

When a phone stops ringing because Call Forward No Reply has been applied to a QSIG call, an Unsolicited Status Message is sent to Meridian Link.

Call Completion to a Busy Subscriber

Call Forward Unconditional and Call Forward Busy

If Call Completion to a Busy Subscriber (CCBS) is to be invoked against a busy diverted-to user, the originating PINX stores the diverted-to number issued by the served PINX in order to use it for further CCBS treatments, without any manipulation.

Call Forward Unconditional (CFU) and Call Forward Busy (CFB) take precedence over CCBS.

Call Completion on No Reply

Call Forward Unconditional, Call Forward Busy, and Call Forward No Reply

If Call Completion on No Reply (CCNR) is to be invoked against a diverted-to user who does not answer, the originating PINX stores the diverted-to number issued by the served PINX in order to use it for further CCNR treatments, without any manipulation.

Call Forward No Reply (CFNR) takes precedence over CCNR.

QSIG Name Display Supplementary Services

Calling Name

After a QSIG Diversion takes place, and Call Diversion Notification information is received at the diverted-to node, the calling name is displayed on the diverted-to user's phone only if this phone has a DNDD class of service. If the diverted-to user's phone has a DNDA class of service, another name or nothing is displayed.

Alerting/Connected Name

When both names from a QSIG Name Display Service and a QSIG Diversion Notification are included in a same message, the name from the QSIG Name Display Service will be ignored. Note that these names should be the same.

If a name, either from QSIG Name Display Service and QSIG Diversion Notification, is received, it will be displayed only if no subsequent subscription option forbids it.

QSIG Path Replacement

QSIG Path Replacement can be triggered after QSIG Diversion has been applied. As QSIG Diversion has its own optimization method (diversion by Rerouting), an IE indicating the method used is sent by the system during QSIG Diversion towards the diverted-to node. At

the diverted-to system node, upon receipt of this IE, QSIG Path Replacement should only be started if diversion has not already optimized the call.

QSIG Call Transfer

There is no interaction when Station A and Station B are connected after a diversion, and then Station B transfers the call to Station C.

Station A and Station B are connected. Station B transfers the call to Station C, who then diverts the call to Station D. The notification depends on when Station B completes the transfer. If the transfer is completed before the diversion occurs, then the two services do not interact. The final notification is given by the Call Diversion Notification. If the diversion is initiated before the transfer has been completed, then the final notification depends on the transfer notification rules.

QSIG Alternate Routing

When a QSIG Alternate Routing is performed, the Facility IE sent with the setup message are not re-sent. Therefore, the diverted-to user does not receive the DLI2 information and is not notified that a diversion occurred.

Preventing Reciprocal Call Forward

Configuring the Preventing Reciprocal Call Forward Allowed (PVCA) option in LD 15 prevents Call Forward looping. A phone that has been call forwarded to another phone, will not have the same call-forwarded back to it.

This functionality is not supported across a QSIG network.

ISDN BRI Supplementary Services - ETSI Call Forward Unconditional

This feature implements the ETSI BRI Call Forward Unconditional feature, which allows a BRI phone to activate Call Forward Unconditional. Two Call Forward DNs can be configured, one for voice and one for data.

A BRI phone, as a served user, has the subscription option choices to notify both originating and diverted-to parties.

Note:

If the originating or diverted-to phone is a BRI phone, notification information is provided to this phone, but the treatment by the phone is subject to the manufacturer's implementation.

Attendant and Network Wide - Remote Call Forward

The Attendant and Network Wide- Remote Call Forward feature expands the phone based Remote Call Forward functionality to apply across an MCDN ISDN network, and to allow Remote Call Forward for a phone to be activated from an attendant.

This functionality cannot be used to activate QSIG Call Diversion.

Total Redirection Count

The Total Redirection Count (TRCL) prompt in LD 15 allows the maximum number of redirections for a customer to be defined between 1 and 7 (0 by default). The counter is incremented each time the call is diverted by Call Forward All Calls, Hunt, Call Forward No Answer, or Call Forward Busy, in an MCDN environment, whether on-node or off-node. The counter is reset when the call is answered or enters the attendant queue.

This feature is supported by QSIG, which transports the equivalent information.

Note that this feature is an available solution for avoiding infinite diversion loops, since Preventing Reciprocal Call Forward is not supported network-wide.

MCDN Trunk Route Optimization

Consider the case presented in Figure 18: Call from MCDN TRO Diverted to QSIG on page 366.

For Call Forward No Answer (CFNA), call (1) is cleared before call (2) starts, which is not the usual mechanism for full QSIG CFNA.

Diversion Notification Information messages are sent from originating node to the diverted-to node. As the Served User's Subscription Option is not known (as well as the name in certain conditions), it is assumed to be "Served user releases the number/name to the diverted-to user: no". As a result, Diversion Notification Information messages provided in this case only contain the diversion counter, diversion reason and eventually the original reason for diversion.

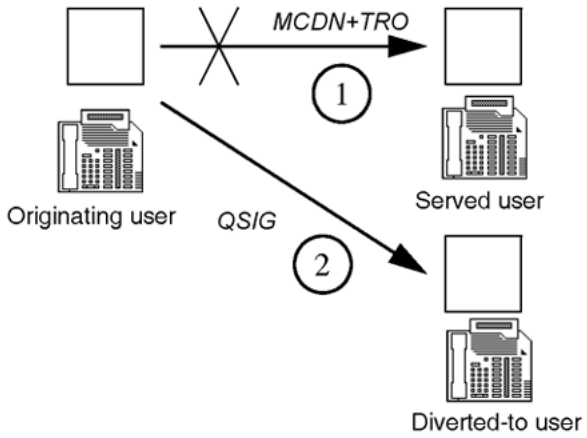


Figure 18: Call from MCDN TRO Diverted to QSIG

MCDN Trunk Route Optimization is blocked on MCDN calls due to the Rerouting method used by QSIG Diversion on No Reply, as far as the original call being kept by the originating PINX to eventually be restored for further treatment. This case arises if the diverted-to user activates TRO before alerting, that is, if Call Forwarded All Calls has been activated against the diverted-to user.

MCDN Trunk Anti Tromboning (TAT)

As the MCDN Trunk Anti Tromboning feature is activated once the call is established, there is no interaction between this feature and QSIG Call Diversion Notification.

ISDN Call Connection Limitations

QSIG Diversion is included in the Call Redirection Threshold, as part of the ISDN Call Connection Limitations feature.

Timed Reminder Recall

In case of QSIG Diversion on No Reply by the Rerouting method, if the original call is under Slow Answer Recall Timing, then the new call resulting from QSIG Diversion is also timed for Slow Answer Recall. This applies if the new diversion call is over an ETSI QSIG link, or to a local analog (500/2500-type) phone or Meridian proprietary phone, but not to a BRI phone.

The timer applied to the new diversion call is the remaining timer of the original call at the time of diversion.

Feature packaging

This feature requires the following packages:

- Coordinated Dialing Plan (CDP) package 59, or, if the Uniform Dialing Plan is used, either Network Alternate Route Selection (NARS) package 58 or Basic Alternate Route Selection (BARS) package 57
- Integrated Services Digital Network (ISDN) package 145
- Primary Rate Access (PRI) package 146
- 2.0 Mbit/s Primary Rate Access (PRI2) package 154
- International Primary Rate Access (IPRA) package 202
- Multi-purpose Serial Data Link (MSDL) package 222
- Q-reference Signaling Point Interface (QSIG) package 263
- QSIG Generic Functional Protocol Transport (QSIG-GF) package 305
- Q-reference Signaling Point Interface Supplementary Services (QSIG-SS) package 316

For ISDN Basic Rate Interface Trunking (BRIT), the following additional packages are required:

- ISDN Basic rate Access (BRI) package 216
- ISDN Basic Rate Trunk Access (BRIT) package 233

For a QSIG/DPNSS1 Gateway, the following additional packages are required:

- Integrated Digital Access (IDA) package 122
- Digital Private Networking Signaling System No. 1 (DPNSS) package 123
- DPNSS Network Services (DNWK) package 231

For notification display, the following additional packages are required:

- Digit Display (DDSP) package 19
- Call Party Name Display (CPND) package 95

Feature implementation

Task summary list

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

1. Table 164: LD 17 - Configure or remove remote capabilities D-Channel for QSIG Call Diversion Notification. on page 370
2. Table 165: LD 10 - Configure QSIG Diversion Notification Class of Service on analog (500/2500-type) phones. on page 371
3. Table 166: LD 11 - Configure QSIG Diversion Notification Class of Service, on Meridian 1 Proprietary Phones. on page 372
4. Table 167: LD 95 - Configure QSIF Call Diversion Notification mnemonics for Call Forward All Calls, Call Forward No Answer, and Call Forward Busy in the Call Party Name Display data block. on page 373
5. Table 168: LD 95 - Define a Calling Party Name to be displayed on a receiving phone. on page 373
6. Table 169: LD 11 - Configure Dialed Name Display Denied or Dialed Name Display Allowed of the originally-called party on the diverted-to phone. on page 373
7. Table 170: LD 16 - Configure the QSIG Call Diversion Notification Remote Capability for a BRI route with QSIG Generic Functional Interface. on page 374
8. Table 171: LD 27 - Configure the Terminal Service Profile (TSP) to implement the QSIG Call Diversion options on BRI phones. on page 375

Remote Capability Meanings, for PRI D-Channels and BRI routes

Table 163: Remote Capability Meanings on page 369 indicates the remote capabilities meanings for both PRI2 D-Channels and BRI routes. When using Remote Capability Meanings consider the following:

- Only nodes subject to be Originating, Served, Diverted or Rerouting nodes with respect to QSIG Call Diversion Notification need to have diversion remote capability configured. Transmit nodes pass the information transparently.
- When choosing the Operation Coding Choice, the interface type should be considered. When the QSIG interface used is ISO (IFC ISGF), operations are mostly coded with Integer Values. For ETSI interfaces (IFC ESGF), the operation coding depends on the standard version. The last ones generally use Integer Values, whereas some older ones can use Object Identifiers.

Only one remote capability allows the QSIG Diversion configuration on a D-Channel/BRI route. This remote capability gathers the three following possibilities for the D-Channel/BRI route:

1. Coding of operations is sent to the remote switch, which can be coded as either as Object Identifier or as Integer Value. If coded as Object Identifier, the remote capability ends with an ÔO', whereas for Integer Value, the remote capability ends with an ÔI'. This means that remote capabilities explained below in 2 and 3 are defined twice.
2. Sending of QSIG Diversion Notification Information to the remote switch: this information is sent only if the remote capability is of first or third type, i.e. DV1x or DV3x, where the x is either ÔI' or ÔO' as explained in 1.
3. Treating of Rerouting requests received from the remote switch: a rerouting request is only processed if the remote capability is of second or third type, i.e. DV2x or DV3x, where x is either ÔI' or ÔO' as explained above in 1.

Table 163: Remote Capability Meanings

Remote capability	Meaning for Operation Coding	Meaning for Notification nformation	Meaning for Rerouting request
None of the following remote capabilities.	Not applicable (nothing sent)	Not sent	Not processed when received
DV1O	Sent coded as Object Identifier	Sent	Not processed when received
DV1I	Sent coded as Integer Value		
DV2O	Sent coded as Object Identifier	Not Sent	Processed when received
DV2I	Sent coded as Integer Value		
DV3O	Sent coded as Object Identifier	Sent	Processed when received
DV3I	Sent coded as Integer Value		

ISDN PRI implementation procedures

The following are ISDN PRI implementation procedures:

- Configure the QSIG Call Diversion Notification emote Capability for the associated D-Channel, using LD 17.
- Configure QSIG Diversion Notification Class of Service, on analog (500/2500 type) phones, using LD 10.

- Configure QSIG Diversion Notification Class of Service, on Meridian Proprietary Phones, using LD 11.
- Configure the QSIF Call Diversion Notification mnemonics, for Call Forward All Calls, Call Forward No Answer, and Call Forward Busy, using LD 95.
- Define a Calling Party Name Display name string for a phone, using LD 95.
- On a Meridian Proprietary Phone, configure whether or not the name of the originally-called party is to be displayed on the phone, if is a diverted-to phone, using LD 11.

ISDN BRIT implementation procedures

The following are the ISDN BRIT implementation procedures:

- Configure the QSIG Call Diversion Notification Remote Capability for a BRI route with QSIG Generic Functional Interface, using LD 16.
- Configure the Terminal Service Profile (TSP), to implement the QSIG Call Diversion options on BRI phones, using LD 27.

ISDN PRI implementation

Table 164: LD 17 - Configure or remove remote capabilities D-Channel for QSIG Call Diversion Notification.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	ADAN	Action Device and Number.
- ADAN	NEW DCH xx	Add a D-Channel on port 0-63 for Large Systems.
- IFC	ESGF ISGF	ETSI QSIG interface with GF platform. ISO QSIG interface with GF platform.
RLS	xx	Software release.
RCAP		QSIG SS Call Diversion Notification remote capability. Configure sending of QSIG Diversion Notification Information, treatment of Rerouting request and coding of operations. If coded as Object Identifier, the remote capability ends with ÔO", whereas for Integer Value, the remote capability ends with ÔI'. Only one remote capability is allowed. Table 2 for further information.
	DV1I DV1O DV2I	Diversion information is sent to remote switch. Rerouting requests from remote switch are processed.

Prompt	Response	Description
	DV2O DV3I DV3O	Diversion information is sent to remote switch. Rerouting requests from remote switch are processed. Diversion information is sent to remote switch. Rerouting requests from remote switch are processed. Precede with ÔX' to remove capability, **but do not specify the coding type**; for example, XDV3 or XDV1 (and not XDV3I or XDV1O).

Table 165: LD 10 - Configure QSIG Diversion Notification Class of Service on analog (500/2500-type) phones.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	500	Phone type.
...		
TN	l s c u	Terminal Number for Large System and CS 1000E.
DN	xx...x yyy	Directory Number and Calling Line Identification entry (yyy range is (0) - value entered for SIZE prompt in LD 15 minus one).
- CPND	aaa	Calling Party Name Display where aaa can be: NEW = Add data block. CHG = Change existing data block. OUT = Remove existing data block.
- -CPND_LANG	aaa	Calling Party Name Display Language where: (ROM) = Roman. KAT = Katakana.
- - NAME	aaaa bbbb	Calling Party Name Display Name where: aaaa = First name bbbb = Last name.
...		
CLS	(DNO3) DNO1 DNO2	QSIG Call Diversion Notification options for calling party. (DNO3) =Notification with diverted-to party's number and name. DNO1 =No notification. DNO2=Notification without diverted-to party's number and name.
	(DNDY) DNDN	QSIG Call Diversion Notification options for diverted-to party.

Prompt	Response	Description
		DNDY = Notification with called party's number and name. DNDN = Notification without called party's number and name.

Table 166: LD 11 - Configure QSIG Diversion Notification Class of Service, on Meridian 1 Proprietary Phones.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	xxxx	Type of phone.
TN	l s c u	Terminal Number for Large System and CS 1000E.
FDN	x...x	Flexible Call Forward No Answer DN for an Internal Directory Number.
CLS	(DNO3) DNO1 DNO2	QSIG Call Diversion Notification options for calling party. (DNO3) =Notification with diverted-to party's number and name. DNO1=No notification. DNO2=Notification without diverted-to party's number and name.
	(DNDY) DNDN	QSIG Call Diversion Notification options for diverted-to party. (DNDY) =Notification with called party's number and name. DNDN=Notification without called party's number and name.
...		
- CPND	aaa	Calling Party Name Display where aaa can be: NEW = Add data block. CHG = Change existing data block. OUT = Remove existing data block.
- -CPND_LANG	aaa	Calling Party Name Display Language where aaa can be: (ROM) = Roman. KAT = Katakana.
- - NAME	aaaa bbbb	Calling Party Name Display Name where: aaaa = first name bbbb = last name.

Table 167: LD 95 - Configure QSIF Call Diversion Notification mnemonics for Call Forward All Calls, Call Forward No Answer, and Call Forward Busy in the Call Party Name Display data block.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	CPND	Call Party Name Display data block.
CUST	0-99	Customer number as defined in LD 15.
RESN	YES	Display Redirection reason allowed. (NO) = Display Redirection reason denied (default).
- CFWD	aaaa	Mnemonic for Call Forward All Call display. (F) = default
- CFNA	aaaa	Mnemonic for Call Forward No Answer display (N) = default.
- HUNT	aaaa	Mnemonic for Call Forward Busy display. (B) = default

Table 168: LD 95 - Define a Calling Party Name to be displayed on a receiving phone.

Prompt	Response	Description
REQ	NEW	Add new data.
TYPE	NAME	Name
CUST	0-99	Customer number as defined in LD 15.
DN	x..x	Directory Number.
- NAME	a..a	Calling Party Name Display in ASCII characters.

Table 169: LD 11 - Configure Dialed Name Display Denied or Dialed Name Display Allowed of the originally-called party on the diverted-to phone.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	a...a	Telephone type. Type **?** for a list of possible responses.
TN	l s c u	Terminal Number for Large System and CS 1000E.
CLS	(DNDD) DNDA	DNDD = Dialed Name Display Denied (the name of the originally-dialed party is not displayed on the phone). DNDA = Dialed Name Display Allowed (the name of the originally-dialed party is displayed on the phone).

Note:

LD 81 can be used to print counting or listing queries pertaining to the new Classes of Service, by entering CNT or LST against the REQ prompt, and the appropriate COS against the FEAT prompt.

Note:

LD 83 can be used to print the new Classes of Service in the TN blocks, by entering TNB against the REQ prompt, and the appropriate COS against the Class of Service prompt.

ISDN BRIT implementation

Table 170: LD 16 - Configure the QSIG Call Diversion Notification Remote Capability for a BRI route with QSIG Generic Functional Interface.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	RDB	Route Data Block.
CUST		Customer number
	0-99	Range for Large System and CS 1000E system.
	0-31	Range for Media Gateway 1000B.
ROUT		Route number
	0-511	Range for Large System and CS 1000E system.
	0-127	Range for Media Gateway 1000B.
TKTP	aaa	Trunk type.
DTRK	YES	Digital Trunk Route.
BRIP	NO	ISDN BRI packet handler route (NO is entered, since packet data is not required).
- DGTP	BRI	Digital trunk type.
ISDN	YES	Integrated Services Digital Network.
...		
- IFC	ESGF ISGF	QSIG interface type where xxxx is: Interface ID for ETSI QSIG with GF capability. Interface ID for ISO with GF capability.
...		
RCAP		QSIG SS Call Diversion Notification remote capability. Configure sending of QSIG Diversion Notification

Prompt	Response	Description
		Information, treatment of Rerouting request and coding of operations. If coded as Object Identifier, the remote capability ends with ÔO'', whereas for Integer Value, the remote capability ends with ÔI'. Only one remote capability is allowed. Table 2 for further information.
	DV1I DV1O DV2I DV2O DV3I DV3O	Diversion information is sent to remote switch. Diversion information is sent to remote switch. Rerouting requests from remote switch are processed. Rerouting requests from remote switch are processed. Diversion information is sent to remote switch. Rerouting requests from remote switch are processed. Diversion information is sent to remote switch. Rerouting requests from remote switch are processed. Precede with ÔX' to remove capability, but do not specify the coding type. For example, XDV3 or XDV1 (and not XDV3I or XDV1O).

Table 171: LD 27 - Configure the Terminal Service Profile (TSP) to implement the QSIG Call Diversion options on BRI phones.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	TSP	Terminal Service Profile.
DSL		Digital Subscriber Loop
	lll s cc dsl#	DSL location, where: • lll (superloop) = 0-156 (must be zero or a number divisible by 4) • s (shelf) = 0-1 • cc (card) = 0-15 and for CS 1000E 1–20 • dsl# (DSL location) = 0-7
USID	x	User Service Identifier.
DN	xxx...x	Directory Number associated with TSP (1 to 7 digits).
FEAT	xxxx	QSIG Call Diversion Notification for calling party where xxxx: DNO1 = no notification DNO2 = notification without forwarded-to (diverted) party's number and name (DNO3) = notification with forwarded-to (diverted) party's number and name when available.

Prompt	Response	Description
FEAT	xxxx	QSIG Call Diversion Notification for diverted-to party where xxxx: DNDN = no notification of called party's number and name notification (DNDY) = notification with called party's number and name when available (default).

Feature operation

No specific operating procedures are required to use this feature.

Chapter 29: ISDN QSIG Call Transfer Notification

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

The QSIG Call Transfer Notification feature supports the Call Transfer Notification capability over QSIG private networks, and over mixed QSIG/MCDN and QSIG/DPNSS1 networks. This applies to 2.0 Mbit/s Primary Rate Interface (PRI2), and ISDN Basic Rate Trunking (BRIT) applications.

The QSIG Call Transfer supplementary service, provided the capability of using the Transfer by Call Join function to join two separate QSIG calls, resulting from a call transfer activation,

into one call between the originating party and the transferred-to party. However, there was no notification, through protocol message exchange between the originating and transferring nodes, that the transfer had taken place.

With the introduction of the QSIG Call Transfer Notification feature, whenever a call transfer occurs over a QSIG link, or over a QSIG/MCDN or QSIG/DPNSS1 gateway, a notification of the transfer is sent to the originating and transferred-to party. The information, conveyed in call transfer messages, includes the redirection and originating number, redirection and originating name, and the redirection reason (transfer). This information is displayed on the phones of originating and terminating party. It is also captured in Call Detail Recording records.

The method of Operation Coding for the Call Transfer Notification feature is configured by Object Identifier, used for European Telecommunication Standard Institute (ETSI) interfaces, or by Integer Value, used for International Standards Organization (ISO) interfaces. The Operation Coding method is configured as a Remote Capability using the RCAP prompt in LD 16 and LD 17).

QSIG Call Transfer Notification at work

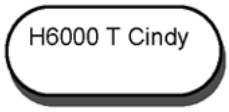
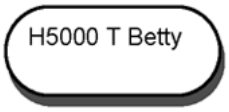
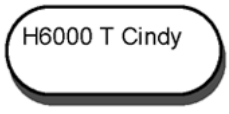
Consider the following application of the QSIG Call Transfer Notification feature being applied to a call within a QSIG system network (the network signaling occurs over QSIG links).

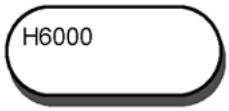
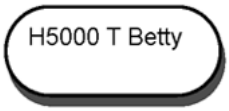
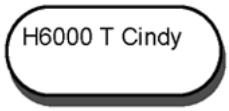
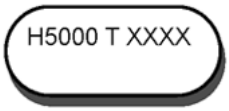
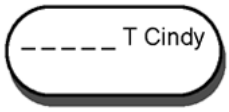
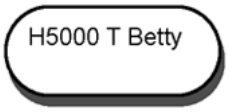
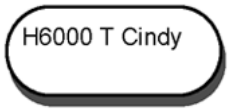
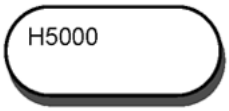
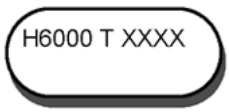
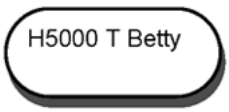
Station A (Betty, DN 5000) calls Station B (Anne, DN 4000), over a QSIG link. Station B transfers to Station C (Cindy, DN 6000), also over a QSIG link.Table 172: Originating phone and transferred-to phone display possibilities on page 378 summarizes the display options pertaining to the originating phone and transferred-to phone.

Note:

The underlined Class of Service values are in effect.

Table 172: Originating phone and transferred-to phone display possibilities

Originating phone Class of Service	Originating phone display Originating phone display after receipt of transfer notification from transferred-to phone	Transferred-to phone Class of Service	Transferred-to phone display Transferred-to phone display after receipt of transfer notification from originating phone
CNDA NAMA DDGA		CNDA NAMA DDGA	
CNDA NAMA DDGD		CNDA NAMA DDGA	

CNDD NAMA DDGA		CNDA NAMA DDGA	
CNDA NAMD DDGA		CNDA NAMA DDGA	
CNDA NAMA DDGA		CNDA NAMA DDGD	
CNDA NAMA DDGA		CNDD NAMA DDGA	
CNDA NAMA DDGA		CNDA NAMD DDGA	

Note:

No reason is displayed if the Class Of Service CNDD is configured on the phone.

Note:

When presentation of the remote party number is not allowed (Class of Service DDGD on the remote party), the other party displays the following instead of the DN:

- The trunk access code and the route member, if a trunk is involved, that is, transfer to an other node; or
- Several dashes, if this party is on the transferring node, that is, local transfer to a node.

Operating parameters

The Call Transfer by rerouting method is not implemented.

The subAdressTransfer invoke APDU is not generated by the system.

The maximum allowed size of the name information is 50 characters for QSIG. However, a maximum of 27 characters is allowed with the Calling Party Name Display (CPND) feature. Therefore, name display information might be truncated.

The call transfer notification is sent whatever the numbering plan is. However, the redirection number that is sent has any meaning only if either the Coordinated Dialing Plan (CDP) or Uniform Dialing Plan (UDP) is used to build the redirection number.

Digit manipulation can not be applied to the redirection number sent in the Call Transfer messages.

The Call Transfer Notification for a call extended by an attendant in an MCDN network is provided by the NAS feature. The Call Transfer QSIG/MCDN gateway does not map this NAS information into QSIG messages.

Feature interactions

Calling Line Identification Presentation/Restriction

CLIP/CLIR applies to the originating and to the transferred-to phone display.

Connected Line Identification Presentation/Restriction

COLP/COLR applies to the originating and to the transferred-to phone display.

Calling Name Identification Presentation/Restriction (CNIP/CNIR)

The Calling, Alerting or Connected name are not included in the ctComplete invoke APDU, at the transferring PINX.

Connected Name Identification Presentation/Restriction (CONP/CONR)

If a name, either from QSIG Name Display Service or QSIG Call Transfer Notification, is received, it is displayed, if no presentation option forbids it.

On-Hold

The interaction with On-Hold is handled with the same manner as between MCDN Call Transfer Display and On-Hold.

Symposium Call Center Server

If a QSIG call is transferred to a Symposium Call Center DN, the call type is updated in the ICC message for QSIG Transfer.

QSIG Name Display Supplementary Services (QNDS)

Calling, Alerting and Connected Name

The Calling, Alerting or Connected name is not included in the ctComplete invoke APDU, at the transferring PINX.

Connected Name

If a name, either from QSIG Name Display Service and QSIG Call Transfer Notification, is received, it will be displayed only if no presentation option forbids it.

QSIG Diversion

Call Forwarding Unconditional/Call Forwarding Busy

When the call transfer is completed, the diverted-to phone displays the Call Transfer primary user DN and the transfer notification reason.

Call Forwarding No Reply

Call Transfer interacts with Call Forward No Reply (CFNR), if CFNR is invoked for an unanswered, transferred call.

Either the Transferring PINX acts as Rerouting PINX (in case of call transfer by join and call forwarding by rerouting, or the Secondary PINX acts as Rerouting PINX (call forwarding by forward switching.

QSIG Path Replacement (QPR)

QSIG Call Transfer as trigger for QPR

QSIG Call Transfer can be configured as a trigger for QSIG Path Replacement. If configured, after the completion of the transfer, QSIG Path Replacement starts to obtain a more efficient connection, if necessary.

Feature packaging

This feature requires the following packages:

- Integrated Services Digital Network (ISDN) package 145
- Primary Rate Access (PRI) package 146
- 2.0 Mbit Primary Rate Access (PRI2) package 154
- International Primary Rate Access (IPRA) package 202
- Multi-purpose Serial Data Link (MSDL) package 222
- Q-reference Signaling Point Interface Supplementary Services (QSIG-SS) package 263
- QSIG Generic Functional Protocol Transport (QSIG-GF) package 305
- QSIG Supplementary Services (QSIG-SS) package 316

For ISDN Basic Rate Interface Trunking (BRIT), the following packages are required:

- ISDN Basic rate Access (BRI) package 216
- ISDN Basic Rate Trunk Access (BRIT) package 233

For a QSIG/DPNSS1 Gateway, the following packages are required:

- Integrated Digital Access (IDA) package 122
- Digital Private Networking Signaling System No. 1 (DPNSS) package 123
- DPNSS Network Services (DNWK) package 231

For notification display, the following packages are required:

- Digit Display (DDSP) package 19
- Call Party Name Display (CPND) package 95

Feature implementation

Task summary list

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

1. Table 173: LD 15 - Configure a Private Integrated Services Network Exchange (PINX) DN for a customer. on page 383
2. Table 174: LD 17 - Configure the method of Operation Coding for the QSIG Call Transfer Notification feature for the associated QSIG D-Channel. on page 384

   The value is configured as a remote capability. It can be either by Object Identifier, used for ETSI, or by Integer Value, used for ISO.
3. Table 175: LD 95 - Configure the Reason for Redirection, and the Call Transfer mnemonic, for a customer. on page 384
4. Table 176: LD 95 - Define a name string for a customer. on page 385
5. Table 177: LD 10 - Configure an analog (500/2500-type) transferring phone. on page 385

   Use LD 10, so that after Call Transfer, the originating phone display and the transferred-to phone display show the name and number associated with the transferred-to phone and originating phone.
6. Table 178: LD 11 - Configure a Meridian 1 proprietary transferring phone. on page 386

   Use LD 11, so that after Call Transfer, the originating phone display and the transferred-to phone display show the name and number associated with the transferred-to phone and originating phone.
7. Table 179: LD 11 - Configure a Meridian 1 Proprietary Phone. on page 386

   Use LD 11, so that if it originates or receives a transferred call over a QSIG link, it displays the redirected/connected name and number after it is received.
8. Table 180: LD 16 - Configure the method of Operation Coding for the QSIG Call Transfer Notification for the associated QSIG ISDN BRI trunk route. on page 387

This section contains the procedures required to configure the QSIG Supplementary Services - Name Display Enhancement feature.

ISDN PRI implementation

Table 173: LD 15 - Configure a Private Integrated Services Network Exchange (PINX) DN for a customer.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	NET	Networking data.
...		
CUST	0-99	Customer number as defined in LD 15.

Prompt	Response	Description
- PINX_DN	xx...x	Private Integrated Services Network Exchange DN (1-7 digits).

Table 174: LD 17 - Configure the method of Operation Coding for the QSIG Call Transfer Notification feature for the associated QSIG D-Channel.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	ADAN	Action Device and Number.
- ADAN	NEW DCH xx	Add a D-Channel on port 0-63, for Large Systems.
- IFC	ESGF ISGF	ETSI QSIG interface with GF platform. ISO QSIG interface with GF platform.
- RCAP		Enter the Operation Coding method for the QSIG Call Transfer Notification feature.
	CTI XCTI CTO XCTO	Call Transfer Operation Coding method is by Integer Value. Remove the Call Transfer Coding by Integer Value. Call Transfer Operation Coding method is by Object ID. Remove the Call Transfer Coding by Object ID. **Note:** CTO and CTI are mutually exclusive.

Table 175: LD 95 - Configure the Reason for Redirection, and the Call Transfer mnemonic, for a customer.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	CPND	Configure the Calling Party Name Display data block.
CUST	0-99	Customer number as defined in LD 15.
RESN	(NO) YES	(Do not) display the reason for redirection. Enter YES.
- XFER	(T) aaaa	1-4 character mnemonic for Call Transfer. The default is T.

Table 176: LD 95 - Define a name string for a customer.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	NAME	Create a new Calling Party Name string.
CUST	nn	Customer number, as previously entered in LD 95 when configuring the CPND data block. For Large System and CS 1000E system.
DN	xxx...x	Directory Number (1 to 7 digits) to which the name string is associated.
NAME	a...a	CPND name, in ASCII characters.

Table 177: LD 10 - Configure an analog (500/2500-type) transferring phone.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	aaa	Type of analog phone aaa = 500 or 2500.
TN	l s c u.	Terminal Number for Large System and CS 1000E.
CUST	0-99	Customer number as defined in LD 15.
DN	xxxx yyy	Directory Number and CLID entry (range for yyy is 0 - value entered in the SIZE prompt in LD 15).
- CPND	NEW CHG OUT	Add new Calling Party Name Display data. Change existing Calling Party Name Display data. Remove Calling Party Name Display data.
- - CPND_LAN	aaa	Calling Party Name Display language.
- - NAME	aaa bbbb	Calling Party Name Display name. aaaa = first name, bbbb = last name.
...		
CLS		Class of Service.
	(XFD) XFA	Transfer (denied) allowed on this phone.
	(CNDD) CNDA	Name Display (denied) allowed on this phone.
	(NAMD) NAMA	Name Display (denied) allowed on other phones.
	(DDGD) DDGA	DN display (denied) allowed on other phones.
CLS	<CR>	The CLS prompt is generated until <cr> is entered.

Table 178: LD 11 - Configure a Meridian 1 proprietary transferring phone.

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	l s c u	Terminal Number for Large System and CS 1000E.
CUST	0-99	Customer number as defined in LD 15.
DN	xxxx yyy	Directory Number and CLID entry (range for yyy is 0 - value entered in the SIZE prompt in LD 15).
- CPND	NEW CHG OUT	Add new Calling Party Name Display data. Change existing Calling Party Name Display data. Remove Calling Party Name Display data.
- - CPND_LAN	aaa	Calling Party Name Display language.
- - NAME	aaaa bbbb	Calling Party Name Display name. First name. Last name.
...		
KEY	xx TRAN	Assign the Transfer key.
...		
CLS		Class of Service.
	(CNDD) CNDA	Name Display (denied) allowed on this phone.
	(NAMD) NAMA	Name Display (denied) allowed on other phones.
	(DDGD) DDGA	DN display (denied) allowed on other phones.
CLS	<CR>	The CLS prompt is generated until <cr> is entered.

Table 179: LD 11 - Configure a Meridian 1 Proprietary Phone.

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	l s c u	Terminal Number for Large System and CS 1000E.
CUST	0-99	Customer number as defined in LD 15.

Prompt	Response	Description
DN	xxxx yyy	Directory Number and CLID entry (range for yyy is 0 - value entered in the SIZE prompt in LD 15).
- CPND	NEW CHG OUT	Add new Calling Party Name Display data. Change existing Calling Party Name Display data. Remove Calling Party Name Display data.
- - CPND_LAN	aaa	Calling Party Name Display language.
- - NAME	aaaa bbbb	Calling Party Name Display name. aaaa = first name. bbbb = last name.
...		
CLS		Class of Service.
	(CNDD) CNDA	Name Display (denied) allowed on this phone.
	(NAMD) NAMA	Name Display (denied) allowed on other phones.
	(DDGD) DDGA	DN display (denied) allowed on other phones.
CLS	<CR>	The CLS prompt is generated until <cr> is entered.

ISDN BRIT implementation

Table 180: LD 16 - Configure the method of Operation Coding for the QSIG Call Transfer Notification for the associated QSIG ISDN BRI trunk route.

Prompt	Response	Description
REQ	NEW	Add new data (ISDN BRI protocol group settings).
TYPE	RDB	Route Data Block.
CUST	0-99	Customer number as defined in LD 15.
ROUT	0-511	Route number for Large System and CS 1000E system.
TKTP	aaa	Trunk type.
DTRK	YES	Digital Trunk Route.
BRIP	NO	ISDN BRI packet handler route (NO is entered, since packet data is not required).
- DGTP	BRI	Digital trunk type.
ISDN	YES	Integrated Services Digital Network.
...		

Prompt	Response	Description
- IFC	ESGF ISGF	ETSI QSIG interface with GF platform. ISO QSIG interface with GF platform.
...		
- RCAP		Enter the Operation Coding method for the QSIG Call Transfer Notification feature.
	CTI XCTI CTO XCTO	Call Transfer Operation Coding method is by Integer Value. Remove the Call Transfer Coding by Integer Value. Call Transfer Operation Coding method is by Object ID. Remove the Call Transfer Coding by Object ID. **Note:** CTO and CTI are mutually exclusive.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 30: ISDN QSIG Channel ID Coding

Contents

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

Feature description

PBXs use different kinds of timeslot mapping in ISDN QSIG networks using PRI2 links. When the system interworks with these switches, the QSIG Channel ID Coding feature allows one of two mapping options to be configured.

The QSIG Channel ID Coding feature provides an option for two different mapping schemes for timeslots 17-31 on a PRI2 loop. The mapping scheme for timeslots 1-15 is not affected.

The mapping scheme is applicable for both encoding and decoding of the Channel ID IE.

Prior to the introduction of QSIG Channel ID Coding, the channel number in the Channel ID Information Element (IE) on a PRI2 QSIG link matched the timeslot number. A summary of the previous mapping is shown in Table 181: Previous implementation of Channel IE and timeslot number on page 389.

Table 181: Previous implementation of Channel IE and timeslot number

Timeslot number	Channel number for system software	Channel number in Channel ID IE.
1-15	1-15	1-15

Timeslot number	Channel number for system software	Channel number in Channel ID IE.
17-31	16-30	17-31
16 (signaling)	31 (D-channel)	not applicable

CHID IE mapping options

With the QSIG Channel ID Coding feature, the D-channel can be configured with one of two CHID IE mapping options.

Timeslot number coded in CHID IE

With this option, the channel number of the CHID IE matches the timeslot number. Timeslots 17-31 are assigned channel numbers 17-31. This is the same as the method used prior to the introduction of the QSIG Channel ID Coding feature. A summary of the mapping performed is shown in Table 182: Timeslot number is coded in CHID IE on page 390.

Table 182: Timeslot number is coded in CHID IE

Timeslot number	Channel number for system software	Channel number in Channel ID IE.
1-15	1-15	1-15
17-31	16-30	17-31
16 (signaling)	31 (D-channel)	not applicable

Channel number coded in CHID IE

With this option, the channel number of the CHID IE matches the system software channel number. Timeslots 17-31 are assigned channel numbers 16-30. A summary of the mapping performed is shown in Table 183: Channel number in system software is coded in CHID IE on page 390.

Table 183: Channel number in system software is coded in CHID IE

Timeslot number	Channel number in system software	Channel number in Channel ID IE.
1-15	1-15	1-15
17-31	16-30	16-30
16 (signaling)	31 (D-channel)	not applicable

The option for the QSIG Channel ID Coding feature is configured in LD 17. The two choices are:

- QCHID = NO – Timeslot values 17-31 of the PRI2 loop are associated with channel number 16-30 of the Channel ID IE.
- QCHID = YES – Timeslot values 17-31 of the PRI2 loop are associated with channel number 17-31 of the Channel ID IE (same as previous operation).

Operating parameters

QSIG Channel ID Coding is supported on all 2.0 Mbit/s QSIG interfaces.

There are two choices of hardware for Large Systems:

- PRI2 card (NT8D72) and MSDL card (NT6D80)
- Dual port DTI/PRI2 card and either the Downloadable D-channel Daughterboard card (NTBK51) or the MSDL card (NT6D80)

These systems require the QPC775 (Release E) or the QPC471 Clock Controller card.

The hardware required for CS 1000E systems is:

- PRI2 card (NTBK50)
- Downloadable D-channel Daughterboard card (NTBK51)
- Clock Controller (NTAK20)

When a system connects to another system, the proper setting for QCHID (NO or YES) depends on the software equipped at both switches. The QCHID can be YES or NO, provided it has the same value on both switches.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature requires the following packages:

- ISDN Signaling (ISDN) package 145
- 2 Mbit PRI (PRI2) package 154

- International PRA (IPRA) package 202
- Multi-Purpose Serial Data Link (MSDL) package 222
- QSIG Interface (QSIG) package 263
- QSIG GF Transport (QSIGGF) package 305

Feature implementation

Table 184: LD 17 - Configure the QCHID parameter for the QSIG PRI2 interface.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	ADAN	Action Device and Number.
- ADAN	CHG DCH 0-63	Change D-channel.
...	...	
- IFC		Interface type for D-channel.
	ISIG ESIG ESGF ISGF EGF4	Interface ID for ISO QSIG Interface ID for ETSI QSIG ETSI QSIG interface with GF capability. ISO QSIG interface with GF capability. ETSI QSIG version 4 with GF capability.
...	...	
- DCHL	xxx	PRI2 loop number for D-channel.
...	...	
- RLS	xx	Release ID of the switch at the far end of the D-channel.
- QCHID	NO	Timeslot values 17-31 of the PRI2 loop are associated with channel number 16-30 of the Channel ID IE.
	(YES)	Timeslot values 17-31 of the PRI2 loop are associated with channel number 17-31 of the Channel ID IE (same as previous operation). This prompt does not appear for PRI (1.5 Mb) links.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 31: ISDN QSIG/ETSI GF Enhancement

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

The Integrated Services Digital Network ISDN (ISDN) Q-reference Signaling Point (QSIG)/ European Telecommunication Standards Institute (ETSI) Generic Functional Transport (GFT) was introduced to provide a set of generic transport mechanisms to support QSIG supplementary services over a Private or Public Integrated Services Network.

It must be noted that the ISDN QSIG/ETSI GF platform by itself does not support any features. It only provides the platform to be used by the various QSIG supplementary services.

The ISDN QSIG/ETSI Generic Functional (GF) Transport Enhancement feature has been introduced to provide improvements in call control, to allow the supported QSIG signaling protocols to work together seamlessly. These improvements comprise the following:

For QSIG GFT:

- Support of Call-Independent gateways between QSIG, and EuroISDN, Meridian Customer Defined Networking MCDN, and Digital Private Networking Signaling No. 1 (DPNSS1).

  The QSIG Cal-Independent Generic Functional Transport - Control capability has been modified, so that at these gateways, the CONNECT message of a QSIG supplementary service is not systematically sent after the receipt of a QSIG SETUP message. Instead, the control of the call processing, such as controlling the expiration of the T303 and T310 timers, is left to the QSIG supplementary service operating over the QSIG GFT.

- Call Related Application Protocol Data Unit (APDU) transport.

  Transit Private Integrated Services Network Exchange (PINX) operation:

  The Call Related APDU transport has been modified, to ensure that a call that is being optimized during QSIG Path Replacement is using the optimum path (the first available alternate route in a route list index). A transit Private Integrated Services Network Exchange (PINX) can determine whether or not to intercept a Facility Information Element (IE) containing a Network Facility Extension (NFE) EndPinx message.

  This modification also allows a Facility IE carried in a Setup message that has encountered congestion to be included in an alternate call, if QSIG Alternate Routing is activated on a transit node.

  The method of calculating the available length for an APDU in basic QSIG call control messages is now more accurate. The available length for an APDU is now calculated as the difference between the maximum message length (260 octets) and the exact length of other mandatory and non-mandatory IEs in the message.

For ETSI GFT:

- Call Related Application Protocol Data Unit (APDU) transport.

  The method of calculating the available length for an APDU in basic ETSI call control messages is now more accurate. The available length for an APDU is now calculated as the

difference between the maximum message length (260 octets) and the exact length of other mandatory and non-mandatory IEs in the message.

Operating parameters

The operating parameters that follow are carried over from the ETSI Generic Functional Transport feature.

The following operating parameters apply to the ISDN QSIG ETSI GF component of the feature enhancements:

- The ETSI GF gateways to and from other signaling systems, such as QSIG, Digital Private Networking Signaling No. 1 (DPNSS1), and Meridian Customer Defined Networking (MCDN), are only supported for Call Independent gateways.
- The following ETSI GF functionalities are not supported:
  - the control of supplementary services using the separate message approach (Hold/ Retrieve messages)
  - the bearer-related broadcast transport mechanism (multipoint configuration)
  - the bearer-independent point-to-point connections transport mechanism
  - the bearer-independent broadcast transport mechanism
  - the generic notification procedures
  - the network-side channel reservation function
  - the generic procedures for supplementary services management
  - the generic status request procedure
  - the support of the Extended Facility Information Element (FIE)
- During call processing, the translation of a DN address requires that the DN be associated with a customer number. For the processing of an ETSI basic call, this customer number association is found through the B-Channel identified in the Channel Identification Information Element.

  For the DN address translation that is not associated with a basic call, the customer number/DN association has to be determined using other methods. For an ISDN BRI trunk Digital Subscriber Loop (DSL) interface, the customer number is associated with a D-Channel. For an ISDN PRI interface, the Private Integrated Services Network Exchange Customer (PINX_CUST) prompt in LD 17 is used to define a customer number to a D-Channel. This implies that bearer-independent messages on an ISDN PRI D-Channel are associated with a single customer, as configured in LD 17.

For example, in a multi-customer configuration, if every customer on a system wants to use the bearer-independent transport service over an ETSI PRI interface, then each customer requires the use of a separate D-Channel.

- The ISDN ETSI GF platform supports up to eight Remote Operation Service Element (ROSE) components in one message, either in a single Facility Information Element (FIE) or multiple FIEs. If a supplementary service requests that the ISDN ETSI GF Transport send a message exceeding this limit, a notification message is sent to the supplementary service.

The following operating parameters apply to the ISDN QSIG ETSI GF component of the feature enhancements:

- The QSIG GF gateways to and from other signaling systems, beyond the Q reference point, such, Digital Private Networking Signaling No. 1 (DPNSS1), EuroISDN, and Meridian Customer Defined Networking (MCDN), are only supported for Call Independent gateways.
- At a QSIG to EuroISDN, DPNSS1, or MCDN gateway, when a call-independent Setup message is received which has a valid gateway destination, and which includes several FIEs that belong to different supplementary services, then the gateway is not supported by this feature.
- The MCDN to QSIG GF "protocol converter" does not actually convert the MCDN features, it only provides the conversion platform. The individual MCDN features have to be ported to the ISDN QSIG GF separately.
- This feature does not implement or enhance any ISDN QSIG supplementary services or networking applications such as Virtual Network Services (VNS).
- Although the transport of Manufacturer Specific Information (MSI) across a ISDN QSIG interface is used by the MCDN to QSIG GF "protocol converter", a tandem PINX receiving this information can discard it. The MSI can be passed to the next node only if the transit PINX is a system.
- During call processing, the translation of a DN address requires that the DN be associated with a customer number. For the processing of a QSIG basic call, this customer number association is found through the B-Channel identified in the Channel Identification Information Element. For the DN address translation that is not associated with a basic call, the customer number/DN association has to be determined using other methods.

  For an ISDN BRI trunk Digital Subscriber Loop (DSL) interface, the customer number is associated with a D-Channel.

  The ISDN QSIG GF transport does not support the use of the MCDN proprietary Private Network Identifier (PNI) numbers for the customer association. For an ISDN PRI interface, the Private Integrated Services Network Exchange Customer (PINX_CUST) prompt in LD 17 is used to define a customer number to a D-Channel. This implies that all call independent messages on an ISDN PRI D-Channel are associated with a single customer, as configured in LD 17.

  For example, in a multi-customer configuration, if every customer on a system wants to use the call-independent transport service over a QSIG PRI interface, then each customer

requires the use of a separate D-Channel. Note that this limitation also applies to MCDN features that are converted by the MCDN to QSIG GF "protocol converter", to function over a QSIG network.

- Basic call setup and call-independent connection setup originating from a QSIG supplementary service can be rejected by the far end due to the unavailability of system resources, and due to overdo control. A notification message is sent to the supplementary service, so that it can initiate a retry.
- The call-independent connection-oriented APDU transport service requires that a transit node PINX retain any information pertaining to a call-independent connection, for the duration of the connection. This information can be required by a particular application to properly route subsequent messages associated with the same call-independent connection.
- This requirement at the transit PINX implies that system resources, such as call registers, are used by the call-independent transport services at the tandem nodes. Therefore, applications that make use of call-independent transport services can be limited by the amount of available system resources allocated at the transit PINX.
- The Call Independent APDU transport can be supported by two different transport mechanisms, connection-oriented, or without connection. The later method is not supported on the ISDN QSIG GF platform.
- The Association Control Service Element (ACSE) and Dialogue Service Element (DSE) for application association is not supported by the ISDN QSIG GF platform on a system. ACSE and DSE protocol data units that are received by the ISDN GF platform facility at a destination Private Integrated Services Network Exchange (PINX) will be discarded. A transit INX will pass on the ACSE and DSE data units transparently to the next node.
- The ISDN QSIG GF platform supports up to eight Application Protocol Data Units (APDUs), that is, Remote Operation Service Element (ROSE) components, in one message, either in a single Facility Information Element (FIE) or multiple FIEs. If a supplementary service requests that the ISDN QSIG GF Transport send a message exceeding this limit, a notification message is sent to the supplementary service.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

This feature requires the following packages:

- Integrated Services Digital Network (ISDN) package 145
- Primary Rate Access (PRA) package 146 for 1.5 Mbit signaling
- International Primary Rate Access (IPRA) package 202
- 2.0 Mbit Primary Rate Access (PRI2) package 154 for 2.0 Mbit signaling
- Multi-purpose Serial Data Link (MSDL) package 222
- Q-reference Signaling Point Interface (QSIG) package 263
- QSIG Generic Functional Protocol Transport (QSIG-GF) package 305

For ISDN Signaling Link (ISL) QSIG, the following additional packages are required:

- ISDN Signaling Link (ISL) package 147

For ISDN Basic Rate Interface Trunking (BRIT), the following additional packages are required:

- ISDN Basic rate Access (BRI) package 216
- ISDN Basic Rate Trunk Access (BRIT) package 233

Feature implementation

There are no specific implementation procedures for this feature.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 32: ISDN QSIG/EuroISDN Call Completion

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

The ISDN QSIG/EuroISDN Call Completion feature expands Ring Again functionalities on QSIG and EuroISDN interfaces. This feature provides Call Completion to Busy Subscriber (CCBS) and Call Completion on No Response (CCNR) supplementary services.

The Call Completion to Busy Subscriber (CCBS) supplementary service allows the calling party to apply a Ring Again request when encountering a busy Directory Number (DN). The system

alerts the calling party when occupied DN is available to receive a call. The calling party has the option of completing this call without making a new call attempt. CCBS is supported on QSIG and EuroISDN signalling protocols.

Call Completion on No Response (CCNR) supplementary service allows the calling party to apply a Ring Again request to an unanswered DN. With this service, the system alerts the calling party when the dialed DN becomes idle after a period of activity. The calling party has the option of completing the call without making a new call attempt. CCNR is only supported on QSIG signaling protocols.

Operating parameters

ISDN QSIG/EuroISDN Call Completion supports Uniform Dialing, Customer Dialing and Group Dialing plans.

Since the Ring Again feature is configured on phone level basis, either by Class of Service or Ring Again key, a phone cannot be programmed with CCBS on EuroISDN denied and Meridian Customer Defined Network (MCDN) Network Ring Again allowed. Ring Again is allowed or denied locally or network wide in the same manner.

ISDN QSIG/EuroISDN is supported in ETSI compliant countries.

This feature uses QSIG GF and ETSI GF platforms. With the development of the QSIG Generic Transport (GF) feature, the Call Completion on No Response (CCNR) and Call Completion to Busy Subscriber service on QSIG cannot be involved in any gateway.

Call Completion to Busy Subscriber and Call Completion on No Response are not supported on ISDN BRI phones.

Gateways between EuroISDN CCBS, MCDN Network Ring Again on Busy and DPNSS1 Call Back When Free are supported.

A Private Integrated Services Network Exchange (PINX) DN must be configured in the Customer Data Block (LD 15). The PINX DN is used for routing free notification on the MCDN or DPNSS1 if the EuroISDN network does not provide a calling number for the service. The DN configured should be consistent with the type of number plan used. PINX DN is also used for incoming Direct Inward Dialing (DID) calls in the same manner as Listed Directory Number 0 (LDN0) is used for a basic call.

Existing limitations, applicable to standalone or network wide Ring Again operation, apply to CCNR and CCBS on QSIG and CCBS on QSIG and EuroISDN.

Feature interactions

Access Restrictions

Call Restriction

Trunk Group Access Restrictions

ISDN QSIG/EuroISDN Call Completion does not override Access, Call Restriction or Trunk Group Access restrictions. When Call Completion is activated, the second call has the same restrictions as the initial call that received either no answer or a busy indication.

Advice of Charge for EuroISDN

Advice of Charge applies to the initial and call completion (second call). The initial call receives the same charging information as a normal busy call. The call completion receives the same charging information as the first call.

Attendant Consoles

Call Completion requests cannot be directed to or from an Attendant Console.

Automatic Call Distribution

Call Completion requests cannot be directed to or from Automatic Call Distribution (ACD) Directory Numbers (DNs).

An ACD phone that uses a normal DN key (not the ACD key) can activate the Ring Again key when encountering a busy or no answer situation.

Call Detail Recording

A Call Detail Recording (CDR) is not produced for Call Completion signaling. The second call receives a CDR as a normal call.

Call Forward All Calls

When the Call Forward feature is activated on a local basis and an incoming Call Completion request is received, the Call Completion request is registered against the forwarded DN.

Call Transfer

Call Completion notification is only presented to the Call Completion originating phone. This notification cannot be transferred to another station. Once the second call is completed, the call can be transferred.

If a user encounters a busy or no answer situation during a transfer operation, Call Completion can be activated.

Call Waiting

On an analog (500/2500-type) phone, Call Completion notification waits until the phone has finished an active call. If Call Waiting is configured on a phone, notification is presented after the Call Waiting call. If an additional call is queued while Ring Again free notification is waiting on a phone, the waiting call takes precedence over the Call Completion notification. An established Call Completion call is also queued if a phone has Call Waiting feature equipped and is occupied on another call.

Conference

A Call Completion request cannot be made on a conference call attempt.

Direct Inward System Access

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

Do Not Disturb

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

EuroISDN Trunk - Network Side

Call Completion to Busy Subscriber is supported on a EuroISDN Trunk - Network Side connectivity interface.

Call Completion on No Reply is supported on QSIG and DPNSS1 (as Call Back When Next Used) interfaces, corresponding to the MCDN Network Ring Again on No Answer feature. It is not supported on a EuroISDN Trunk - Network Side connectivity interface.

Flexible Feature Codes

Analog (500/2500-type) phone can use Flexible Feature Codes (FFCs) to activate Call Completion to Busy Subscriber requests.

Group Call

Call Completion cannot be applied to a Group Call.

Group Hunt

Call Completion to Busy Subscriber cannot be applied to Pilot DN when no idle phone is located during a Group Hunt call.

Hot Line

Call Completion cannot be used in conjunction with the Hot Line feature.

Maintenance Busy

Call Completion on Busy Subscriber is not accepted against a phone in Maintenance Busy state.

Make Set Busy

Phones that have Make Set Busy (MSB) activated can request Call Completion to another DN, as the free notification overrides the MSB feature. Incoming Call Completion to Busy Subscriber (CCBS) requests do not override the MSB feature. A phone is considered busy while MSB is active. A CCBS request is registered against a busy phone, but only advances when the MSB feature is deactivated and the phone remains free.

Permanent Hold

Analog (500/2500 type) phones with Permanent Hold cannot use the Ring Again functionalities.

Ring Again

Ring Again No Answer

Analog (500/2500 type) phones can have only one Call Completion to Busy Subscriber request at a given time. Meridian 1 Proprietary Phones can make Ring Again requests based on the number of Ring Again keys programmed on a phone.

Feature packaging

ISDN QSIG/EuroISDN Call Completion (QSIG-SS) is package 316. The package dependencies for QSIG-SS are:

- Optional Features (OPTF) package 1, and
- ISDN QSIG GF Transport (QSIGGF) package 305

Depending on the application, the following packages are also required.

For the QSIG ISDN PRI interface:

- Digit Display (DDSP) package 19
- Coordinated Dialing Plan (CDP) package 59
- Integrated Services Digital Network (ISDN) package 145
- Primary Rate Interface (PRI) package 146, or
- 2.0 Mbit Primary Rate Access (PRI2) package 154
- Flexible Numbering Plan (FNP) package 160
- Overlap Signaling (OVLP) package 184 (optional)
- International Primary Rate Access (IPRA) package 202
- Multipurpose Serial Data Link (MSDL) package 222
- QSIG Interface (QSIG) package 263.

For the QSIG ISDN BRI interface:

- Digit Display (DDSP) package 19
- Integrated Services Digital Network (ISDN) package 145
- Overlap Signaling (OVLP) package 184 (optional)
- Basic Rate Interface (BRI) package 216
- ISDN BRI Trunk Application (BRIT) package 233
- QSIG Interface (QSIG) package 263

The EuroISDN Call Completion Supplementary Service (ETSI-SS) is package 323.

For the EuroISDN ISDN PRI interface:

- Digit Display (DDSP) package 19
- Coordinated Dialing Plan (CDP) package 59
- Integrated Services Digital Network (ISDN) package 145
- 2.0 Mbit Primary Rate Access (PRI2) package 154
- Flexible Numbering Plan (FNP) package 160
- ISDN Supplementary Features (ISDN INTL SUPP) package 161
- Overlap Signaling (OVLP) package 184 (optional)
- International Primary Rate Access (IPRA) package 202
- Multipurpose Serial Data Link (MSDL) package 222
- EuroISDN (EURO) package 261
- Calling Party Privacy (CPP) package 301

For the EuroISDN ISDN BRI interface:

- Digit Display (DDSP) package 19
- Integrated Services Digital Network (ISDN) package 145
- ISDN Supplementary Features (ISDN INTL SUPP) package 161
- Overlap Signaling (OVLP) package 184 (optional)
- Basic Rate Interface (BRI) package 216
- ISDN BRI Trunk Application (BRIT) package 233
- EuroISDN (EURO) package 261

Feature implementation

Task summary list

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

1. Table 185: LD 17 - Configure Remote Capabilities to Call Completion for QSIG or EuroISDN on Primary Rate Interface. on page 406
2. Table 186: LD 16 - Configure Remote Capability to Call Completion QSIG or EuroISDN on Basic Rate Interface. on page 407
3. Table 187: LD 10 - Add or Change Ring Again on analog (500/2500-type) phones. on page 407
4. Table 188: LD 11 - Add or Change Ring Again on Meridian 1 proprietary phones. on page 408
5. Table 189: LD 57 - Define Ring Again Flexible Feature Codes (FFCs) and FFC Confirmation Tone. on page 408
6. Table 190: LD 15 - Configure a Private Integrated Services Network Exchange (PINX) DN. on page 408

Table 185: LD 17 - Configure Remote Capabilities to Call Completion for QSIG or EuroISDN on Primary Rate Interface.

Prompt	Response	Description
REQ	CHG	Change
TYPE	ADAN	Configuration Record
- ADAN	CHG DCH x	Change D-channel
...		
- IFC	xxxx	Interface type for route where xxxx is: ESGF = Interface ID for ETSI QSIG with GF capability. ISGF = Interface ID for ISO QSIG with GF capability. EURO = Interface ID for EuroISDN.
- - CNTY	xxx	Country. Prompted only when IFC = EURO. Where xxx is: ETSI = ETS 300 102 basic protocol (default). GER = Germany. DEN = Denmark. NET = Master Mode. SWI = Switzerland.

Prompt	Response	Description
...		
- RCAP	CCBS	Add Call Completion to Busy Subscriber as remote capability.
- RCAP	CCNR	Add Call Completion No Response. This prompt is only applicable to QSIG interfaces.

Table 186: LD 16 - Configure Remote Capability to Call Completion QSIG or EuroISDN on Basic Rate Interface.

Prompt	Response	Description
REQ	CHG	Change
TYPE	RDB	Route Data Block
...		
DTRK	YES	Digital Trunk Route.
- DGTP	BRI	Basic Rate Interface. Allowed if TKTP = TIE, COT, or DID.
...		
-IFC	xxxx	Interface type for route where xxxx is: ESGF = Interface ID for ETSI QSIG with GF capability. ISGF = Interface ID for ISO QSIG with GF capability. EURO = Interface ID for EuroISDN.
- - CNTY	xxx	Country. Prompted only when IFC = EURO. Where xxx is: ETSI = ETS 300 102 basic protocol (default). GER = Germany. DEN = Denmark. NET = Master Mode. SWI = Switzerland.
...		
- RCAP	CCBS	Add Call Completion to Busy Subscriber as a remote capability. The response XCCBS removes this capability.
- RCAP	CCNR	Add Call Completion No Response as a remote capability. This prompt is only applicable to QSIG interfaces.

Table 187: LD 10 - Add or Change Ring Again on analog (500/2500-type) phones.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	500	Type of phone.

Prompt	Response	Description
TN	l s c u	Terminal Number for Large System and CS 1000E.
CLS	XRA	Ring Again allowed. XRD = Ring Again Denied (default).

Table 188: LD 11 - Add or Change Ring Again on Meridian 1 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	l s c u	Terminal Number for Large System and CS 1000E.
KEY	xx RGA	Assign Ring Again key where xx = key number. On M2317 phones, xx = 27.

Table 189: LD 57 - Define Ring Again Flexible Feature Codes (FFCs) and FFC Confirmation Tone.

Prompt	Response	Description
REQ	CHG	Change
TYPE	FFC	Flexible Feature Code.
CUST	0-99	Customer number as defined in LD 15.
FFCT	YES	Confirmation tone provided after FFC is activated. NO = Confirmation tone not provided.
CODE	RGAA	Ring Again activate.
RGAA	xx	Ring Again code.
CODE	RGAD	Ring Again deactivate.
RGAD	xx	Ring Again deactivation code.
CODE	RGAV	Ring Again verification.
RGAV	xx	Ring Again verification code.

Table 190: LD 15 - Configure a Private Integrated Services Network Exchange (PINX) DN.

Prompt	Response	Description
REQ	CHG	Change
TYPE	NET	Networking
CUST	0-99	Customer number as defined in LD 15.

Prompt	Response	Description
- PINX_DN	xx...x	Node DN to a maximum of seven digits with DN extension.

Feature operation

Call Completion on Busy Subscriber

Meridian 1 Proprietary Phones

1. First, Phone A calls Phone B on an outgoing QSIG/EuroISDN route and receives a busy indication.
2. Before releasing this call, Phone A presses the Ring Again key (RGA) to activate Ring Again functionality. The DN lamp flashes.
3. When the far end receives a Call Completion on Busy Subscriber (CCBS) request, Phone A's RGA lamp key is steadily lit, and the DN lamp darkens.
4. Then Phone A awaits notification that Phone B is free. Phone A can make or receive calls while waiting for notification.
5. When Phone B goes on-hook, Phone A is notified in the form of a one second burst of ringing and the RGA key lamp fast flashes.
6. The recall is accepted by selecting the DN key and pressing the RGA key. Phone A has thirty seconds to accept the recall before it is cancelled.
7. If Phone A accepts the recall and Phone B is free, the call is rung. However, if Phone B has either received or originated another call, Phone A receives a busy indication. Phone A has the option of requesting CCBS again.

Note:

If Phone A presses the RGA key before receiving notification then the request is cancelled.

Analog (500/2500-type) phones without RGA key

1. First, Phone A calls Phone B on an outgoing QSIG/EuroISDN route and receives a busy indication.
2. Then Phone A performs a switch-hook flash and a special dial tone is heard.
3. Next, Phone A dials RGA Activate Flexible Feature Code (FFC) or SPRE +1 to activate CCBS feature. This request cancels and replaces previous RGA Activate requests.
4. If an FFC is dialed, an FFC confirmation tone if programmed indicates that the CCBS request is registered.
5. Finally, Phone A goes on-hook and is free to make or receive other calls.
6. When Phone B goes on-hook, Phone A receives a burst of tone and has thirty seconds to accept the recall before it is cancelled.
7. If Phone B has originated or received another call, Phone A receives a busy indication and has the option of requesting the CCBS again.

Note:

If Phone A presses the RGA key before receiving notification then the request is cancelled.

Call Completion No Response

Meridian 1 Proprietary Phones

- First, Phone A calls Phone B on an outgoing QSIG route. The call rings but is not answered. Phone A receives a ringback tone.
- Then Phone A presses RGA key to activate Ring Again on No Answer prior to releasing the call. The DN key lamp darkens.
- Next, Phone A goes on-hook and is free to make or receive calls.
- When Phone B makes a call and goes on-hook. Phone A is notified by a one second burst of tone and the RGA lamp fast flashes.
- Finally, Phone A can accept the call by selecting a DN key and pressing the RGA key. Phone A has thirty seconds to accept this call.
- If Phone A accepts the call and Phone B is free, the call goes through. However, if Phone B receives or originates another call, Phone A receives a busy indication. Phone A has the option of requesting CCBS.

Note:

If Phone A presses the RGA key before receiving notification then the request is cancelled.

Chapter 33: ISDN QSIG/EuroISDN Call Completion Enhancement

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

The ISDN QSIG/EuroISDN Call Completion was introduced, to provide Call Completion to a Busy Subscriber (CCBS) over QSIG and EuroISDN networks, and Call Completion on No Response (CCNR) over QSIG networks.

- Call Completion to a Busy Subscriber (CCBS). This service allows a calling party to apply a Ring Again request to a called party that is busy. When the busy party becomes free,

the calling party receives notification and can complete the call without making a new call attempt.

- Call Completion on No Response (CCNR). This service allows a calling party to apply a Ring Again request to a called party that is idle, and does not answer. The calling party receives notification after the system notices subsequent activity on the called extension, that has again become idle.

With enhancement, the QSIG/EuroISDN Call Completion Enhancement feature introduces the following functionalities:

- Allows the method of Operation Coding for the CCBS and CCNR Supplementary Services to be configured by Object Identifier, used for European Telecommunication Standard Institute (ETSI) interfaces, or by Integer Value, used for International Standards Organization (ISO) interfaces. The Operation Coding method is configured as a remote capability using the RCAP prompt for an ISDN PRI D-Channel for ISDN PRI, or for an ISDN BRI route for ISDN BRI trunking.
- Supports all ETSI/QSIG to MCDN and DPNSS1 gateways.
- Allows all EuroISDN countries to fully support the QSIG/EuroISDN Call Completion services.

Operating parameters

Call Completion to a Busy Subscriber (CCBS) is available only on EuroISDN interfaces supporting the ETSI QSIG Generic Functional Protocol Transport.

Call Completion to a Busy Subscriber (CCBS) is not supported on ISDN BRI phones.

The QSIG/EuroISDN Call Completion Enhancement feature cannot use route access codes. The feature can only use a Coordinated Dialing Plan or Uniform Dialing Plan. Also, this feature does not support digit manipulation tables.

Since Ring Again is configured for each phone, using Class of Service for analog (500/2500-type) phones or defining a Ring Again key on Meridian Proprietary Phones, it is not possible to configure a phone so that Call Completion to a Busy Subscriber is denied for EuroISDN interfaces, and MCDN Network Ring Again is allowed.

Call completion on QSIG uses two signaling types, Connection Retention, where the signaling connection remains established until call completion is achieved, or the call completion attempt is cancelled, and Connection Release, where the signaling connection is cleared after each phase of call signaling, and a new signaling connection is established for each subsequent phase of call independent signaling. At the terminating side of the call connection, both signaling methods are mandatory, and are supported by the system. At the originating side, although both methods are possible, the system supports only the Connection Retention type, since it matches the EuroISDN signaling.

A call completion recall over a QSIG interface uses two methods of establishing the recall, Path Reservation, where a bearer connection between the originating and terminating sides

is established before presenting the recall to the originating party, in order to avoid network congestion, and Non-path Reservation, where a bearer connection between the originating and terminating sides is established after the originating party answers the recall, and the recall attempt is cancelled if network congestion is encountered. At the terminating side of the call connection, both methods of recall are mandatory, and are supported by the system. At the originating side, although both methods are possible, the system supports only the Non-path Reservation method, since it is used by MCDN and EuroISDN. Also, the Path Reservation method is not supported over QSIG to MCDN or DPNSS1 gateways.

For a QSIG or EuroISDN call completion attempt, if the called party is found to be busy again after the originating party responds to a call completion recall, there are two methods of behavior that the system takes. With the Request Retention method the call completion request remains in place at both the originating and terminating side. The terminating side begins monitoring the terminating party. With the Service Cancellation method the call completion request is cancelled at both the originating and terminating sides. The system does not support The Request Retention method.

Busy name information sent with a DISCONNECT message is only supported for Overlap Receiving (the OVLR prompt is set to YES in LD 17).

Feature interactions

Access Restrictions/trunk Group Access Restrictions

QSIG/EuroISDN Call Completion cannot override access restrictions placed on a call. The second call due to either CCBS or CCNR is under the same access restrictions as the first call.

Advice of Charge for EuroISDN

The Advice of Charge for EuroISDN applies to both the first call and second call due to CCNR. The second call receives the same call charging information as the first call (that is, as a normal call).

Attendant Console

A CCBS or CCNR cannot originate from, or terminate to, an attendant console.

Automatic Call Distribution

A CCBS or CCNR cannot originate from, or terminate to, an ACD Directory Number.

Call Connection Restriction

QSIG/EuroISDN Call Completion cannot override call connection restrictions placed on a call. The second call due to either CCBS or CCNR is under the same call connection restrictions as the first call.

Call Detail Recording

There is no CDR information created for a CCBS attempt. A new call that is established as part of CCBS is subject to CDR, the same as a normal call.

Call Forward

When a CCBS or CCNR request is received on a phone which has been call forwarded, the CCBS or CCNR request is registered against the DN to which the phone has been forwarded.

Call Transfer

Call Completion free notification can only be presented to the originating station, and cannot be transferred to another station. However, once the call has been established, it behaves as a normal call, with respect to call transfer.

During a call transfer operation, a caller that encounters a busy or no answer situation, can still activate the call transfer.

A call cannot be completed that is being transferred to a busy destination, as part of the EuroISDN Continuation feature, or to a no answer destination.

Call Waiting

A Call Completion free notification is presented to an analog (500/2500-type) phone after an active call has been completed. If there is a call waiting on the phone, the Call Completion free notification is presented to the phone after the call waiting call has been completed. If another call waiting occurs while a Ring Again free notification is waiting on the phone, the new call waiting presentation takes precedence.

A Call Completion call can wait to be established on a phone that is busy on another call, and has the Call Waiting feature activated.

Conference

A Call Completion request cannot be made as part of a conference call attempt.

Customer Controlled Routing

A Call Completion attempt request be applied to a Customer Controlled Routing DN.

Direct Inward System Access

CCBS cannot be applied to DISA calls when the terminating party is busy. CCNR cannot be applied to DISA calls when the terminating party does not answer.

Directory Number - Multiple Appearance Directory Number Redirection Prime

MARP does not apply to a Call Completion attempt.

Do Not Disturb

Stations with the Do Not Disturb feature active can make Call Completion requests against other stations, subject to the same restrictions as if the feature were not active. The incoming Call Completion free notification overrides the Do Not Disturb state.

Flexible Numbering Plan

A Flexible Numbering Plan can be used with QSIG/EuroISDN Call Completion, as long as the numbering plan is consistent over the network.

Group Call

A Call Completion request cannot be applied to a group call.

Group Hunt

CCBS cannot be applied to a PLDN when no idle phone is found during a group hunt call.

Hot Line

A Call Completion request cannot be used in conjunction with the Hot Line or Enhanced Hot Line feature.

Initialize

If a system initialize occurs at a system, all Call Completion requests at the system are deleted. If the requesting party is not on the system, which has initialized, it is not notified that a Call Completion request has been cancelled.

Maintenance Busy

A Call Completion request cannot be made against a phone in maintenance busy state.

Make Set Busy

If a phone with Make Set Busy active makes a Call Completion request to another station, then the Call Completion free notification overrides the Make Set Buy state. Incoming CCBS requests, however, do not override the Make Set Busy state. A CCBS request can only be completed once Make Set Busy is deactivated.

Permanent Hold

Analog (500/2500-type) phones with a call on permanent hold cannot use the Ring Again feature.

Pretranslation

Pretranslation can be used in conjunction with a Call Completion request.

Ring Again/Ring Again No Answer

Analog (500/2500-type) phones can only have one Call Completion request active at a time. Meridian Proprietary Phones can make as many Call Completion requests as available keys to support the requests.

Ring Again Inhibition

Ring Again cannot be activated if a limit of 12.5% of all available call registers in the Ring Again queues is exceeded.

DPNSS1 Call Back When Free and Next Used

In a QSIG/EuroISDN to DPNSS1 gateway, a caller on one node can activate Ring Again to any other destination party at another node, using a combination of QSIG/EuroISDN and DPNSS1 links.

Electronic Switched Network

QSIG/EuroISDN Call Completion uses a Uniform Dialing Plan or Coordinated Dialing Plan Numbering Plan.

The use of the Connection Retention procedure (the relevant discussion on Operating parameters on page 414, as part of the Operating parameters section) on QSIG does not require that the Numbering Plan be symmetrical (although this approach is not recommended). The called destination, which is being monitored as part of CCBS or CCNR, does not require notification on how to connect to the originating party.

This only applies if the Connection Retention procedure is used. If the Connection Release procedure is used, the numbering plan has to be symmetrical, so that the destination party can use the CLID of the originating party, to send the Call Connection free notification.

Hunt and Group Hunt

Hunt and Group Hunt is not supported by the QSIG/EuroISDN Call Completion feature.

ISDN Basic Rate Interface

QSIG/EuroISDN Call Completion is not supported on ISDN BRI phones. CCBS and CCNR cannot neither be activated or received on a ISDN BRI phone.

Network Call Redirection - EuroISDN

If a call over a EuroISDN interface has not been locally call forwarded, then a Call Completion request is ignored.

Network Call Redirection - QSIG Diversion

Call Completion to Busy Subscriber

If CCBS is invoked on a busy destination that has been diverted over a QSIG interface, the originating Private Integrated Services Network Exchange (PINX) stores the diverted-to number issued by the served PINX, to use it for CCBS call treatment. This diverted-to number is used as received, without any digit manipulation.

Call Forward Unconditional and Call Forward Busy take precedence over CCBS.

Call Completion on No Reply

If CCNR is invoked on a destination that does not answer, and that has been diverted over a QSIG interface, the originating Private Integrated Services Network Exchange (PINX) stores the diverted-to number issued by the served PINX, to use it for CCNR call treatment. This diverted-to number is used as received, without any digit manipulation.

Call Forward No Reply takes precedence over CCNR.

Feature packaging

The following packages are required for the QSIG component of the QSIG/EuroISDN Call Completion Enhancement feature:

For ISDN PRI:

- Ring Again (RGA) package 1
- Digit Display (DDSP) package 19
- Integrated Services Digital Network (ISDN) package 145
- Primary rate Access (PRI) package 146
- 2.0 Mbit Primary Rate Access (PRI2) package 154
- Flexible Numbering Plan (FNP) package 160
- Overlap Signaling (OVLP) package 184 (optional)
- International Primary Rate Access (IPRA) package 202
- Multi-purpose Serial Data Link (MSDL) package 222
- Q-reference Signaling Point Interface (QSIG) package 263
- QSIG Generic Functional Protocol Transport (QSIG-GF) package 305
- QSIG Supplementary Services (QSIG-SS) package 316
- Coordinated Dialing Plan (CDP) package 59, and one of the following:
    - Network Alternate Route Selection (NARS) package 58

- Basic Alternate Route Selection (BARS) package 57

For ISDN BRI trunking:

- Digit Display (DDSP) package 19
- Integrated Services Digital Network (ISDN) package 145
- Overlap Signaling (OVLP) package 184 (optional)
- ISDN Basic rate Access (BRI) package 216
- ISDN Basic Rate Trunk Access (BRIT) package 233
- Q-reference Signaling Point Interface (QSIG) package 263

The following packages are required for the EuroISDN component of the QSIG/EuroISDN Call Completion Enhancement feature:

For ISDN PRI2:

- Ring Again (RGA) package 1
- Digit Display (DDSP) package 19
- Coordinated Dialing Plan (CDP) package 59 or Network Alternate Route Selection (NARS) package 58, or Basic Alternate Route Selection (BARS) package 57
- Integrated Services Digital Network (ISDN) package 145
- 2.0 Mbit Primary Rate Access (PRI2) package 154
- Network Attendant Service (NAS) package 159 (for MCDN to EuroISDN gateways)
- Flexible Numbering Plan (FNP) package 160
- ISDN International Supplementary Features (ISDN SUPP) package 161
- Overlap Signaling (OVLP) package 184 (optional)
- International Primary Rate Access (IPRA) package 202
- Multi-purpose Serial Data Link (MSDL) package 222
- ETSI Supplementary Services (ETSI-SS) package 323

For ISDN BRI trunking:

- Digit Display (DDSP) package 19
- Integrated Services Digital Network (ISDN) package 145
- ISDN International Supplementary Features (ISDN SUPP) package 161
- Overlap Signaling (OVLP) package 184 (optional)
- International Primary Rate Access (IPRA) package 202
- ISDN Basic rate Access (BRI) package 216

- ISDN Basic Rate Trunk Access (BRIT) package 233
- ETSI Supplementary Services (ETSI-SS) package 323

Feature implementation

Task summary list

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

1. Table 191: LD 17 - Configure the method of Operation Coding for the Call Completion to a Busy Subscriber supplementary service for the associated QSIG or EuroISDN D-Channel. on page 423
2. Table 192: LD 17 - Configure the method of Operation Coding for the Call Completion on No Reply supplementary service for the associated QSIG D-Channel. on page 425
3. Table 193: LD 10 - Configure Ring Again Class of Service on analog (500/2500-type) phones. on page 425
4. Table 194: LD 15 - Configure a Special Prefix Number (SPRE) for a customer. on page 425
5. Table 195: LD 15 - Configure the Ring Again on No Answer and Call Completion on No Response (CCNR) option, and a Private Integrated Services Network Exchange (PINX) node Directory Number. on page 426
6. Table 196: LD 57 - Configure the Flexible Feature Code for Calling Private Privacy, and define whether Flexible Feature Code confirmation tone is to be provided. on page 426
7. Table 197: LD 17 - Configure Network Attendant Service on an MCDN D-Channel. on page 427
8. Table 198: LD 16 - Configure the method of Operation Coding for the Call Completion to a Busy Subscriber supplementary service for the associated QSIG or EuroISDN BRI trunk route. on page 427
9. Table 199: LD 16 - Configure the method of Operation Coding for the Call Completion on No Reply supplementary service, for the associated QSIG ISDN BRI trunk route. on page 428

This section contains the procedures required to configure the Call Completion to a Busy Subscriber (CCBS) over QSIG and EuroISDN networks, and Call Completion on No Response (CCNR) over a QSIG network, on a system.

ISDN PRI implementation

When configuring the interface type, consider the following:

- In order to support countries that have not yet upgraded to the ETS 300 403 standard (as introduced by the EuroISDN ETS 300 403 Compliance Update feature), the system still interworks with Central Offices conforming to the ETS 300 102 standard. So, when programming the D-Channel for PRI2 trunks (in LD 17) or a PRI2 route for ISDN trunks (in LD 16) for an ETS 300 403 interface, the following applies:
- If IFC = E403 and CNTY = ETSI (ETS 300 403 for the user side) or NET (ETS 300 403 for the network side), then the interface is fully compliant with ETS 300 403.
- If IFC = E403 and CNTY = any of the supported country entries except ETSI and NET, then the interface behaves like an ETS 300 102 extended version, that is, in addition to the existing ETS 300 102 capabilities, the bearer capability and High Layer Compatibility selection procedures (fall-back mechanism), and the basic telecommunication service identification are also implemented in order to take advantage of new teleservices, such as 7kHz telephony and Videotelephony.

  A user can still configure an interface fully compliant with the ETS 300 102 standard if IFC = EURO and CNTY = any of the supported country values.

Consider the following when configuring the method of Operation Coding for a QSIG interface:

- When the QSIG interface is ISO (IFC = ISGF), operations are mostly coded with Integer Values (RCAP = CCBI or CCNI).
- For ETSI interfaces (IFC = ESGF), the operation coding depends on the standard version. The current versions generally use Integer Values (RCAP = CCBI or CCNI), whereas some older versions can use Object Identifies (RCAP = CCBO or CCNO).

Table 191: LD 17 - Configure the method of Operation Coding for the Call Completion to a Busy Subscriber supplementary service for the associated QSIG or EuroISDN D-Channel.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	ADAN	Action Device and Number.
- ADAN	NEW DCH xx	Add a D-Channel on port 0-63
...		
- IFC	ESGF ISGF EURO E403	ETSI QSIG interface with GF platform. ISO QSIG interface with GF platform. EuroISDN. EuroISDN Interface conforming to the ETS 300 403, or the extended ETS 300 102 version for country-specific interfaces.

Prompt	Response	Description
- - CNTY		Enter country pertaining to the E403 interface. If either ETSI or NET are entered, the interface is fully compliant with ETS 300 403. .
	ETSI	ETS 300 403 for the user side.
	NET	ETS 300 403 for the network side. If any of the countries listed below are entered, the interface functions with the extended ETS 300 102 capabilities.
		Enter country pertaining to EuroISDN interface (this prompt is generated if IFC = EURO).
	AUS	Austria.
	DEN	Denmark.
	(ETSI)	ETS 300-102 basic protocol.
	FIN	Finland.
	GER	Germany.
	ITA	Italy.
	NOR	Norway.
	POR	Portugal.
	SWE	Sweden.
	EIR	Ireland.
	DUT	Holland.
	SWI	Switzerland.
	BEL	Belgium.
	ESP	Spain.
	UK	United Kingdom.
	FRA	France.
	CIS	Commonwealth of Independent States (Russia and the Ukraine).
...		
- RCAP		Enter the Operation Coding method for the QSIG/ETSI CCBS supplementary service.
	CCBO	Coding by Object ID.
	CCBI	Coding by Integer Value.
	XCCB	Remove the CCBO or CCBI value. **Note:** CCBO and CCBI are mutually exclusive.

Note:

The value is configured as a remote capability. It can be either by Object Identifier, used for ETSI, or by Integer Value, used for ISO.

Table 192: LD 17 - Configure the method of Operation Coding for the Call Completion on No Reply supplementary service for the associated QSIG D-Channel.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	ADAN	Action Device and Number.
- ADAN	NEW DCH xx	Add a D-Channel on port 0-63
...		
- IFC	ESGF ISGF	ETSI QSIG interface with GF platform. ISO QSIG interface with GF platform.
...		
- RCAP		Enter the Operation Coding method for the QSIG CCNR supplementary service.
	CCNO CCNI XCCN	Coding by Object ID. Coding by Integer Value. Remove the CCBO or CCBI value.
		Note: CCNO and CCNI are mutually exclusive.

Table 193: LD 10 - Configure Ring Again Class of Service on analog (500/2500-type) phones.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	500	Type of phone.
TN	l s c u	Terminal Number for Large System and CS 1000E.
DES	d...d	Office Data Administration System (ODAS) Station Designator, 1-6 alphanumeric characters.
CUST	0-99	Customer number as defined in LD 15.
CLS	(XRD) XRA	Deny or allow Ring Again Class of Service.

Table 194: LD 15 - Configure a Special Prefix Number (SPRE) for a customer.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	FTR	Features and options data.
CUST	0-99	Customer number as defined in LD 15.

Prompt	Response	Description
SPRE	xxxx	Special Prefix Number (xxxx = 1-4digits).

Table 195: LD 15 - Configure the Ring Again on No Answer and Call Completion on No Response (CCNR) option, and a Private Integrated Services Network Exchange (PINX) node Directory Number.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	NET	Networking data.
CUST	0-99	Customer number as defined in LD 15.
OPT	(RND) RNA	Deny or allow Ring Again on No Answer or Call Completion on No Response.
ISDN	YES	Integrated Services Digital Network.
- PINX_DN	xx...x	Private Integrated Services Network Exchange node Directory Number (up to seven digits). Precede the existing value with an X to delete it. If the Call Completion to a Busy Subscriber (CCBS) service does not provide CLID to the MCDN or DPNSS1 gateway, the PINX_DN is used to build the calling number. On an incoming CCBS request, the PINX_DN is used to determine the length of the DN extension.

Table 196: LD 57 - Configure the Flexible Feature Code for Calling Private Privacy, and define whether Flexible Feature Code confirmation tone is to be provided.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	FFC	Flexible Feature Code data.
CUST	0-99	Customer number as defined in LD 15.
FFCT	(NO) YES	(Do not) provided FFC confirmation tone after the FFC is dialed.
CODE	RGAA	The type of FFC is Ring Again Activate.
- RGAA	xxxx	Enter the Ring Again Activate FFC.
CODE	RGAA	The type of FFC is Ring Again Activate.
- RGAD	xxxx	Enter the Ring Again Deactivate FFC.
CODE	RGAA	The type of FFC is Ring Again Activate.
- RGAV	xxxx	Enter the Ring Again Verify FFC.

Table 197: LD 17 - Configure Network Attendant Service on an MCDN D-Channel.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	ADAN	Action device and number.
- ADAN	CHG DCH xx	Change a D-channel on logical port 0-63, for Large Systems.
- IFC	SL1	MCDN interface.
- NASA	YES	Allow Network Attendant Service.

ISDN BRI implementation

Table 198: LD 16 - Configure the method of Operation Coding for the Call Completion to a Busy Subscriber supplementary service for the associated QSIG or EuroISDN BRI trunk route.

Prompt	Response	Description
REQ	NEW	Add new data (ISDN BRI protocol group settings).
TYPE	RDB	Route Data Block.
CUST	0-99	Customer number as defined in LD 15.
ROUT	0-511	Route number for Large System and CS 1000E system.
TKTP	aaa	Trunk type.
DTRK	YES	Digital Trunk Route.
BRIP	NO	ISDN BRI packet handler route (NO is entered, since packet data is not required).
- DGTP	BRI	Digital trunk type.
ISDN	YES	Integrated Services Digital Network.
- IFC	ESGF ISGF EURO E403	ETSI QSIG interface with GF platform. ISO QSIG interface with GF platform. EuroISDN. EuroISDN Interface conforming to the ETS 300 403, or the extended ETS 300 102 version for country-specific interfaces.
- - CNTY		Enter country pertaining to the E403 interface. If either ETSI or NET are entered, the interface is fully compliant with ETS 300 403.
	ETSI	ETS 300 403 for the user side.

Prompt	Response	Description
	NET	ETS 300 403 for the network side. If any of the countries listed following are entered, the interface functions with the extended ETS 300 102 capabilities. Enter country pertaining to EuroISDN interface (this prompt is generated if IFC = EURO).
	AUS DEN (ETSI) FIN GER ITA NOR POR SWE EIR DUT SWI BEL ESP UK FRA CIS	Austria. Denmark. ETS 300-102 basic protocol. Finland. Germany. Italy. Norway. Portugal. Sweden. Ireland. Holland. Switzerland. Belgium. Spain. United Kingdom. France. Commonwealth of Independent States (Russia and the Ukraine).
...		
RCAP		Enter the Operation Coding method for the QSIG/ETSI CCBS supplementary service.
	CCBO CCBI XCCB	Coding by Object ID. Coding by Integer Value. Remove the CCBO or CCBI value. **Note:** CCBO and CCBI are mutually exclusive.

Table 199: LD 16 - Configure the method of Operation Coding for the Call Completion on No Reply supplementary service, for the associated QSIG ISDN BRI trunk route.

Prompt	Response	Description
REQ	NEW	Add new data (ISDN BRI protocol group settings).
TYPE	RDB	Route Data Block.
CUST	0-99	Customer number as defined in LD 15.
ROUT	0-511	Route number for Large System and CS 1000E system.

Prompt	Response	Description
TKTP	aaa	Trunk type.
DTRK	YES	Digital Trunk Route.
BRIP	NO	ISDN BRI packet handler route (NO is entered, since packet data is not required).
- DGTP	BRI	Digital trunk type.
ISDN	YES	Integrated Services Digital Network.
...		
- IFC	ESGF ISGF	ETSI QSIG interface with GF platform. ISO QSIG interface with GF platform.
...		
RCAP		Enter the Operation Coding method for the QSIG CCNR supplementary service.
	CCNO CCNI XCCN	Coding by Object ID. Coding by Integer Value. Remove the CCBO or CCBI value. **Note:** CCNO and CCNI are mutually exclusive.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 34: ISDN QSIG Generic Functional Transport

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

This feature provides a generic transport platform that will support QSIG compliant supplementary services and ISDN networking applications on a Private Integrated Services Digital Network (PISN), in accordance with the International Standards Organization (ISO) and the European Telecommunication Standards Institute (ETSI.)

ISDN QSIG GF Transport is supported on the following interfaces with the D-channel configured as International Standards Organization (ISO QSIG) or European Telecommunication Standard Institute (ETSI QSIG):

- ISDN 1.5 Mbit/s Primary Rate Interface (PRI)
- ISDN 2 Mbit/s Primary Rate Interface (PRI2)
- ISDN Basic Rate Interface (BRI) Trunk
- ISDN Signaling Link (ISL)

Operating parameters

DN address translation requires the association with a customer number. For a BRI trunk Digital Subscriber Loop (DSL) interface, the customer number association is with the D-channel. For PRI, a prompt creates a customer number association with a given D-channel.

Feature interactions

There are no feature interactions associated with this feature.

Feature packaging

The following packages are required for ISDN QSIG GF Transport:

- ISDN QSIG GF Transport (QSIGGF) is package 305

The following packages are required for QSIG on the ISDN PRI interface:

- Integrated Services Digital Network (ISDN) package 145
- Primary Rate Access (PRI) package 146
- 2.0 Mbit Primary Rate Interface (PRI2) package 154
- International Primary Rate Access (IPRA) package 202
- Multi-purpose Serial Data Link (MSDL) package 222
- QSIG Interface (QISG) package 263
- QSIG GF Transport (QSIGGF) package 305

The following packages are required for ISDN Signaling Link (ISL) interface:

- Integrated Service Digital Network Signaling Link (ISL) package 147
- Multi-purpose Serial Data Link (MSDL) package 222
- QSIG Interface (QSIG) package 263
- QSIG GF Transport (QSGIGF) package 305

The following packages are required for ISL shared mode:

- Integrated Services Digital Network (ISDN) package 145
- Primary Rate Access (PRI) package 146
- 2.0 Mbit Primary Rate Interface (PRI2) package 154

Feature implementation

Task summary list

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

1. Table 200: LD 16 - Configure the Route Data Block. on page 433
2. Table 201: LD 17 - Configuration Record. on page 434

Table 200: LD 16 - Configure the Route Data Block.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	RDB	Route Data Block.
...		
DTRK	YES	Digital Trunk Route.
...		
DGTP	PRI PRI2 BRI	1.5 Mbit PRI. 2.0 Mbit PRI. Basic Rate Interface.
IFC	ESGF ISGF	ESGF = ETSI QSIG interface with GF platform. ISGF = ISO QSIG interface with GF platform. The ESIG and ISIG prompt can be entered if both QSIG and QSIGGF packages are included.

Table 201: LD 17 - Configuration Record.

Prompt	Response	Description
REQ	CHG	Change
TYPE	ADAN	Action Device and Number
ADAN	aaa aaa x	Add, Move or Change Input/Output Device
...		
- IFC	ESGF ISGF	ESGF = ETSI QSIG interface with GF platform. ISGF = ISO QSIG interface with GF platform.
PINX_CUST	xx	Customer number.
ISDN_MCNT	60-(300)-350	Layer 3 call control message count five per second time interval.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 35: ISDN QSIG Name Display

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

The Integrated Services Digital Network (ISDN) Q Reference Signaling Point (QSIG) Name Display feature provides three supplementary services pertaining to the calling or connected party name display. These supplementary services allow the calling/connected party to either present or restrict the display of name identification on an ISDN PRI or ISDN BRI link.

ISDN QSIG Name Display feature supports the following services:

- Calling Name Identification Presentation (CNIP),
- Connected Name Identification Presentation (CONP), and
- Calling/Connected Name Identification Restriction (CNIR).

Calling Name Identification Presentation

Calling Name Identification Presentation (CNIP) service is available to the called/connected party. When this service is enabled, the calling party's name is displayed on the connected party's phone. CNIP service is available on a permanent basis only, and requires the calling party to have the Class of Service Name Presentation Allowed, defined in LD 10, or LD 11.

Connected Name Identification Presentation

Connected Name Identification Presentation (CONP) is a service available to the calling party. When this service is enabled, the called/connected party's name is displayed on the calling party's phone. CONP is provided on a permanent basis only and requires the called/connected party to have the Class of Service Name Presentation Allowed, defined in LD 10, or LD 11.

Calling/Connected Name Identification Restriction

Calling/Connected Name Identification Restriction (CNIR) prevents the calling/connected party's name from being presented to a called/calling party. CNIR is invoked on either a permanent basis, provided the calling/connected party has a Class of Service Name Presentation Denied defined in LD 10 or LD 11, or on a per-call basis. When CNIR is activated on a per-call basis, the Class of Service Name Presentation Allowed is configured, and Calling Party Privacy (CPP) Flexible Feature Code (FFC) is dialed prior to initiating a call. This supplementary service restricts presentation of the calling/connected party's name during a normal call establishment and also when the possibility of name presentation arises from the operation of other features, such as Call Transfer, Call Forwarding or Hunting.

Operating parameters

The ISDN QSIG Name Display feature requires the QSIG General Functional (GF) transport platform. QSIG GF protocols provide name information that is contained in the facility information element. This protocol transports call control messages from one network to another.

Fifty characters string length is the maximum allowable size of name display information for QSIG. The Call Party Name Display (CPND) feature accepts 27 characters string length. If a name display exceeds these parameters, the name is truncated on the receiving phone.

Interworking with existing Meridian Customer Defined Network (MCDN) Name Display feature is dependent upon the remote capability of the associated D-channel defined in LD 16 for BRI routes and LD 17 for PRI routes.

Individual service profiles cannot override Calling Name Identification Restriction requests. Therefore, when name display information has been restricted, it is not possible to obtain this information on the display of the phone.

For ISDN BRI phones to have the capability to activate Calling/Connected Name Identification Restriction (CNIR) on a permanent basis, the PRES prompt in LD 27 must be defined PRES = NO.

BRI phones do not support Calling Party Privacy (CPP) Flexible Feature Codes (FFC). Therefore, BRI phones do not support CNIR for individual calls.

Calling/Connected Name Identification Restriction (CNIR) takes precedence over Calling Name Identification Presentation (CNIP) and Connected Name Identification Presentation (CONP) services.

Feature interactions

Call Modification

If an incoming QSIG with name display presentation allowed receives call modification treatment, such as Call Pickup, Call Transfer, Conference, Call Forward All Types or Hunt, the reason for call modification is not displayed on the calling party's phone. Name display information is not updated on the calling party's phone. However, if a CONNECT message is sent, the calling party's name display is updated on receipt of this message.

Call Pickup

An incoming QSIG call with name display presentation allowed has name information displayed on the phone that picks up the call. If the incoming QSIG call has presentation denied, the calling party's name is not displayed on the phone picking up the incoming call.

Call Pickup Network Wide

When a QSIG call with name display presentation allowed is picked up on a MCDN, the calling party's name information is displayed on the phone that answers the call. If presentation restricted is defined, then name information is not displayed.

Call Transfer

After the completion of a call transfer, an incoming QSIG call with name display presentation allowed has name information displayed on the destination phone. If the incoming QSIG call has presentation denied, name information is not displayed to the destination phone.

Call Transfer Network Wide

When a QSIG call with name display presentation allowed is transferred over a MCDN, the calling party's name information is displayed on the phone that answers the call. If presentation restricted is defined, then name information is not displayed.

Conference

Conference Network Wide

An incoming QSIG call with name display presentation allowed is conferenced locally. When a conferee drops out of the conference, calling party's name information is displayed and is passed on to another conferee. Name display information remains until the last local phone remains on the call.With presentation restricted, the calling party's name information is not displayed as conferees leave the call.

Call Forward All Calls

When an incoming QSIG call, with name display presentation allowed Name Display, is forwarded locally, the calling party's name information is displayed on the forwarding phone. With presentation restriction, the calling party's name information is not displayed to the destination phone.

Call Forward All Calls - Network Wide

When a QSIG call with name display presentation allowed is forwarded over a MCDN, the calling party's name information is displayed on the forwarded phone. If presentation restricted is defined, then name information is not displayed on the destination phone.

Call Party Name Display

Calling Party Name Display Denied

Call Party Name Display and Calling Party Name Display Denied interact with ISDN QSIG Name Display, depending on the Name Display configuration in LD 16 for BRI or LD 17 for PRI. When a QSIG network is interacting with an MCDN network providing network capability ND3, both the MCDN and QSIG Name Display feature function on the same level.

Calling Party Privacy

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

Hunt

When an incoming QSIG call with name display presentation allowed is hunted locally, the calling party's name information is displayed on the destination phone. With presentation restriction, the calling party's name information is not displayed.

Hunt - Network Wide

When a QSIG call with name display presentation allowed is forwarded over a MCDN, the calling party's name information is displayed on the forwarded phone. If presentation restricted is defined, then name information is not displayed on the forwarded phone.

Incoming DID Digit Conversion

Incoming DID Digit Conversion Network Wide

IDC trunk and name information is passed and displayed to the terminating party when no name information is received from the Direct Inward Dial (DID) trunk. The Incoming DID Digit

Conversion (IDC) feature is activated, and name information is associated with the converted digit sequence.

Name information received from a DID trunk takes precedence over an IDC trunk name.

Feature packaging

This feature requires the following packages:

- Digit Display (DDSP) package 19
- Calling Party Name Display (CPND) package 95
- International Supplementary Features (SUPP) package 131
- Flexible Feature Codes (FFC) package 139
- Calling Party Privacy (CPP) package 301;
- Integrated Services Digital Network (ISDN) package 145
- Primary Rate Access (PRI) package 146
- 2.0 Mbit Primary Rate Interface (PRI2) package 154
- International Primary Rate Access (IPRA) package 202
- Multipurpose Serial Data Link (MSDL) package 222
- ISDN Basic Call QSIG (QSIG) package 263
- QSIG GF Transport (QSIGGF) package 305

The following packages are required for a Basic Rate Interface QSIG Interface:

- Integrated Services Digital Network (ISDN) package 145
- Basic Rate Interface (BRI) package 216
- Multipurpose Serial Data Link (MSDL) package 222
- ISDN BRI Trunk Access (BRIT) package 233
- ISDN Basic Call QSIG (QSIG) package 263
- QSIG GF Transport (QSIGGF) package 305

Feature implementation

Task summary list

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

1. Table 202: LD 95 - Modify the Call Party Name Display (CPND) data block. on page 441
2. Table 203: LD 10 - Define Class of Service Name Display Allowed on analog (500/2500-type) phones. on page 442
3. Table 204: LD 11 - Define Class of Service Name Display Allowed on Meridian 1 Proprietary Phones. on page 442
4. Table 205: LD 12 - Define Class of Service Name Display Allowed on Attendant Consoles. on page 443
5. Table 206: LD 27 - Define data for ISDN BRI phones. on page 443
6. Table 207: LD 16 - QSIG BRI route configuration. on page 443
7. Table 208: LD 16 - Configure Remote Capability for an MCDN BRI route. on page 444
8. Table 209: LD 17 - Configure Remote Capability for QSIG D-channel. on page 444
9. Table 210: LD 17 - Configure Remote Capability for MCDN D-channel. on page 444
10. Table 211: LD 57 - Define the Calling Party Privacy Flexible Feature Code for Meridian 1 Proprietary and analog (500/2500-type) phones. on page 445

Note:

Call Party Name Display is also defined at the phone level in either LD 10 or LD 11 depending on type of phone.

Table 202: LD 95 - Modify the Call Party Name Display (CPND) data block.

Prompt	Response	Description
REQ	NEW	Create Call Party Name Display data block.
TYPE	CPND	Call Party Name Display data block.
CUST	0-99	Customer number as defined in LD 15.

Prompt	Response	Description
REQ	NEW	Create new name strings.
TYPE	NAME	Name Strings.
CUST	0-99	Customer number as defined in LD 15.
DN	xxxx	Directory Number.

Table 203: LD 10 - Define Class of Service Name Display Allowed on analog (500/2500-type) phones.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	500	Type of phone.
CUST	0-99	Customer number as defined in LD 15.
CLS	CNDA	Called Party Number Display Allowed. Allows called/connected party name to be viewed by calling party. (CNDD) = Call Number Display Denied (default).
CLS	NAMA	Name Display Allowed on the far end. (NAMD) = Name Display Denied (default).
FTR	CPND	Allow Call Party Name Display name assignment.

Table 204: LD 11 - Define Class of Service Name Display Allowed on Meridian 1 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.
CUST	0-99	Customer number as defined in LD 15.
CLS	CNDA	Call Party Name Display Allowed. Allows display of original DN name on redirection. (CNDD) = Call Party Name Display Denied (default).
CLS	DNDA	Dialed Name Display Allowed. This displays the originally dialed DN Name on redirected calls. (DNDD) = Dialed Name Display Denied.
CLS	NAMA	Name Display Allowed on the far end (default). (NAMD) = Name Display Denied.

Table 205: LD 12 - Define Class of Service Name Display Allowed on Attendant Consoles.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	2250	Attendant Console type.
TN	l s c u	Terminal Number for Large System and CS 1000E.
CPND	CNDA	Call Party Name Display Allowed. (CNDD) = Call Party Name Display Denied (default).

Table 206: LD 27 - Define data for ISDN BRI phones.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	TSP	Terminal Service Profile.
DSL	l s c dsl	Digital Subscriber Loop for Large System and CS 1000E system.
USID	x	User Service Identifier.
DN	xxx...x	Directory Number associated with TSP (1 to 7 digits).
- CLIP	YES	Calling Line Identification Presentation for Incoming Calls Allowed. NO = Denied.
- PRES	YES	Presentation of Name and Calling Line Identification (CLID) to far end on outgoing calls allowed. NO = Denied.

Table 207: LD 16 - QSIG BRI route configuration.

Prompt	Response	Description
REQ	CHG	Change.
TYPE	RDB	Route Data Block.
CUST	0-99	Customer number as defined in LD 15.
DTRK	xxxx	Digital Trunk Type for route.
- IFC	xxxx	Interface type for PRI route where xxxx is: ESGF = Interface ID for ETSI QSIG with GF capability. ISGF = Interface ID for ISO QSIG with GF capability.
- RCAP	NDS	Allow Name Display Services (NDS) as new remote capability.

Table 208: LD 16 - Configure Remote Capability for an MCDN BRI route.

Prompt	Response	Description
REQ	CHG	Change
TYPE	RDB	Route Data Block
CUST	0-99	Customer number as defined in LD 15.
DTRK	xxxx	Digital Trunk Type for route.
- IFC	xxxx	Interface type for D-channel where xxxx is: ESGF = Interface ID for ETSI QSIG with GF capability. ISGF = Interface ID for ISO with GF capability.
- RCAP	ND3	Allow Name Display 3 (ND3) remote capability.
- NCNA	YES	Network Calling Name Allowed.

Table 209: LD 17 - Configure Remote Capability for QSIG D-channel.

Prompt	Response	Description
REQ	CHG	Change
TYPE	ADAN	Action Device and Number
- ADAN	aaa dch x	Change or New D-channel where aaa = NEW or CHG.
...		
- IFC	SL1	Interface type for D-channel.
...		
- RCAP	NDS	Allow Name Display Services (NDS) as new remote capability.

Table 210: LD 17 - Configure Remote Capability for MCDN D-channel.

Prompt	Response	Description
REQ	CHG	Change
TYPE	ADAN	Action Device and Number
- ADAN	aaa dch x	Change Input/Output Device.
...		
- IFC	SL1	Interface type for D-channel.
...		
- RCAP	ND3	Allow Name Display 3 (ND3) as new remote capability.

Table 211: LD 57 - Define the Calling Party Privacy Flexible Feature Code for Meridian 1 Proprietary and analog (500/2500-type) phones.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	FFC	Flexible Feature Code data block.
CUST	0-99	Customer number as defined in LD 15.
FFCT	YES	Flexible Feature Confirmation Tone.
CPP	xxxx	Calling Party Privacy Flexible Feature Code (typically *67). CPP is only prompted if the CPP package is equipped.

Feature operation

To activate Calling/Connected Name Identification Restriction (CNIR) on a per call basis, the calling party dials the Calling Party Privacy (CPP) Flexible Feature Code prior to initiating the call.

Chapter 36: ISDN QSIG Name Display Enhancement

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

The ISDN QSIG Supplementary Service - Name Display feature, supports the following Name Display supplementary services across an ISDN QSIG network:

- Calling Name Identification Presentation (CNIP). This service is offered to the called party, to provide the name of the calling party. Once configured, CNIP is permanently activated.

  CNIP is based on the Class of Service of the originating phone, as configured in LD 10 or LD 11. If the Class of Service prompt is configured as Name Presentation Allowed (NAMA), the name of the calling party is displaprompt is set to Name Presentation Denied (NAMD), or if the calling parme is not available (due, for example, to the name not being registered, or because of call interworking), then CNIP ihe caller's name to the called party.

  Note:

  To restrict the presentation of a calling party on ISDN BRI phones, the Presentation (PRES) prompt in LD 27 is set to NO. This applies for the other Name Display supplementary services as well.
- Connected Name Identification Presentation (CONP). This service applies to an established call. It is offered to the calling party, providing alerting information, or the called or connected party name. Once configured, CONP is permanently activated.

  CNOP is based on the Class of Service of the terminating phone, as configured in LD 10 or LD 11. If the Class of Service prompt is configured as Name Presentation Allowed (NAMA), the name of the called party is displayed on the phone of the calling party. If the Class of Service prompt is set to Name Presentation Denied (NAMD), or if the called party name is not available (due, for example, to the name not being registered, or because of call interworking), then the called party name is not provided to the calling party.
- Calling/Connected Name Identification Restriction (CNIR). This service prevents the name display of a calling party from being presented to the called party.

  CNIR can be activated for all calls by setting the Class of Service prompt in LD 10 or LD 11 to Name Presentation Denied (NAMD). If the Class of Service has been set to Name Presentation Allowed (NAMA), CNIR can be activated for each call, by dialing the Calling Party Privacy Flexible Code before making a call.

  Note:

  The capability of activating CNIR on ISDN BRI phones is not supported.

These Name Display supplementary services are provided across an ISDN QSIG network using the QSIG Generic Functional (GF) protocol transport platform.

The QSIG Supplementary Service - Name Display Enhancement feature, provides the option of choosing the method of Operation Coding for the Name Display supplementary services

described above. The choice can be by Object Identifier, used for European Telecommunication Standard Institute (ETSI) interfaces, or by Integer Value, used for International Standards Organization (ISO) interfaces. The Operation Coding method is configured as a remote capability using the RCAP prompt for an ISDN PRI D-Channel for ISDN PRI, or for an ISDN BRI route for ISDN BRI trunking.

Operating parameters

QSIG allows a maximum of 50 characters to be displayed for Name Display information. However, up to only 27 characters are allowed with the Calling Party Name Display feature. Consequently, if the name display string exceeds 27 characters, it will be truncated.

The QSIG Supplementary Service - Name Display Enhancement feature can interwork with the MCDN Name Display feature, depending on the Network Name Display (ND) method (either ND1, ND2, or ND3) configured as the Remote Capability (using RCAP prompt in LD 17) for the associated D-Channel. ND3 ensured the same level of service between the MCDN Name Display and QSIG Name Display supplementary services.

The user-based service profile cannot be used to override the Calling Name Identification Restriction service.

For Calling/Connected Name Identification Restriction (CNIR), restricting the presentation of name information is supported only for a calling party.

Busy name information sent with a DISCONNECT message is only supported for Overlap Receiving (the OVLR prompt is set to YES in LD 17).

Feature interactions

Call Pickup (over a QSIG link)

Referring to Figure 19: Call Pickup over a QSIG link on page 450, Station A, at the originating node, calls Station B, at the terminating node, over a QSIG link. Station C, also at the terminating node, picks up the call.

If Station A has a Class of Service of Name Presentation Allowed (CLS = NAMA in LD 10 or LD 11), then Station A's name information will be presented to Station C. If Station A has a Class of Service of Name Presentation Denied (CLS = NAMD in LD 10 or LD 11), then Station A's name information will not be presented to Station C. Station A's display will be updated

with the connected party (Station C) information, once the call pickup has been established. The reason for the call modification (Pickup) will not be presented to Station A.

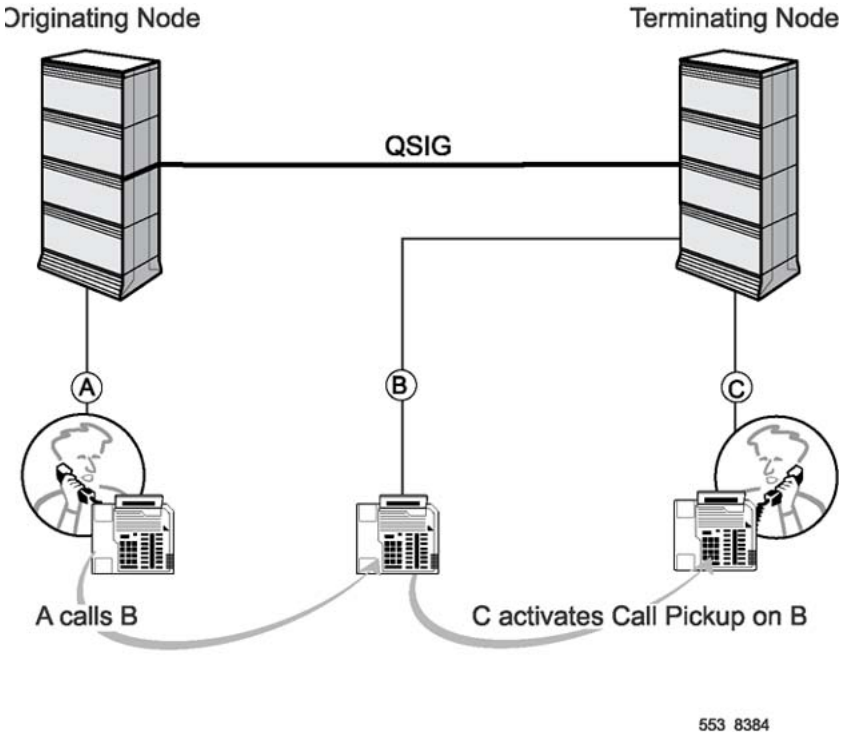


Figure 19: Call Pickup over a QSIG link

Call Transfer (over a QSIG link)

Referring to Figure 20: Call Transfer over a QSIG link on page 451, Station A, at the originating node, calls Station B, at the terminating node, over a QSIG link. Station B transfers the call to Station C, also at the terminating node.

If Station A has a Class of Service of Name Presentation Allowed (CLS = NAMA in LD 10 or LD 11), then Station A's name information will be presented to Station C, after the transfer has been completed.

If Station A has a Class of Service of Name Presentation Denied (CLS = NAMD in LD 10 or LD 11), then Station A's name information will not be presented to Station C, after the transfer has been completed. Station A's display will not be updated with the connected party (Station C) information, once the call transfer has been completed. The reason for the call modification (Transfer) will not be presented to Station A.

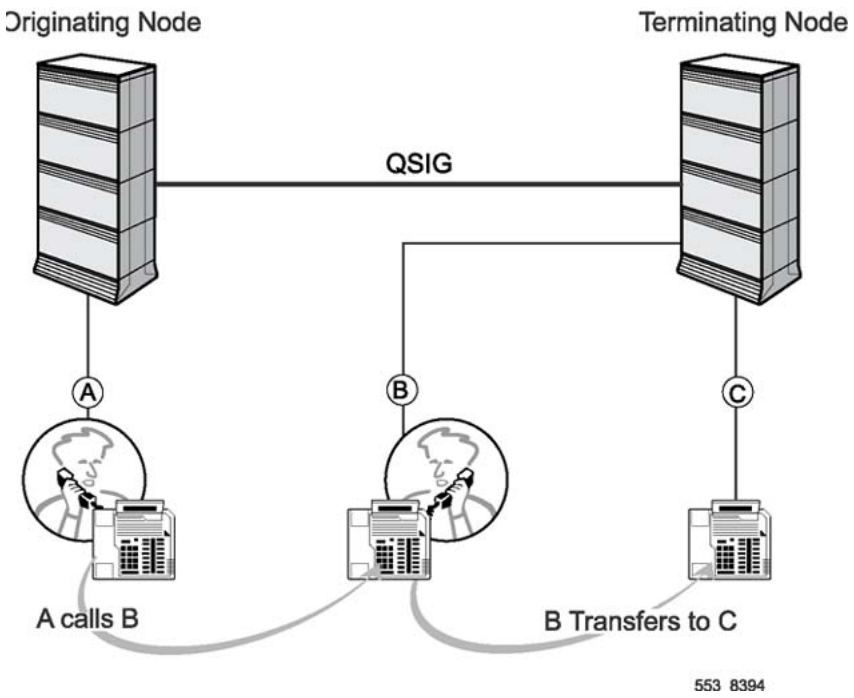


Figure 20: Call Transfer over a QSIG link

Conference (over a QSIG link)

Referring to Figure 21: Conference over a QSIG link on page 452, Station A, at the originating node, calls Station B, at the terminating node, over a QSIG link. Station B conferences the call to Station C, also at the terminating node.

If Station A has a Class of Service of Name Presentation Allowed (CLS = NAMA in LD 10 or LD 11), then Station A's name information will be presented to Station C, after Station B drops out of the conference. If Station A has a Class of Service of Name Presentation Denied (CLS = NAMD in LD 10 or LD 11), then Station A's name information will not be presented to Station C, after Station B drops out of the conference. Station A's display will be updated with Station B's name received in alert and connect messages.

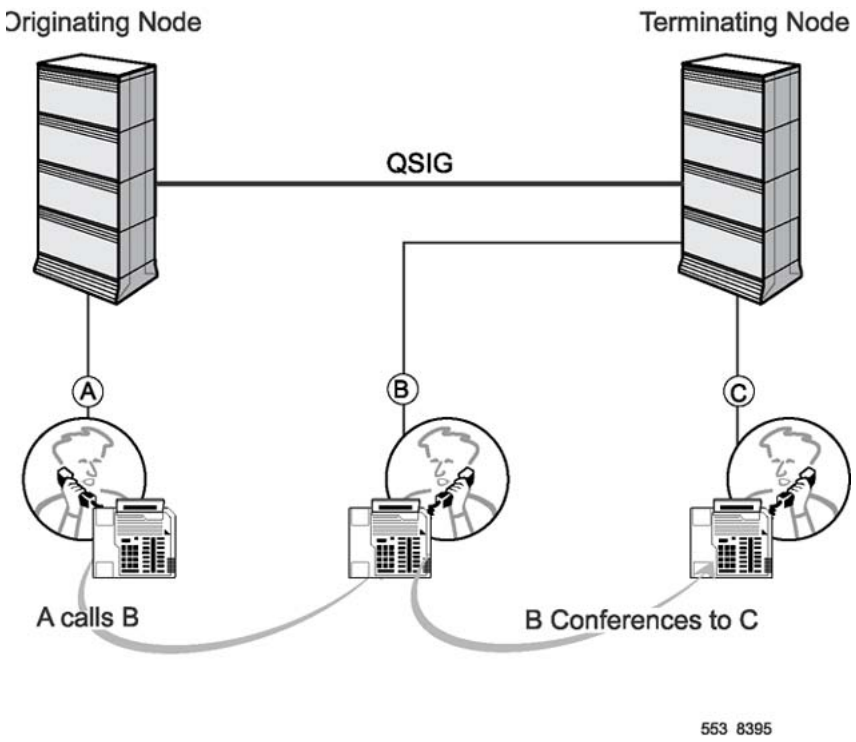


Figure 21: Conference over a QSIG link

Call Forward all types (over a QSIG link)

Call Forward all types comprises the following call modifications:

- Call Forward All Calls (CFW)
- Call Forward Busy (CFB)
- Call Forward by Call Type (CFT)
- Call Forward External Deny (CFX)
- Call Forward, Internal Calls
- Call Forward No Answer/Flexible Call Forward No Answer (CFNA)
- Call Forward No Answer, Second Level

Referring to Figure 22: Call Forward All Types over a QSIG link on page 453, Station A, at the originating node, calls Station B, at the terminating node, over a QSIG link. Station B forwards the call to Station C, also at the terminating node.

If Station A has a Class of Service of Name Presentation Allowed (CLS = NAMA in LD 10 or LD 11), then Station A's name information will be presented to Station C, after Station B completes the call forward. If Station A has a Class of Service of Name Presentation Denied

(CLS = NAMD in LD 10 or LD 11), then Station A's name information will not be presented to Station C, after Station B completes the call forward.

In the case where Station B's phone rings before Station C's phone (during Call Forward No Answer), then Station A's display will be updated with Station C's name information, once Station C's phone rings. In the case where Station C's phone rings directly (during Call Forward All Calls), then Station A's display will be updated directly with Station C's name information. The reason for the call forward (type of call forward) will not be presented to Station A.

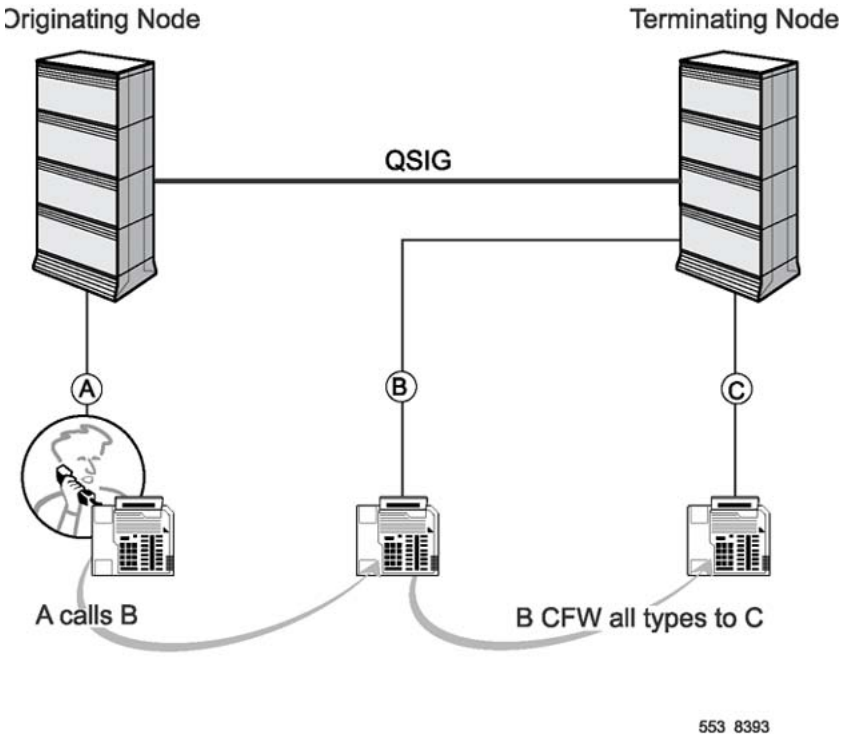


Figure 22: Call Forward All Types over a QSIG link

Hunt (over a QSIG link)

Referring to Figure 23: Hunt over a QSIG link on page 454, Station A, at the originating node, calls Station B, at the terminating node, over a QSIG link. The call is hunted from Station B to Station C, also at the terminating node.

If Station A has a Class of Service of Name Presentation Allowed (CLS = NAMA in LD 10 or LD 11), then Station A's name information will be presented to Station C, after the call hunt has been completed to Station C (Station C's phone is ringing). If Station A has a Class of Service of Name Presentation Denied (CLS = NAMD in LD 10 or LD 11), then Station A's name information will not be presented to Station C, after the call hunt has been completed to Station C (Station C's phone is ringing).

Station A's display will be updated with the Station C's name information, while Station C's phone is ringing. The reason for the call modification (Hunt) will not be presented to Station A.

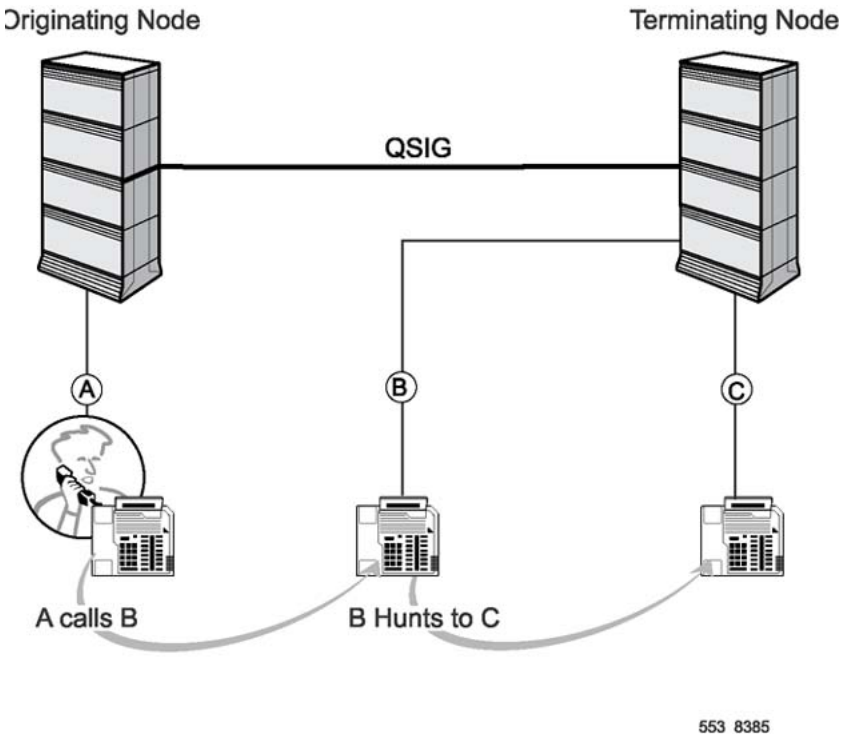


Figure 23: Hunt over a QSIG link

Incoming Digit Conversion (over a QSIG link)

Referring to Figure 24: Incoming Digit Conversion over a QSIG link on page 455, a Central Office DID call comes over a Direct Inward Dial (DID) trunk to an originating node, and terminates on Station C at a terminating node, over a QSIG link.

If no name information is received from the DID trunk, and Incoming Digit Conversion (IDC) is activated and the incoming name information is associated to the new digit sequence, then this name information is passed to Station C at the terminating node. If name information is received from the DID trunk, this takes precedence over the IDC trunk name.

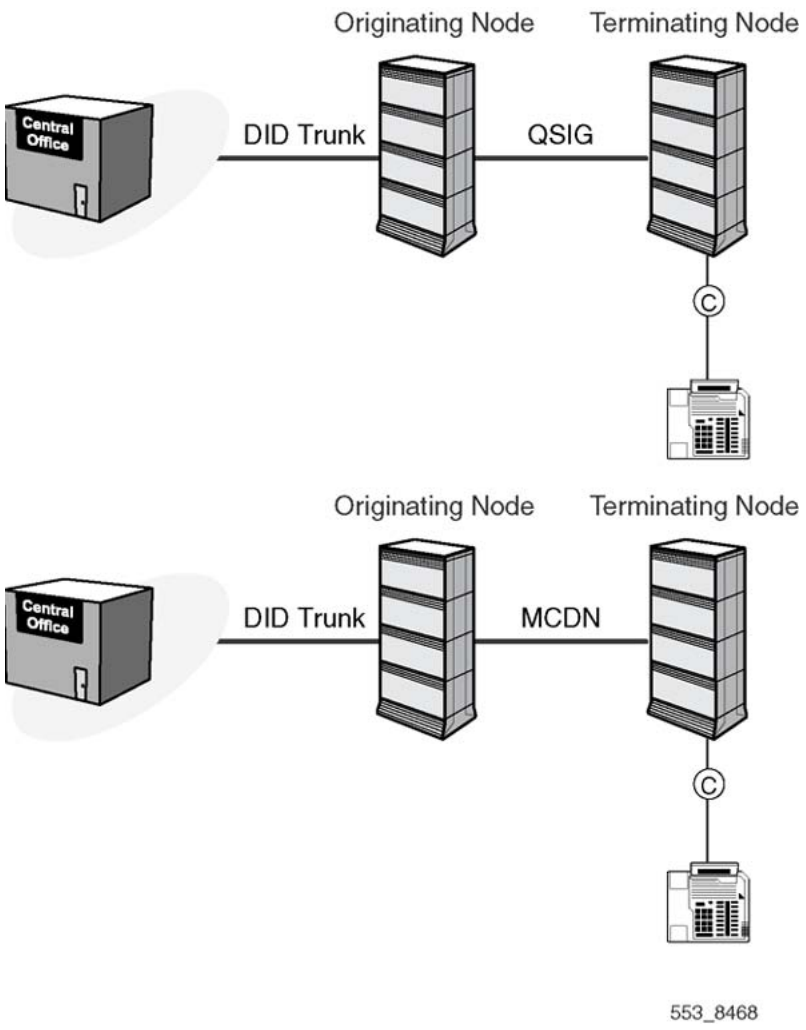


Figure 24: Incoming Digit Conversion over a QSIG link

Call Forward All Types (QSIG/MCDN link)

Call Forward all types comprises the following call modifications:

- Call Forward All Calls (CFW)
- Call Forward Busy (CFB)
- Call Forward by Call Type (CFT)
- Call Forward External Deny (CFX)
- Call Forward, Internal Calls
- Call Forward No Answer/Flexible Call Forward No Answer (CFNA)
- Call Forward No Answer, Second Level

Referring to Figure 25: Call Forward All Calls over a QSIG/MCDN link on page 456, Station A, at the originating node, calls Station B, at a tandem node, over a QSIG link. Station B forwards the call to Station C, at the terminating node, over an MCDN link.

If Station A has a Class of Service of Name Presentation Allowed (CLS = NAMA in LD 10 or LD 11), then Station A's name information will be presented to Station C, after Station B completes the call forward. If Station A has a Class of Service of Name Presentation Denied (CLS = NAMD in LD 10 or LD 11), then Station A's name information will not be presented to Station C, after Station B completes the call forward.

In the case where Station B's phone rings before Station C's phone (as in the case of Call Forward No Answer), then Station A's display will be updated with Station C's name information, once Station C's phone rings. In the case where Station C's phone rings directly (as in the case of Call Forward All Calls), then Station A's display will be updated directly with Station C's name information. The reason for the call forward (type of call forward) will not be presented to Station A.

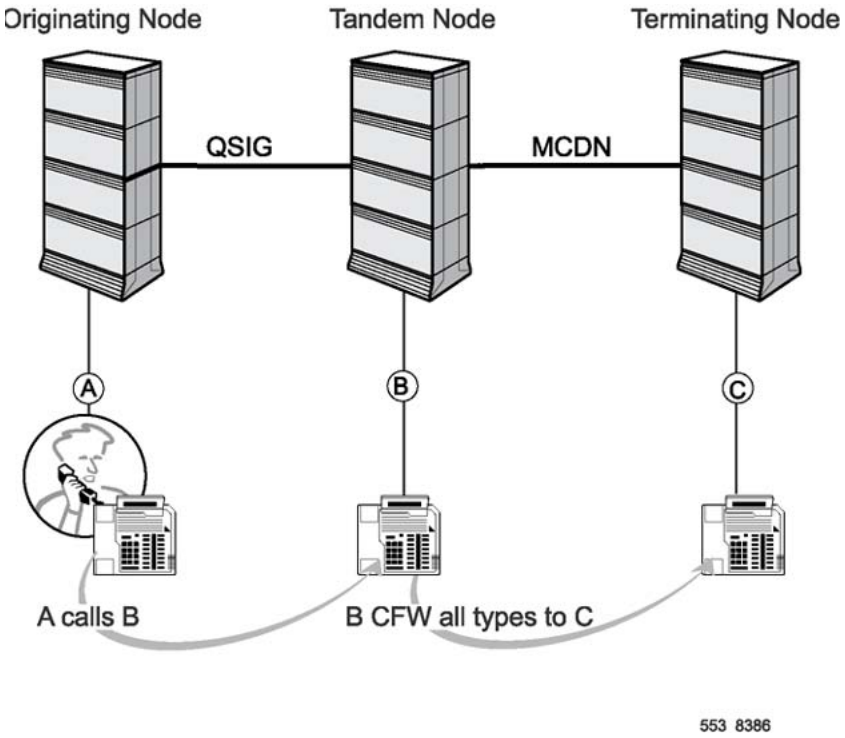


Figure 25: Call Forward All Calls over a QSIG/MCDN link

Hunt (QSIG/MCDN link)

Referring to Figure 26: Hunt over a QSIG/MCDN link on page 457, Station A, at the originating node, calls Station B, at the terminating node, over a QSIG link. The call is hunted over an MCDN link from Station B to Station C, at the terminating node.

If Station A has a Class of Service of Name Presentation Allowed (CLS = NAMA in LD 10 or LD 11), then Station A's name information will be presented to Station C, after the call hunt has been completed to Station C (Station C's phone is ringing).

If Station A has a Class of Service of Name Presentation Denied (CLS = NAMD in LD 10 or LD 11), then Station A's name information will not be presented to Station C, after the call hunt has been completed to Station C (Station C's phone is ringing). Station A's display will be updated with the Station C's name information, while Station C's phone is ringing. The reason for the call modification (Hunt) will not be presented to Station A.

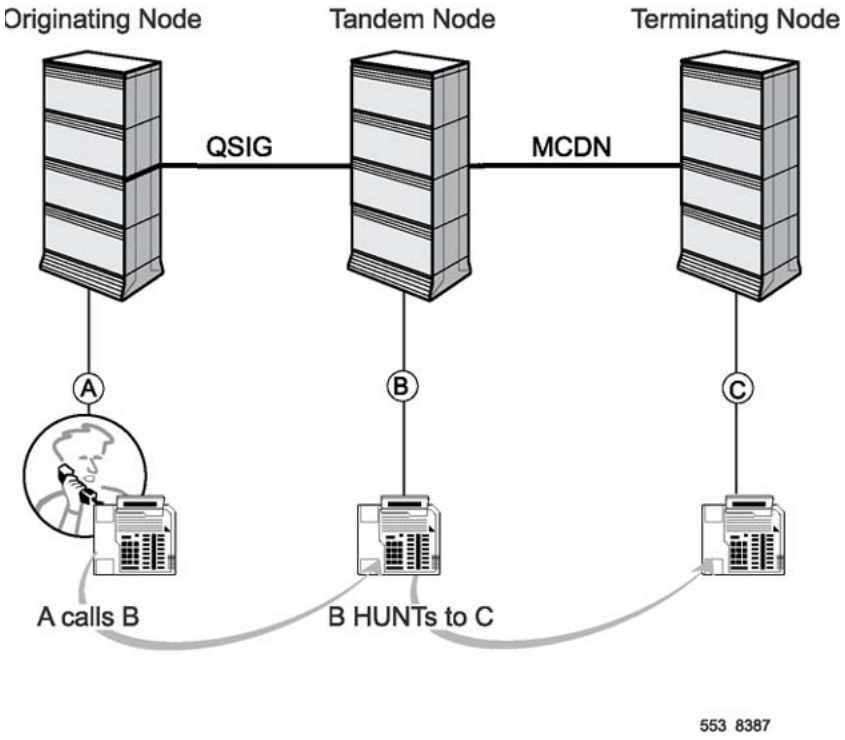


Figure 26: Hunt over a QSIG/MCDN link

Call Pickup Network Wide (QSIG/MCDN link)

Referring to Figure 27: Call Pickup Network Wide over a QSIG/MCDN link on page 458, Station A, at the originating node, calls Station B, at the terminating node, over a QSIG link. Station C, at the terminating node, picks up the call over an MCDN link.

If Station A has a Class of Service of Name Presentation Allowed (CLS = NAMA in LD 10 or LD 11), then Station A's name information will be presented to Station C. If Station A has a Class of Service of Name Presentation Denied (CLS = NAMD in LD 10 or LD 11), then Station A's name information will not be presented to Station C. Station A's display will be updated with the connected party (Station B) information, once the call pickup has been established. The reason for the call modification (Pickup) will not be presented to Station A.

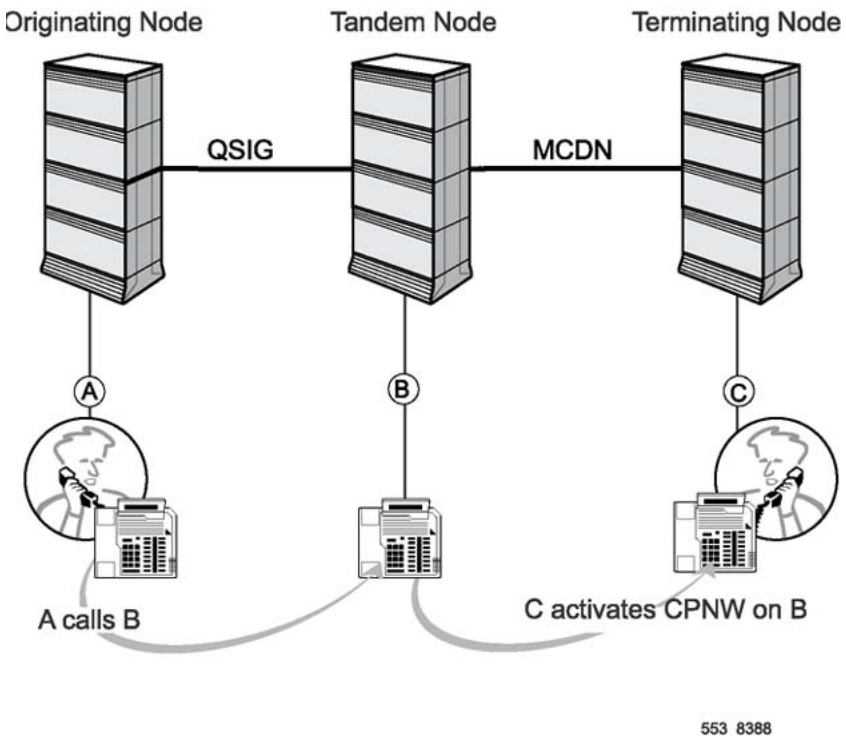


Figure 27: Call Pickup Network Wide over a QSIG/MCDN link

Call Transfer (QSIG/MCDN link)

Referring to Figure 28: Call Transfer over a QSIG/MCDN link on page 459, Station A, at the originating node, calls Station B, at the terminating node, over a QSIG link. Station B transfers the call to Station C, at the terminating node, over an MCDN link.

If Station A has a Class of Service of Name Presentation Allowed (CLS = NAMA in LD 10 or LD 11), then Station A's name information will be presented to Station C, after the transfer has been completed.

If Station A has a Class of Service of Name Presentation Denied (CLS = NAMD in LD 10 or LD 11), then Station A's name information will not be presented to Station C, after the transfer has been completed. Station A's display will not be updated with the connected party (Station C) information, once the call transfer has been completed. The reason for the call modification (Transfer) will not be presented to Station A.

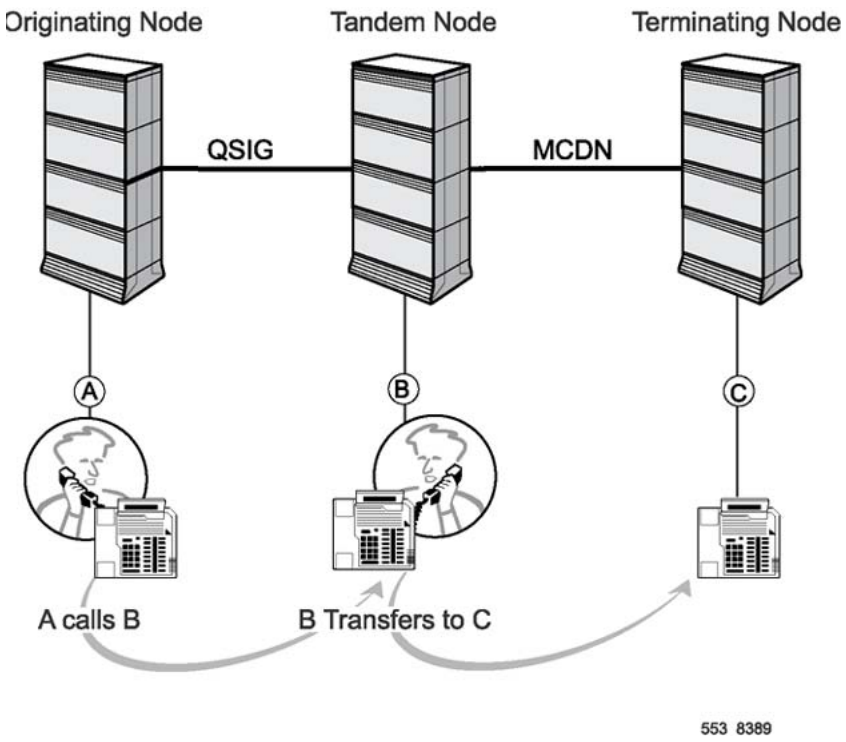


Figure 28: Call Transfer over a QSIG/MCDN link

Conference (QSIG/MCDN link)

Referring to Figure 29: Conference over a QSIG/MCDN link on page 460, Station A, at the originating node, calls Station B, at the terminating node, over a QSIG link. Station B conferences the call to Station C, at the terminating node, over an MCDN link.

If Station A has a Class of Service of Name Presentation Allowed (CLS = NAMA in LD 10 or LD 11), then Station A's name information will be presented to Station C, after Station B drops out of the conference. If Station A has a Class of Service of Name Presentation Denied (CLS = NAMD in LD 10 or LD 11), then Station A's name information will not be presented to Station C, after Station B drops out of the conference. Station A's display will be updated with Station B's name received in alert and connect messages.

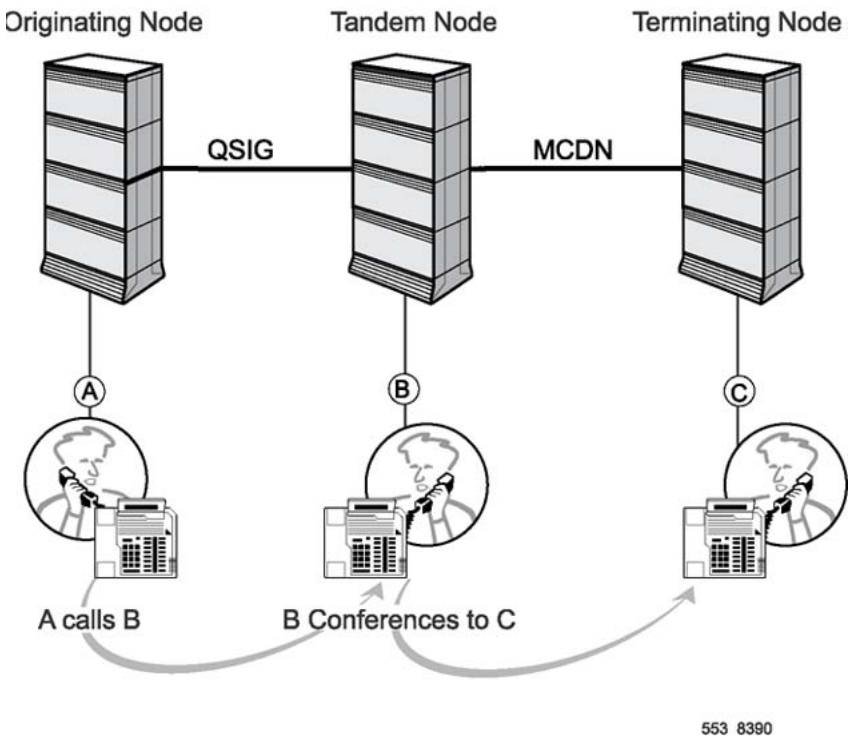


Figure 29: Conference over a QSIG/MCDN link

Incoming Digit Conversion (QSIG/MCDN or MCDN/QSIG link)

Referring to Figure 30: Incoming DID Digit Conversion over a QSIG/MCDN, or MCDN/QSIG link on page 461, A Central Office DID call comes over a Direct Inward Dial (DID) trunk to an originating node. The call is routed to a tandem node over a QSIG link. From the tandem node, it is routed to Station C at a terminating node, over an MCDN link. Or, a Central Office DID call comes over a Direct Inward Dial (DID) trunk to an originating node. The call is routed to a tandem node over an MCDN link. From the tandem node, it is routed to Station C at a terminating node, over a QSIG link.

If no name information is received from the DID trunk, and Incoming Digit Conversion (IDC) is activated and the incoming name information is associated to the new digit sequence, then this name information is passed to Station C at the terminating node. If name information is received from the DID trunk, this takes precedence over the IDC trunk name.

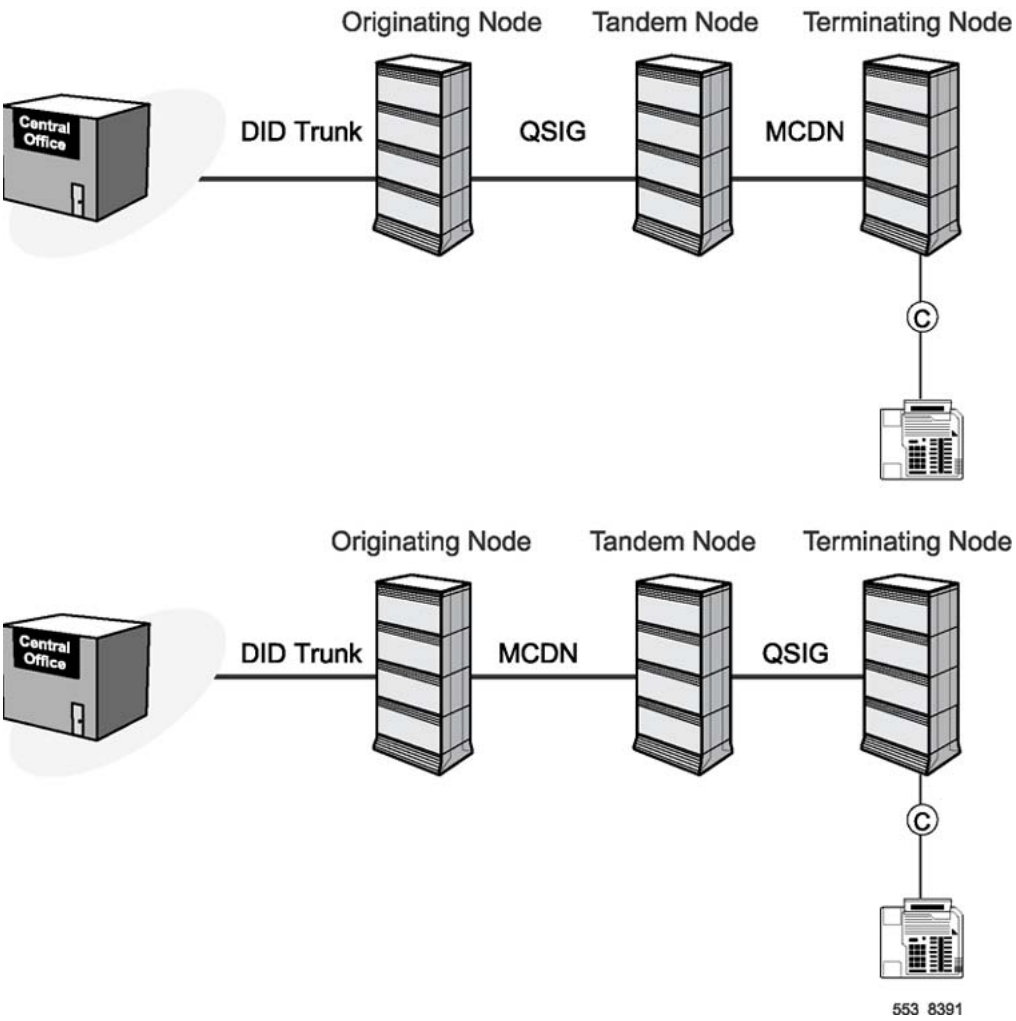


Figure 30: Incoming DID Digit Conversion over a QSIG/MCDN, or MCDN/QSIG link

MCDN Name Display features (QSIG/MCDN link)

When a QSIG network interworks with an MCDN network, consistent interworking of the QSIG Supplementary Service - Name Display Enhancement feature and the MCDN Name Display features depends on how the Remote Capability (RCAP) has been configured for the MCDN Name Display feature.

For MCDN, the RCAP prompt is used to configure the Network Name Display (ND) method (either ND1, ND2, or ND3) for the associated D-Channel. ND3 ensures the same level of service between the MCDN Name Display and QSIG Name Display supplementary services. Figure 31: MCDN Name Display features over a QSIG/MCDN link on page 462.

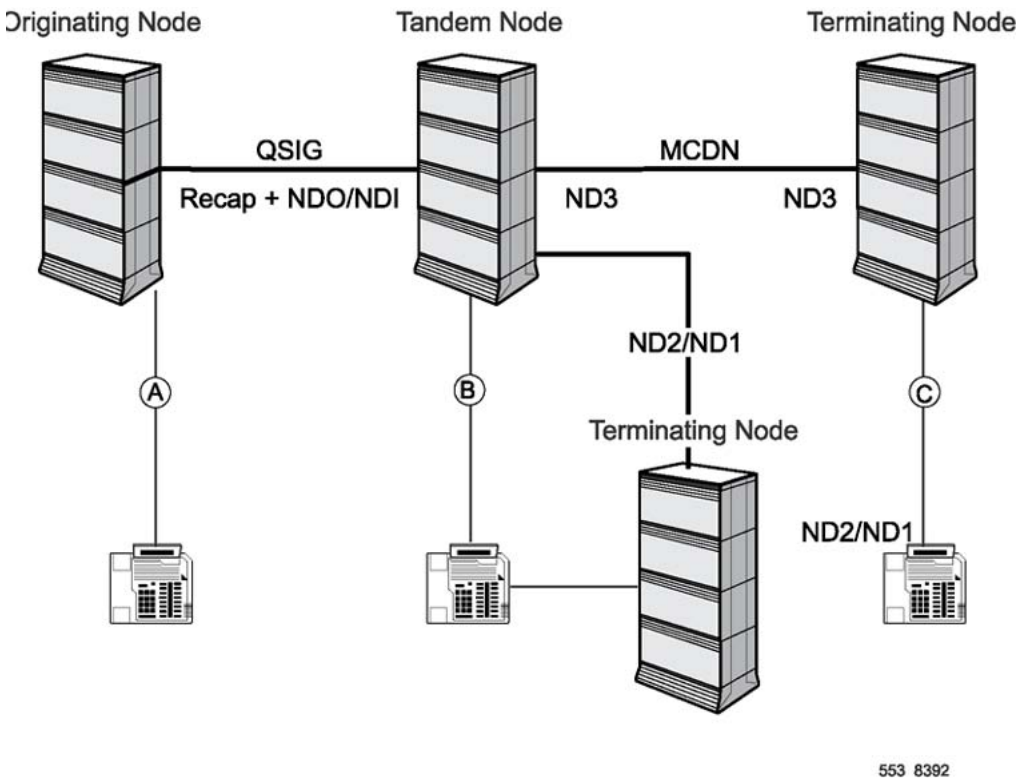


Figure 31: MCDN Name Display features over a QSIG/MCDN link

QSIG Call Diversion Notification

Calling Name

A QGIG Call Diversion occurs to a station. After the Diversion Notification Information is received at the diverted-to node, the station's display will be updated with the calling name information, if it has a Class of Service of Dialed Name Display Denied (CLS = DNDD in LD 10 or LD 11). If the phone has a Class of Service of Dialed Name Display Allowed (CLS = DNDA in LD 10 or LD 11), then another name, or none, is displayed on the phone.

Alerting and Connected Name

When the names from both a QSIG Name Display Service and a QSIG Call Diversion are included in the same message, the name from the QSIG Name Display Service will be ignored (the QSIG Name Display and the QSIG Call Diversion names should be the same).

QSIG Path Replacement

For QSIG Path Replacement, the name conveyed in any message of a new call is not displayed on any phone.

QSIG Call Transfer

Calling, Alerting, and Connected Name

For QSIG Call Transfer, the Calling, Alerting, and Connected Name are not provided at the transferring Private Integrated Services Network Exchange (PINX).

Connected Name

If connected name information is received, either as part of a QSIG Name Display Service or QSIG Call Transfer Notification, it will be displayed, unless there are subsequent presentation restriction options that can prevent it from being displayed.

Feature packaging

This feature requires the following packages:

- Q-reference Signaling Point Interface (QSIG) package 263
- QSIG Generic Functional Protocol Transport (QSIG-GF) package 305
- QSIG Supplementary Services (QSIG-SS) package 316
- Integrated Services Digital Network (ISDN) package 145
- Multi-purpose Serial Data Link (MSDL) package 222
- International Primary Rate Access (IPRA) package 202
- Primary Rate Access (PRI) package 146
- 2.0 Mbit Primary Rate Access (PRI2) package 154

For the QSIG Name Display service, the following packages are required:

- Digit Display (DDSP) 19
- Call Party Name Display (CPND) package 95
- International Supplementary Features (SUPP) package 131

  This allows the Digit and Name Party restriction capability (without package 131, Digit and Name presentation is always allowed).
- Calling Party Privacy (CPP) package 301
- Flexible Feature Code (FFC) package 139

For ISDN Basic Rate Interface Trunking (BRIT), the following additional packages are required:

- ISDN Basic rate Access (BRI) 216
- ISDN Basic Rate Trunk Access (BRIT) package 233

Feature implementation

Task summary list

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

1. Table 212: LD 95 - Configure Calling Party Name Display data block for a customer. on page 464
2. Table 213: LD 95 - Define a name string for a customer. on page 465
3. Table 214: LD 10 - Define the Calling Name Class of Service for analog (500/2500-type) phones. on page 465
4. Table 215: LD 11 - Define the Calling Name Class of Service for Meridian 1 Proprietary Phones. on page 466
5. Table 216: LD 12 - Define the Calling Name Class of Service for attendant consoles. on page 466
6. Table 217: LD 17 - Configure the method of Operation Coding for the Name Display supplementary services for the associated QSIG D-Channel. on page 467
7. Table 218: LD 57 - Configure the Flexible Feature Code for Calling Private Privacy. on page 467
8. Table 219: LD 16 - Configure the method of Operation Coding for the Name Display supplementary services for the associated QSIG ISDN BRI trunk route. on page 468
9. Table 220: LD 27 - Configure the CLIP service, and Calling Line Identification presentation option in the Terminal Service Profile for an ISDN BRI QSIG DSL. on page 469

This section contains the procedures required to configure the QSIG Supplementary Services - Name Display Enhancement feature on a system.

ISDN PRI implementation

Table 212: LD 95 - Configure Calling Party Name Display data block for a customer.

Prompt	Response	Description
REQ	NEW	Add new data.

Prompt	Response	Description
	CHG	Change existing data.
TYPE	CPND	Configure the Calling Party Name Display data block.
CUST	0-99	Customer number as defined in LD 15.

Table 213: LD 95 - Define a name string for a customer.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	NAME	Create a new Calling Party Name string.
CUST	0-99	Customer number, as previously entered in LD 95 when configuring the CPND data block. For Large System and CS 1000E system.
DN	xxx...x	Directory Number (1 to 7 digits) to which the name string is associated.

Table 214: LD 10 - Define the Calling Name Class of Service for analog (500/2500-type) phones.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	500	Type of analog phone.
TN	l s c u	Terminal Number for Large System and CS 1000E.
DES	d...d	Designator The response d...d represents an Office Data Administration System (ODAS) Station Designator of 1-6 alphanumeric characters.
CUST	0-99	Customer number, as previously entered in LD 95.
DN	x..x	Directory Number.
- CPND	NEW CHG OUT	Calling Party Name Display Add data block. Change existing data block. Remove existing data block.
- NAME		Calling Party Name Display name.
	aaaa yyyy	aaaa = first name, yyyy = last name.
- XPLN	xx	Expected name length.

Prompt	Response	Description
...		
CLS	NAMA	Allow calling name display at the far end. (NAMD) = Name Display Denied is the default.

Table 215: LD 11 - Define the Calling Name Class of Service for Meridian 1 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	l s c u	Terminal Number for Large System and CS 1000E.
DES	d...d	Designator The response d...d represents an Office Data Administration System (ODAS) Station Designator of 1-6 alphanumeric characters.
CUST	0-99	Customer number, as previously entered in LD 95 when configuring the CPND data block.
CLS		Enter each non-default Class of Service entry separated by a space.
	CNDA	Allow the calling name to be displayed on this phone. (CNDD) = Deny calling name display is default.
	DNDA	Allow the name of the original caller to be displayed on this phone, for redirected calls. (DNDD) = Dialed Name Display Denied is default.
	NAMA	Allow calling name display at the far end. (NAMD) = Name Display on other phone Denied is default.
KEY	xx aaa yyyy	Phone function key and assignments.
- CPND	NEW CHG OUT	Calling Party Name Display Add data block. Change existing data block. Remove existing data block.
- NAME		Calling Party Name Display name.
	aaaa yyyy	aaaa = first name, yyyy = last name.

Table 216: LD 12 - Define the Calling Name Class of Service for attendant consoles.

Prompt	Response	Description
REQ	NEW	Add new data.

Prompt	Response	Description
	CHG	Change existing data.
TYPE	aaaa	Type of attendant console.
TN	l s c u	Terminal Number for Large System and CS 1000E.
SETN	l s c u	Second Terminal Number for Large System and CS 1000E.
CUST	0-99	Customer number, as previously entered in LD 95 when configuring the CPND data block.
ANUM	1-63	Attendant Number.
CPND	CNDA	Allow the calling name to displayed on this console. (CNDD) = Call Party Name Display feature Denied is the default.

Note:

LD 17 – Configure the method of Operation Coding for the Name Display supplementary services for the associated QSIG D-Channel. The value is configured as a remote capability. It can be either by Object Identifier, used for ETSI, or by Integer Value, used for ISO.

Table 217: LD 17 - Configure the method of Operation Coding for the Name Display supplementary services for the associated QSIG D-Channel.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	ADAN	Action Device and Number.
- ADAN	CHG DCH xx	Change a D-Channel on port 0-63 for Large Systems.
- IFC	ESGF ISGF	ESGF = ETSI QSIG interface with GF platform. ISGF = ISO QSIG interface with GF platform.
- RCAP		Enter the Operation Coding method for the QSIG Name Display supplementary services.
	NDO NDI	NDO = Coding by Object ID (IFC should be set to ESGF). NDI = Coding by Integer Value (IFC should be set to ISGF). Precede the entry with an X to delete it. **Note:** NDO and NDI are mutually exclusive.

Table 218: LD 57 - Configure the Flexible Feature Code for Calling Private Privacy.

Prompt	Response	Description
REQ	CHG	Change existing data.

Prompt	Response	Description
TYPE	FFC	Flexible Feature Code data.
CUST	0-99	Customer number, as previously entered in LD 95 when configuring the CPND data block.
CODE	CPP	Configure a Flexible Feature Code for Calling Party Privacy.
CPP	xxxx	Enter the Flexible Feature Code for Calling Party Privacy.

ISDN BRI implementation

Table 219: LD 16 - Configure the method of Operation Coding for the Name Display supplementary services for the associated QSIG ISDN BRI trunk route.

Prompt	Response	Description
REQ	NEW	Add new data (ISDN BRI protocol group settings).
TYPE	RDB	Route Data Block.
CUST	0-99	Customer number as defined in LD 15.
ROUT	0-511	Route number for Large System and CS 1000E system.
TKTP	aaa	Trunk type.
DTRK	YES	Digital Trunk Route.
BRIP	NO	ISDN BRI packet handler route (NO is entered, since packet data is not required).
- DGTP	BRI	Digital trunk type.
ISDN	YES	Integrated Services Digital Network.
IFC	ESGF ISGF	ESGF = ETSI QSIG interface with GF platform. ISGF = ISO QSIG interface with GF platform.
RCAP		Enter the Operation Coding method for the QSIG Name Display supplementary services.
	NDO NDI	NDO = Coding by Object ID (IFC should be set to ESGF). NDI = Coding by Integer Value (IFC should be set to ISGF). Precede the entry with an ÔX' to delete it. **Note:** NDO and NDI are mutually exclusive.

For the following table for LD 27, configure the Calling Line Presentation (CLIP) service, and Calling Line Identification presentation option in the Terminal Service Profile for an ISDN BRI QSIG Digital Subscriber Loop (DSL).

Table 220: LD 27 - Configure the CLIP service, and Calling Line Identification presentation option in the Terminal Service Profile for an ISDN BRI QSIG DSL.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	TSP	Terminal Service Profile.
DSL		Digital Subscriber Loop
	l s c dsl	DSL location, where: • lll (superloop) = 0-156 (must be zero or a number divisible by 4) • s (shelf) = 0-1 • cc (card) = 0-15 and for CS 1000E 1–20 • dsl# (DSL location) = 0-7
DN	xxx...x	Directory Number (1 to 7 digits) associated with the TSP.
- CLIP	(YES) NO	Allow or deny Calling Line Identification Presentation for Incoming Calls.
- PRES	(YES) NO	Allow or deny Presentation of Calling Line Identification to far end, for outgoing calls.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 37: ISDN QSIG Path Replacement

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

ISDN QSIG Path Replacement allows an active connection through an ISDN QSIG private network to be replaced with more efficient connection. Path replacement service is invoked by the following triggers:

- QSIG Call Diversion (QSIG Call Diversion redirects Call Forwarding Busy, Call Forwarding No Reply and Call Forwarding Unconditional calls to another phone over a QSIG network).
- Network congestion on the first Private Integrated Service Network Exchange (PINX) node.

- Private Integrated Service Network Exchange (PINX) can be used interchangeably with Private Branch Exchange (PBX).
- Detection that the connected number differs from the called number.

QSIG Path Replacement does not directly attempt to reduce the number of trunks involved in a call. This feature attempts to replace non-optimum paths across a QSIG network with an optimum path. The optimum path between to PINX nodes is the path which takes the first choice route to all PINXs. The optimum path is not set up until the terminating phone answer the call.

ISDN QSIG Path Replacement uses the following replacement mechanisms: triangulation and anti-tromboning. These are described in more detail in Triangulation Path Replacement on page 473.

QSIG Path Replacement is permanently active on the PINX. This service is triggered internally or externally by another PINX. QSIG Path Replacement can be invoked on a mixed network of QSIG and Non-QSIG (such as MCDN). Path replacement uses the PINX_DN in the Customer Data Block as a rerouting number. Only QSIG interfaces are optimized. The Meridian Defined Customer Network (MCDN) is not optimized.

The section that follows discusses the use of Private Integrated Service Network Exchange (PINX) DNs as part of the Path Replacement feature.

ISDN PINX DN

The Path Replacement feature must scan the PINX nodes to find the optimum path to a remote PINX node. To accomplish this trunk optimization function, Path Replacement makes non-call associated requests (D-Channel messages only, no B-Channels are used) to the Path Replacement feature at the remote PINX node. The requests are not directed to any existing DN on the remote PINX node, but to a processing function on that PINX node.

To be able to route those types of calls through the network from one originating node to a destination node, an ISDN PINX DN must be defined for each PINX node in the network.

The ISDN PINX DN is a DN taken from the customer's numbering plan used to aid with the routing of network calls. It does not correspond to a real terminal on the node, so it is never busy. The ISDN PINX DN uses the same number of digits as the numbering plan; either UDP or CDP. It is defined so that the Path Replacement feature on a distant node can make non-call associated calls to the Path Replacement feature on another node when no particular destination exists. Those calls are made using existing routing configurations, e.g. a CDP distant steering code or a UDP location code expecting a fixed number of digits. For example, the far end PINX node has a digit sequence composed of the steering code plus the ISDN PINX DN with the same length and digit sequence used to route normal calls from the distant node to the local node for CDP type calls and location code plus a 4 digit ISDN PINX DN for UDP type calls.

One ISDN PINX DN should be configured for each network location. This ISDN PINX DN is used for the Path Replacement feature and other features such as Electronic Lock Network Wide/Electronic Lock for Private Lines and Call Pickup Network Wide features.

No user or feature other than Path Replacement, Electronic Lock Network Wide/Electronic Lock for Private Lines and Call Pickup Network Wide features will be allowed to dial the ISDN PINX DN. If a user dials this DN, they will receive overflow tone.

Triangulation Path Replacement

With Triangulation Path Replacement, either the originating or the terminating node involved in a QSIG call recognizes that a call is using a non-optimum path due to call diversion, network congestion or detection of a different number than dialed. If this call is still ringing, the originating/terminating node will wait for an answer signal before initiating triangulation path replacement. If the call is established, then replacement can be performed immediately.

The ISDN QSIG Path Replacement feature replaces non-optimum paths across a QSIG network with optimum paths. An optimum path between two nodes is the first choice route at all nodes.This feature does not attempt to reduce the number of trunks involved in call but does attempt to locate the optimum path.

As shown in Figure 32: Triangulation Path Replacement on page 473, Phone A on PINX 1 calls Phone B on PINX 2. Phone B has activated Call Forward All Calls to Phone C on PINX 3. Trunk optimization occurs between PINX 1 and PINX 3. Trunks between PINX 1 and PINX 2 and PINX 2 and PINX 3 are torn down after the call is answered by Phone C.

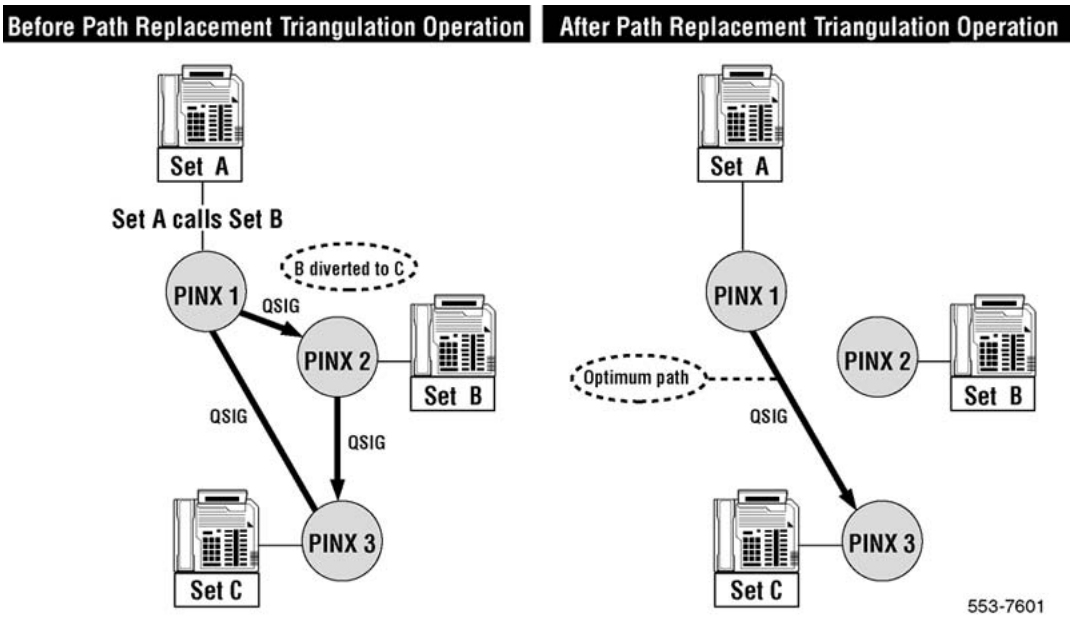


Figure 32: Triangulation Path Replacement

Anti-Tromboning Path Replacement

This feature performs anti-tromboning replacements to eliminate any dual parallel trunking that are caused by call modification. Anti-tromboning path replacement is only invoked after the

divert-to caller answer the call. Anti-tromboning is only performed when all the trunks are associated with the same customer.

As shown in Figure 33: Tromboning Path Replacement on page 474, Phone A on PINX 1 calls Phone B on PINX 2. Phone B has activated Call Forward All Calls to Phone C on PINX 1. Antitromboning occurs between PINX 1 and PINX 2 and these trunks are torn down after Phone C answers the call.

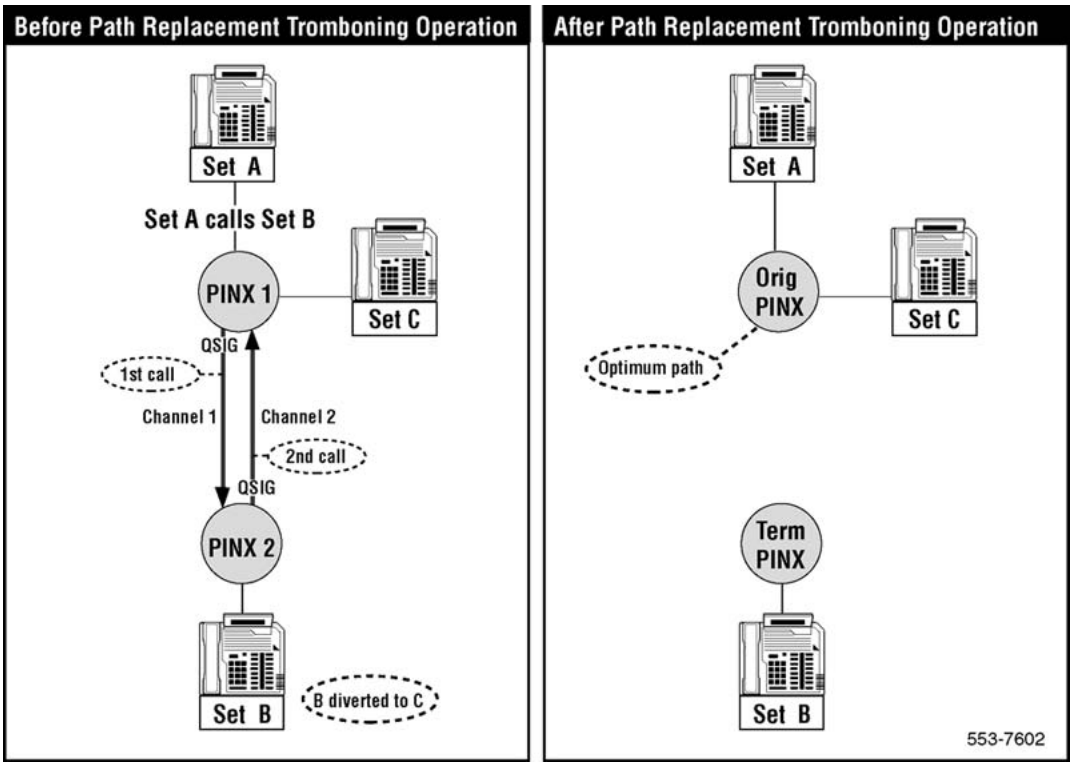


Figure 33: Tromboning Path Replacement

Retaining Path Replacement

When path replacement has been invoked, the cooperating PINX determines whether or not to retain part of the original connection provided that the original connection used the optimum route to reach the rerouted number. As shown in Figure 34: Path Replacement Retaining Part of Old Connection on page 475, the connection between the cooperating and transit PINX is retained after path replacement has occurred.

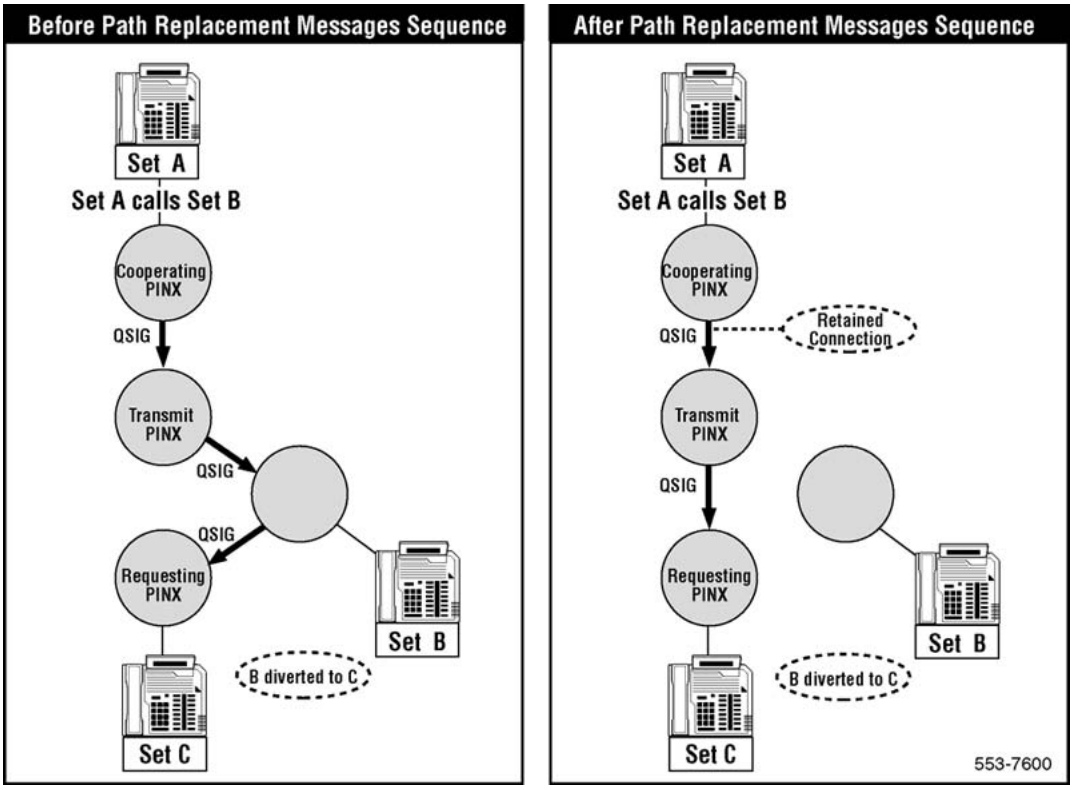


Figure 34: Path Replacement Retaining Part of Old Connection

Operating parameters

Network congestion occurs when a non-optimum path from the first PINX node is detected.

When call diversion is not supported by a PINX node, path replacement is triggered when the connected number differs from the called party's number.

ISDN QSIG Path Replacement requires the Multi-Serial Data Link (MSDL) card.

This feature is only supported on Coordinated and Uniform Dialing Plans. Route Access Codes are not supported. Digit manipulation is not supported.

If a path replacement request is triggered and fails with the cause "invalid Rerouting Number" the request is discontinued. No attempt is made with another rerouting number.

When a call has several triggers for path replacement and the first trigger fails, no other attempts are performed using the other triggers. Path replacement triggers are chosen according to the following order:

1. Diversion of QSIG Call
2. Congestion on the first PINX
3. Different Connected Number

The configuration of diversion and connected number triggers can generate a lot of path replacement collisions if the QSIG Call Diversion feature is configured on the QSIG network. In this situation, it is recommended that only the diversion trigger be configured. This establishes the optimized call in the same direction as the originating call.

New traffic reports are generated for QSIG Path Replacement. TSF009 reports traffic activity for D-channels and TFS011 reports traffic activity for Multi-Purpose ISDN Signalling processor for QSIG Path Replacement service.

Feature interactions

Attendant Console

Path replacement calls invoked by an attendant are not supported.

Attendant Barge In

Attendant Break In

The Attendant Barge In and Attendant Break In features are not supported during path replacement operation.

Automatic Call Distribution

ISDN QSIG Path Replacement calls placed in an Automatic Call Distribution (ACD) queue are initially rejected. Path replacements requests are accepted after the call is answered by an ACD agent.

Call Transfer

When a local is transferred by the system during an ISDN QSIG Path Replacement request, then the replacement request is terminated. After Call Transfer, the system can accept path replacement from another PINX.

Call Hold

Path Replacement cannot be invoked on a held call. Path replacement is only accepted on an active call.

Conference Calls

The system does not invoke path replacement when callers are involved in an established conference. Any replacement requests from a far end PINX are rejected when a call is on an established conference.

Customer Controlled Routing

Following a Customer Controlled rerouting, path replacement is only invoked by the connected number trigger.

Direct Inward System Access

Path replacement does not occur on a Direct Inward System Access (DISA) call. The system does not invoke the Path Replacement feature on DISA calls.

End-to-End Signalling

When End to End Signalling is detected the path replacement attempt is terminated. However, on Analog (500/2500 type) phones, replacement is not terminated because the Digit Tone Multi-Frequency (DTMF) generated by the phone is detected.

Music

Radio Paging

Recorded Announcement

The system does not invoke the Path Replacement feature on Music, Paged or Recorded Announcement calls.

Feature packaging

This feature requires the following packages:

- Network Alternate Route Selection (NARS) package 58 (for Uniform Dialing Plan), or Coordinated Dialing Plan (CDP) package 59
- Flexible Numbering Plan (FNP) package 160
- QSIG Interface (QSIG) package 263
- QSIG-GF Transport (QSIGGF) package 305
- QSIG Supplementary Services (QSIG-SS) package 316
- Primary Rate Access (PRI) package 146
- 2.0 Mbit Primary Rate Access (PRI2) package 154
- International Primary Rate Access (IPRA) package 202
- Multipurpose Serial Data Link (MSDL) package 222

ISDN Basic Rate Interface Trunk Access (BRIT) QSIG Interface requires:

- Basic Rate Interface (BRI) package 216
- ISDN Basic Rate Interface Trunk Access (BRIT) package 233

Feature implementation

Task summary list

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

1. Table 221: LD 15 - Configure Private Integrated Services Network Exchange Directory Number. on page 479
2. Table 222: LD 16 - Configure New Path Replacement parameters for an ISDN PRI or BRI trunk. on page 479
3. Table 223: LD 17 - Configure QSIG Path Replacement for a QSIG ETSI or ISO interface. on page 480

Table 221: LD 15 - Configure Private Integrated Services Network Exchange Directory Number.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	NET	Networking data.
CUST	0-99	Customer number as defined in LD 15.
ISDN	YES	Integrated Services Digital Network.
- PINX_DN	xx…x	Private Integrated Services Network Exchange Directory Number, up to seven digits.

Table 222: LD 16 - Configure New Path Replacement parameters for an ISDN PRI or BRI trunk.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	RDB	Route Data Block.
CUST	0-99	Customer number as defined in LD 15.
ROUT	0-511	Route number for Large System and CS 1000E system.
DTRK	YES	Digital Trunk Type.
- DGTP	PRI BRI	PRI trunk. BRI trunk.
- IFC	ESGF ISGF	ETSI QSIG interface with GF platform. ISO QSIG interface with GF platform.
…		
RCAP		Add Path Replacement as a remote capability.
	PRI PRO	PRI = The encoding method uses Integer values. PRO = The encoding method uses Object Identifier. PRI and PRO are mutually exclusive, and cannot be configured together at the same time on the same link. Precede PRI and PRO with X to remove. The previously configured PR parameters will be valid, if PR is deactivated and then reactivated.
…		
PR_TRIGS		Path Replacement Triggers.
	<CR>	<CR> = If REQ = NEW, the parameters are set to the default values, which are: DIV 2 3 (two PR attempts, with a three minute delay for Diversion triggers) CNG 2 3 (two

Prompt	Response	Description
		PR attempts, with a three minute delay for Congestion triggers) XCON (Connected Number is not a trigger). If REQ = CHG, the Path Replacement parameters are not modified.
	DIV xx-y CNG xx-y CON xx-y	DIV = Diversion is used to trigger Path Replacement. CNG = Congestion is used to trigger Path Replacement. CON = A Connected number different from a called number is used to trigger Path Replacement. • xx = 0-(2)-15, the number of Path Replacement attempts (the default is two). If 0 is entered, the number will be set to infinite. • yy = 1-(3)-7, the delay, in minutes, between two consecutive Path Replacement attempts (the default is three minutes). Precede DIV, CNG, and CON with X to remove the Path Replacement trigger.
PR_RTN	YES (NO)	Retain option is (is not) supported by the far end PINX.

Table 223: LD 17 - Configure QSIG Path Replacement for a QSIG ETSI or ISO interface.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	CFN	Configuration Record.
ADAN	CHG DCH xx	Change D-Channel number.
…		
- IFC	ESGF ISGF	ETSI QSIG interface with GF platform. ISO QSIG interface with GF platform.
…		
- RCAP	aaa	Add Path Replacement as remote capability where: PRI = Integer values as encoding method. PRO = Object identifier as encoding method. Precede with X to remove.
	DV3I DV3O	QSIG SS Call Diversion Notification remote capability. Configure sending of QSIG Diversion Notification Information, treatment of Rerouting request and coding of operations. If coded as Object Identifier, the remote capability ends with 'O", whereas for Integer Value, the remote capability ends with 'I'. Only one remote capability is allowed. DV3I = Diversion information is sent to remote switch. DV3O = Rerouting requests from remote switch are processed.

Prompt	Response	Description
		Precede with 'X' to remove capability, **but do not specify the coding type**; for example, XDV3 or XDV1 (not XDV3I or XDV1O).
…		
PR_TRIGS		Path Replacement Triggers.
	<CR>	<CR> = If REQ = NEW, the parameters are set to the default values, which are: DIV 2 3 (two PR attempts, with a three minute delay for Diversion triggers) CNG 2 3 (two PR attempts, with a three minute delay for Congestion triggers) XCON (Connected Number is not a trigger). If REQ = CHG, the Path Replacement parameters are not modified.
	DIV xx-y CNG xx-y CON xx-y	DIV = Diversion is used to trigger Path Replacement. CNG = Congestion is used to trigger Path Replacement. CON = A Connected number different from a called number is used to trigger Path Replacement. • xx = 0-(2)-15, the number of Path Replacement attempts (the default is two). If 0 is entered, the number will be set to infinite. • yy = 1-(3)-7, the delay, in minutes, between two consecutive Path Replacement attempts (the default is three minutes). Precede DIV, CNG, and CON with X to remove the Path Replacement trigger.
PR_RTN	YES (NO)	Retain option is (is not) supported by the far end PINX.

Feature operation

No specific operating procedures are required to use this feature.

Chapter 38: ISDN Semi Permanent Connections for Australia

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

The ISDN Semi Permanent Connections (ISPC) feature allows a number of B-channels, through an Australian ISDN Central Office (CO), to be permanently connected between the systems (nailed up B-channels). These connections are used transparently by the ISDN Signaling Link (ISL) as if they were leased lines or channels on a TIE trunk.

ISL calls between PBX end users linked by the permanent connections are performed using all capabilities provided by Meridian Customer Defined Network (MCDN) features. Other

possibilities provided by this feature are to use ISPC links to convey D-channel signaling between systems.

ISL calls through an established ISPC link use two trunks. The physical trunk is the permanent connection initiated by the Central Office between both PBXs on the B-channel of a PRI2 trunk. The logical trunk is an intermediate data structure without any associated hardware. This trunk is also referred to as the Phantom DTI2 trunk.

The physical trunk allows an ISPC link to be established between the PBXs; whereas, the logical trunk allows end users to perform calls, using established ISPC links.

Semi Permanent Connection Links to convey D-Channel Signals

An ISPC link can be used to convey D-channel signaling. However, D-channel signaling requires a different configuration than the ISPC link used by the ISDN Signaling Link (ISL). Both sides of an ISPC link that convey D-channel signaling must be equipped with either a Meridian 1 Communication Adaptor (MCA) or a Meridian 1 Communication Unit (MCU) data interface.

The Dial Tone Detection feature allows the data interfaces at each end of the ISPC links to synchronize. For synchronization to occur, one side of the ISPC link must be defined as MASTER and the other side must be defined as SLAVE.

Master Side

The data interface, connected to the D-channel, must be configured in autodialing mode. The dialing sequence configured on the MASTER SIDE, rings the data interface on the SLAVE SIDE of the ISPC link through the associated route of the Phantom TN. The Dial Tone Detection feature must be configured for this route so that dial tone is provided by the SLAVE SIDE of the ISPC link. The dial tone indicates to the MASTER SIDE that both legs of the ISPC link have been successfully setup by the Central Office.

As illustrated in Figure 35: Mechanism Used to Synchronize Data Interfaces on page 485, when the data interface initiates its dialing sequence, a path is set between the interface and the ISPC link. If no dial tone is detected, the path is released. Additional attempts are performed by the data interface because it has been configured in autodialing mode. When the dialing tone is successfully detected, the Directory Number associated to the data interface used on the ISPC SLAVE SIDE, is dialed through the ISPC link using Dual Tone Multi-Frequency (DTMF) tones.

Slave Side

The data interface connected to the SLAVE SIDE must be configured to autoanswer mode. When an ISPC link is established, a path is made with a TDS card to provide Dial Tone to the ISPC link, and DTMF digits are expected. The DTMF digits identify the Directory Number where

the data interface that is to ring is connected. When the data interface on the SLAVE SIDE is rung, it answers and initiates its own protocol sequence and D-channels can be established.

After 30 minutes of unsuccessful attempts to connect, the path between the ISPC link, the TDS card and the DTR cards is removed, provided that the default mode is set to "MANUAL and 1 attempt". A technician can change these default parameters in LD 96.

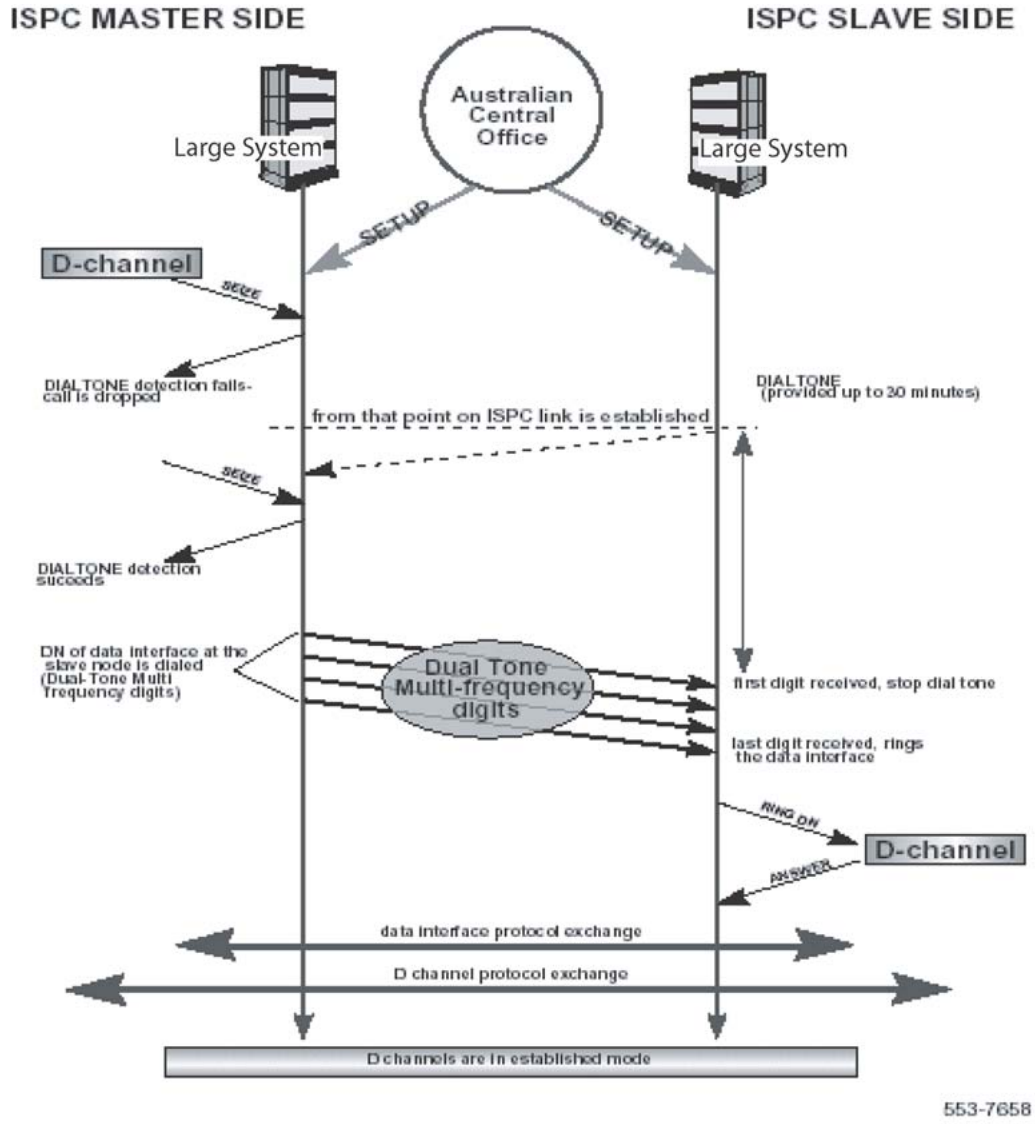


Figure 35: Mechanism Used to Synchronize Data Interfaces

Operating parameters

For this feature, system users must initially subscribe for ISPC services from Telecom Australia. ISDN Semi Permanent Connections (ISPC) service is only available from Telecom Australia on Central Office (CO) and Direct Inward Dialing (DID) trunks.

The ISPC feature is based on the Asia Pacific ISDN protocol for Australia. The ISPC requires PRI2 connections to an Australian Central Office using Asia-Pacific ISDN connectivity for Australia.

ISPC is not supported over Basic Rate Interface (BRI) trunks or AXE-10 Australia Primary Rate Interface connectivity.

Data interfaces, supported when the ISPC links are configured to convey D-channel signaling to forward signals from MSDL, DCHI or DDCHI card ports into the system, are Meridian Communication Unit (MCU) and Meridian Communication Adapter (MCA).

Integrated Signaling Links (ISL) in a D-channel configured shared or dedicated mode are supported.

ISPC reference numbers, provided by Telecom Australia, are limited to a maximum of seven digits.

Established ISPC links are only used in conjunction with the ISL feature functionality. Therefore, ISPC links are interpreted as available TIE trunks with an established ISL D-channel.

Outband signaling, through an ISL D-channel, is required to control the usage of established ISPC links. When an ISPC link conveys D-channel signaling, a special mechanism based on the Dial Tone Detection feature is used.

The number of established ISPC links on a PRI2 loop is limited to the number of configured links and the idle B-channels on the loop.

Each ISPC link requires an associated TN defined on a Phantom DTI2 loop.

Established ISPC links cannot be used for Virtual Network Services (VNS) bearer trunks.

ISPC links are established by the Australian Central Office (CO) as a 64 Kbit/s cleared data link. The CO and systems do not provide a mechanism for echo cancellation for ISL calls using ISPC links.

The Phantom DTI2 trunks used by this feature must belong to the same group as the real PRI2 trunk.

When establishing an ISL D-channel between two systems, the following hardware is required:

- D-channel (DCH), Dual Density D-channel (DDCHI) or Multi-Serial Data Link (MSDL) card on each system site;
- Meridian Communication Unit (MCU) or Meridian Communication Adapter (MCA) on each site for the exchange of D-channel signaling between sites; and
- Tone Detector Card (XTD or DTD) on one site and a Tone Generation Card (XCT or TDS) at the other site in situations where at least one ISPC link is used to convey D-channel signaling.

Feature interactions

Attendant Barge-In

When an attendant attempts to Barge-In on 2.0 Mbit/s Primary Rate Interface B-channel used as an ISPC link with the Central Office, a fast busy tone is provided.

Attendant Incoming Call Indicators

Calls using an ISPC link are always presented as calls over TIE trunks.

Call Detail Recording

Calls using an established ISPC link are identified by the calling party, called party, time and duration of call. Call Detail Recording tickets are not generated for the ISPC links that have been established.

Call Forward by Call Type

ISPC calls are handled according to the configuration of the route associated to the phantom trunk TN. This configuration is independent of the route associated to the real TN.

Dial Tone Detection

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

Distinctive

New Distinctive Ringing

For ISPC calls, Distinctive/New Distinctive Ringing is provided according to the configuration of the route associated to the phantom trunk TN. This configuration is independent of the route associated to the real TN.

Integrated Service Digital Network Signaling Link

Redirection or optimization of calls will not release established ISPC links.

Trunk Barring

For calls using or requesting an ISPC link, Trunk Barring is provided according to the configuration of the route associated to the phantom trunk TN. This configuration is independent of the route associated to the real TN.

Feature packaging

ISDN Semi Permanent Connection for Australia requires the following packages:

- 2.0 Mbit Digital Trunk Interface (DTI2) package 129
- ISDN Signaling Link Package (ISL) package 145 and
- ISDN Semi Permanent Connection for Australia (ISPC) package 313

The following packages are required for Asia-Pacific ISDN Primary Rate:

- Integrated Digital Services Network (ISDN) package 145
- 2.0 Mbit Primary Rate Interface (PRI2) package 154

- International PRA (IPRA) package 202
- Intelligent Peripheral Equipment (XPE) package 203
- Multi-Purpose Serial Data Link (MSDL) package 222

The following packages are required for an ISPC link to convey D-channel signaling:

- International Supplementary Features (SUPP) package 131 (for SLAVE nodes only)
- Dial Tone Detector (DTD) package 138 (for MASTER nodes only)

Feature implementation

Task summary list

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

1. Table 224: LD 17 - Configure Phantom Loops. on page 490
2. Table 225: LD 73 - Configure Signaling Categories used for ISPC links. on page 490
3. Table 226: LD 16 - Configure Route Data Block for ISPC link. on page 492
4. Table 227: LD 14 - Configure Phantom Trunk Terminal Numbers. on page 494
5. Table 228: LD 11 - Configure Data Interface. on page 495
6. Table 229: LD 13 - Configure Dial Tone Detection on the Master Side. on page 496
7. Table 230: LD 96 - D-Channel Message Monitoring. on page 496

Following subscription to the ISDN Semi Permanent Connections (ISPC) service from a Telecom entity, the overlays listed below are used to configure ISPC links when they are used to convey D-channel signaling or used for ISL calls.

For both cases:

- at least one Phantom DTI2 loop must be defined in LD 17; and
- a specific SICA Table must be defined in LD 73.

ISPC links used for ISL calls:

- Configure a TIE ISL Route in LD 16; and
- For each ISPC Link, a TN on the Phantom DTI2 loop must be defined in LD 14.

ISPC links used to convey D-channel signaling:

- The configuration is asymmetrical. One side of the link is defined as MASTER and the other is defined as SLAVE.
- On the MASTER SIDE:
  - Configure Dial Tone Detection Feature in LD 13;
  - Configure Direct Inward Dialing route (non-ISL) in LD 16;
  - Configure Meridian Communication Adaptor or Meridian Communication Unit in LD 11; and
  - Define one TN on Phantom DTI2 loop in LD 14
- On the SLAVE SIDE:
  - Configure Direct Inward Dialing route (non-ISL) in LD 16;
  - Configure Data Interface in LD 11; and
  - Define one TN on the Phantom DTI2 loop in LD 14.

It is assumed that the ISL D-channel has been configured (LD 16). The following settings are recommended: RS-232 for 19.2 Kbit/s and below, HS (RS-422) for speeds above 19.2 Kbit/s, MCA/MCU configured as 64 Kbit/s and DTE device configuration.

Note:

ISPC uses a new set of commands in LD 96 to monitor the SLAVE SIDE. These new commands are STAT ISPC lp ch, EST ISPC lp ch and RLS ISPC lp ch. Additionally, in LD 60, the STAT command identifies the channel used as ISPC link to a PBX by indicating the ISPC mnemonic.

Table 224: LD 17 - Configure Phantom Loops.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	CEQU	Change Common Equipment Parameters.
...		
- DTI2	xx	2.0 Mbit Digital Trunk Interface or DTI loop number. Precede loop with an "N" to configure this loop as a phantom loop for trunks.

Table 225: LD 73 - Configure Signaling Categories used for ISPC links.

Prompt	Response	Description
REQ	NEW	Create new data block.
TYPE	DTI2	2.0 Mbit DTI data block.

Prompt	Response	Description
FEAT	ABCD	ABCD Signaling category feature.
SICA	2 - 16	Signaling Category table number.
DFLT	1	Default signaling category used for default tables. For phantom DTI2 trunks choose Table 1 as default table.
Prompts for Incoming/Outgoing Calls.		
IDLE (S)	1001	Idle (sender).
IDLE (R)	1001	Idle (receiver).
Incoming Calls		
...		
SEZA (S)	N	Seize signal acknowledgment sent. For phantom DTI2 trunks, SEZA (S) is not configured.
...		
E CONN (S)	(1001)	Connect send. CONN (S) must be the same as IDLE (S) for phantom DTI2 trunks.
CONN (R)	(1001)	Connect received. E CONN (R) must be the same as IDLE (R) for phantom DTI2 trunks.
...		
C CLRB (S)	N	Clear Back send signal. For phantom DTI2 trunks, there is no C CLRB signal.
...		
Outgoing Calls		
...		
SEZA (R)	N	Seize signal acknowledgment received. For phantom DTI2 trunks, SEZA (R) is not configured.
...		
CONN (S)	1001	Connect send. E CONN (S) must be the same as IDLE (S) for phantom DTI2 trunks.
E CONN (R)	1001	Connect received. E CONN (R) must be the same as IDLE (R) for phantom DTI2 trunks.
...		
C CLRB (R)	N	Clear Back.

Table 226: LD 16 - Configure Route Data Block for ISPC link.

Prompt	Response	Description
REQ	NEW	New route.
TYPE	RDB	Route Data Block.
CUST	0-99	Customer number as defined in LD 15.
ROUT	0-511	Route number for Large System and CS 1000E system.
The following prompts must be answered when ISPC link is used for ISL calls		
TKTP	TIE	TIE Trunk Route Type. TIE trunks must be used for ISPC links used for ISL calls.
DTRK	YES	Digital Trunk Route.
- DGTP	DTI2	2.0 Mbit/s Digital Trunk Interface. ISL route that uses Phantom Trunk TNs must be configured as DTI2 routes.
ISDN	YES	Integrated Services Digital Network. ISDN must be set to YES for ISL signaling to control ISPC link usage.
- MODE	ISLD	Route uses ISDN Signaling Link. ISDN is allowed only if ISDN = YES and ISL package 147 is equipped.
- DCH	1 - 63	D-channel Port Number that controls ISPC links usage.
- IFC	SL1	Interface type for PRI route. Connection is direct to another system.
- PNI	(0) - 32700	Private Network Identifier. Each customer data block must use a unique PNI when equipped with multi-customer option. It must match the PNI in the far end Customer Data Block to support Network Ring Again, Network ACD and Network Message Service features.
...		
ICOG	IAO ICT OGT	Incoming and outgoing trunk, Incoming trunk only, or Outgoing trunk only.
...		
ACOD	xxxx	Trunk Route Access Code. ACOD must not conflict with numbering plan. The code entered can be up to four digits or up to seven digits, if the DNXP package 150 is equipped.
The following prompts must be answered when ISPC link is used to convey D-channel signaling		
TKTP	DID	Trunk Route Type. DID must be used when associated with an ISPC link used to convey D-channel signaling.
...		
DTRK	YES	Digital Trunk Route.

Prompt	Response	Description
...		
- DGTP	DTI2	2.0 Mbit Digital Trunk Interface.
ISDN	NO	Integrated Services Digital Network PRI option.
- DSEL	DTA	Data selection. DTA is configured as "data only" when ISPC link is used to convey D-channel signaling.
ICOG	IAO OGT ICT	Incoming and outgoing trunk, Outgoing trunk only, or Incoming trunk only. ICOG configuration is dependent on the side of the ISPC link with which the route is to be associated. On the MASTER side (SMAS = YES in LD 14), ICOG = OGT or IAO. On the SLAVE side (SMAS = NO in LD 14), ICOG = ICT or IAO.
...		
ACOD	xxxx	Trunk Route Access Code. ACOD must not conflict with numbering plan. The code entered can be up to four digits or up to seven digits, if the DNXP package 150 is equipped.
...		
CNTL	YES	Change to Controls or Timers. CNTL must to answered to trigger DTD prompt.
...		
- DTD	YES	Dial Tone Detection provided. On MASTER side of ISPC link (SMAS = YES in LD 14) DTD = YES.
...		
NEDC	ETH	Near End Disconnect Control.
FEDC	ETH	Far End Disconnect Control.
CPDC	NO	The system is not the only controlling party on incoming calls.
DLTN	NO	Dial Tone on originating calls provided on this route. On SLAVE side of ISPC link (SMAS = NO in LD 14) DLTN = YES.
...		
PRDL	BSY	Busy signal is sent on time-out. The PRDL prompt must be set to "BSY" in order to disconnect ISPC phantom DTI2 trunks when an incomplete or invalid DN is received on the ISPC SLAVE D-channel.

Note:

ISPC Reference Numbers supplied by Telecom entity must be associated with Phantom Trunk TNs defined in LD 14.

Table 227: LD 14 - Configure Phantom Trunk Terminal Numbers.

Prompt	Response	Description
REQ	CHG	Change existing data block.
	END	Exit overlay program.
	NEW x	Add new data block (1-255).
	OUT x	Remove data block (1-255).
The following answers are required when ISPC link is used for ISL calls.		
TYPE	TIE	Trunk type.
TN	l s c u	Terminal Number for Large System and CS 1000E.
...		
SICA	1 - 16	Signaling Category Table Number configured in LD 73.
RTMB		Route number and Member Number
	0-511 1-4000	Range for Large System and CS 1000E system.
CHID	1-382	ISL Channel Identification for trunk. A different channel is required for each phantom TN used for the ISPC links. The same Channel ID must be configured for the ISPC reference number on both systems linked by the connection.
TKID	nnnnnnn	Trunk Identifier.
SREF	XXXXXXX	Semi Permanent Connection Reference Number as defined by the Telecom administration at subscription. Reference number can be a maximum of seven digits. SREF is prompted only when Trunk TN is on a phantom loop.
SDCH	NO	The ISPC link is not used to convey D-channel signaling.
The following answers are required when ISPC link is used to convey D-channel signaling		
TYPE	DID	Trunk type.
SICA	1 - 16	Signaling Category Table Number configured in LD 73.
RTMB		Route number and Member Number
	0-511 1-4000	Range for Large System and CS 1000E system.
CLS	DTN	Digitone Class of Service. CLS = DTN for ISPC link to convey D-channel signaling.

Prompt	Response	Description
SREF	yy..yy	ISPC Reference Numbers supplied by Telecom entity to a maximum of 7 digits.
SDCH	YES	ISPC link used to convey D-channel signaling. For SMAS prompt to be displayed, SDCH = YES.
SMAS	(NO) YES	Data Interface over the ISPC link is MASTER or SLAVE. YES = ISPC link is on MASTER side. NO = ISPC link is on SLAVE side. One side of the ISPC link must be configured with SMAS = YES and the other side with SMAS = NO.

Table 228: LD 11 - Configure Data Interface.

Prompt	Response	Description
REQ	NEW	Add new data.
TYPE	a...a	Meridian Modular Telephones (MMT) types equipped with Meridian Communication Adapter (MCA) or Meridian Communication Unit (MCU). Where a...a = 2006, 2008, 2216, 2616 or MCU.
...		
CLS	DTA	Class of Service Data Terminal.
...		
DTAO	MCA	Data Option software for Meridian Communication Adaptor (MCA) that determines the downloaded data, system and operating parameters. The DTAO prompt appears for all Meridian Modular Telephone data ports except MCU.
...		
OPE	YES	Change data port parameters.
- PSDS	NO	Public Switched Data Service (PSDS) option.
- TRAN	SYN	Synchronous data transmission.
...		
The following is for the MCA/MCU used on the MASTER side of the ISPC link.		
- HOT	ON	Enable Hotline for data port.
...		
- BAUD	0-(11)-12	Data rate in bit/s for the data port. Baud rate 12 (64 Kbit/s) is recommended.
...		
KEY	0 SCR yyyy	Single Call Ringing key used by Hotline, where: 0 = Key number, SCR = Single Call Ringing, yyyy = DN.

Prompt	Response	Description
KEY	3 ADL yy z...z	Autodial key used by hotline feature, where: 3 = Key number, ADL = Autodial, yy = maximum DN length and z....z = DN preceded by route access code of the Data Interface on the ISPC SLAVE side of link.
The following is for the MCA/MCU used on the SLAVE side of the ISPC link.		
- HOT	OFF	Hotline feature disabled.
- AUT	ON	Autoanswer.
- BAUD	12	Data rate for 64 Kbit/s.
...		
KEY	0 SCR yyyy	Single Call Ringing key, where: 0 = Key number, SCR = Single Call Ringing, yyyy = DN. The Autodial Key on the MASTER side must be configured to access this DN.

Table 229: LD 13 - Configure Dial Tone Detection on the Master Side.

Prompt	Response	Description
REQ	NEW	Add data block for Digitone Receiver.
TYPE	aaa	Type of data block, where aaa = DTD (Dial Tone Detector) or XTD (Extended Dial Tone Detector).
TN	l s c u	Terminal Number for Large System and CS 1000E.

Table 230: LD 96 - D-Channel Message Monitoring.

Command	Description
STAT ISPC l ch	Get status of data interface establishment process at an ISPC SLAVE side of an ISPC link (where l = loop and ch = channel) which is configured to convey D-channel signaling. The status can be: 1. DISABLED = ISPC link is disabled 2. AWAITING ESTABLISHMENT = the data interface establishment process is waiting to receive DTMF digits. While waiting, dial tone is provided to the ISPC link at the SLAVE side. 3. RELEASED = the data interface establishment process is stopped because either an "RLS ISPC" command has been entered or because the maximum number of tries has been reached.

Command	Description
	4. ESTABLISHED = data interfaces at both ends of the ISPC link are established. 5. AWAITING RELEASE = the ISPC link is being disconnected. Once disconnected, the link will transition to the "RELEASED" state. The mode can be: 1. MODE MANUAL N = the data interface establishment process is running with up to "N" number of tries 2. MODE AUTOMATIC = the system provides dial tone to the ISPC slave D-channel.
	Establish data interface process at the ISPC SLAVE side of an ISPC link at loop (l) and channel (ch), where "N" = the "number of tries" counter. The actual range of N = 0-(1)-255. When N = 0, the mode is set to AUTOMATIC. This mode requires one TDS slot and one DTR unit. This mode is not recommended for use on systems loaded with heavy traffic. When N > 0, the process is activated for a period of up to N * 30 minutes. The available range goes from 30 minutes (N = 1) to 128.5 hours or 5 days (N = 255). If no value is entered for N, it defaults to a value of 1. When N = 1, one attempt will be made to establish the data interface process before the ISPC slave D-channel is released.
RLS ISPC l ch	Stop the data interface establishment process. When this command is entered, the mode becomes MANUAL and the number of attempts to be performed = 0.

Feature operation

No specific operating procedures are required to use this feature.

Operating ISDN Semi Permanent Connections for Australia

Perform the following steps to operate ISDN Semi Permanent Connections for Australia:

1. Subscribe for ISDN Semi Permanent Connections (ISPC) service from Telecom entity.
2. Install hardware for ISPC links usage (if required).
3. Configure systems for ISPC links usage.

4. Wait for ISPC links to be established.
5. Wait for ISL D-channel to be established (if using).
6. Perform callthrough established ISPC links.

Chapter 39: ISDN Signaling Link

Contents

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

Feature description

ISDN Signaling Link (ISL) provides the capability to replace both digital and analog conventional trunk signaling with out-of-band ISDN D-channel signaling.

When the D-channel is inactive, ISL trunk calls can automatically revert to conventional trunk signaling. This applies to both primary and backup D-channels.

ISL supports TIE and ISA trunk types with system to system connectivity.

The TIE lines and trunk used for the D-channel can be leased from the Central Office (CO). With leased lines, the function of the CO is simply to provide the trunk wire facilities between systems for circuit switched connections.

The ISDN Signaling Link supports a uniform trunk hardware configuration. This configuration uses the QPC237 4-wire E&M TIE trunk card in place of other analog trunk cards (such as the QPC71 TIE trunk card or the QPC72 DID/DOD trunk card).

Note:

If the QPC237 TIE trunk card is not used, backup D-channel is not available.

There are two modes of ISL operation:

1. Shared mode: The DCHI supports ISDN PRI signaling and ISL trunks. The configuration is basically the same as the PRI D-channel, with the D-channel also supporting ISL trunks.
2. Dedicated mode: The DCHI does not support ISDN PRI signaling. The DCHI is reserved for ISL use. The D-channel can communicate with the far end by means of a dedicated leased line, dial-up modem, or DTI/PRI trunk.

ISDN Signaling Link (ISL) supports Calling Line Identification (CLID) and CLID in Call Detail Recording (CDR), Electronic Switched Network (ESN) interworking, and Network Ring Again (NRAG) applications.

For ISL to use an ISA route, the ISA route must be identified as ISL in LD16. This is done by assigning the MODE prompt to ISLD.

When an ISL call utilizes a Call-by-Call Service, the D-channel associated with the ISA route for the Call-by-Call Service is used for signaling.

Note:

The High Speed Data Module (HSDM) is used in place of both the Asynchronous/ Synchronous Interface Module (ASIM) and the Add-On Data Module (ADM).

Reverting to conventional trunk signaling

This feature handles ISL trunk calls by reverting to conventional trunk signaling when the D-channel is inoperative. This applies to the primary and backup D-channels as well as ISA.

When a primary or backup D-channel goes 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 means of D-channel signaling are dropped. The user must reinitiate 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 backup D-channel re-establishes:

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

When a D-channel re-establishes that does not have a backup:

- 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 that the call was setup with.

When the ISL D-channel goes down, the following scenario occurs. Since ISL trunks can convert to conventional trunk signaling, the ISL route (which is associated with a D-channel) is bypassed using conventional trunk signaling.

This package allows for the configuration of the NTAK02 as an ISDN Signaling Link (ISL), and for the transport of CLID and ESN/ISDN features.

This link provides ISDN Feature Networking Services to locations in a customer's network where a DTI/PRI link is not available or not economically justified.

The ISL uses a dedicated channel at speeds between 2.4 to 64 Kbit/s to establish the ISDN "D" channel for transmission of the ISDN Q.931 signaling protocol. The D-channel information is transported to the far end using one of the following methods:

- PRI channel in the case of ISL shared mode (in the same manner as PRI)
- data adapter or modem in the case of ISL dedicated mode

The associated voice or data is sent over the existing conventional analog or digital trunks.

Note:

Auto recovery for ISL TIE trunks to revert back to in-band signaling upon failure of the D-channel Handler Interface is available on systems.

Operating parameters

There are no operating parameters associated with this feature.

Feature interactions

ISDN QSIG Basic Call

ISDN Signaling Link (ISL) on QSIG operation is not supported on ISDN BRI trunking.

ISDN Semi Permanent Connections for Australia

Redirection or optimization of calls will not release established ISDN Semi Permanent Connections links.

Feature packaging

This feature requires the ISDN Signaling Link (ISL) package 147, which is dependant on ISDN package 145.

Feature implementation

Task summary list

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

1. Table 231: LD 17 - Configure the D-Channel for ISL. on page 503
2. Table 232: LD 15 - Define a PRI customer and networking data. on page 504
3. Table 233: LD 16 - Enable ISL for each route. on page 505
4. Table 234: LD 14 - Assign a channel identification to each trunk with the ISL option. on page 506

Table 231: LD 17 - Configure the D-Channel for ISL.

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	CFN	Configuration Record.
ADAN	NEW DCH x	Add a primary D-channel (can also CHG and OUT DCH).
CTYP	DCHI MSDL	Card type.
DNUM	xx	Device number: physical port (odd) for D-channel on DCH, physical card address for MSDL.
- PORT	0-15	Port number.
USR	PRI ISLD SHA	Primary Rate Interface Integrated Services Signaling Link Dedicated. Shared mode.
- ISLM	1-382	Number of ISL trunks controlled by the D-Channel.
DCHL	xx	PRI loop number for DCHI or MSDL.
OTBF	1-(32)-127	Number of output request buffers.
- BPS	xxxx	Baud rate for ISL D-channel on MSDL port (default 64000).
- PARM	R232 R422 DCE DTE	ISL D-channel interface and transmission mode (MSDL port only).
PRI	xxx yy	Additional PRI loops using the same D-channel, and interface ID.
DRAT	(56K) 64KC 64KI	D-channel transmission. 56 Kbit/s, default when LCMT (LD17) is AMI. 64 Kbit/s clear, default when LCMT (LD17) is B8S. 64 Kbit/s inverted HDLC (64 Kbit/s restricted. DRAT is not prompted when configuring ISL D-channels because the speed is controlled by the modem baud rate.
IFC	aaa	Interface type: (D100), D250, ESS4, ESS5, SL1, S100.
SIDE	(USR) NET	System node type.
RLS	xx	Release ID of the switch at the far end of the D-Channel.
RCAP	aaa	Remote D-channel capabilities: MSL, NCT, ND1, ND2, RVQ.
- CLOK	(EXT) INT	Internal or external clock on ISL D-channels.
LAPD	(NO) YES	D-channel LAPD parameters.
- T23	1-(20)-31	Interface guard timer (DCHI only)

Prompt	Response	Description
- T200	2-(3)-40	Retransmission timer
- N200	1-(3)-8	Maximum number of retransmissions
- N201	4-(260)	Maximum number of octets in information field
- T203	2-(10)-40	Maximum time (in seconds) without frames being exchanged
- K	1-(7)	Maximum number of outstanding frames
ADAN	NEW BDCH x	Add a backup D-channel. x = 0-63.
PDCH	0-63	Primary D-channel
CTYP	DCHI MSDL	Card type.
DNUM	0-15	Physical port (odd) for D-channel, card number for MSDL.
- PORT	0-3	Port number on MSDL card.
BCHL	xxx	PRI loop number for backup D-channel.
RCVP	(NO) YES	Auto-recovery to primary D-channel option.

Table 232: LD 15 - Define a PRI customer and networking data.

Prompt	Response	Description
REQ	NEW	Add new data.
	CHG	Change existing data.
TYPE	NET	Networking data.
CUST	0-99	Customer number as defined in LD 15.
- ISDN	YES	Integrated Services Digital Network.
- PNI	1-32700	Private Network Identifier (for CLID).
- PFX1	xxxx	Prefix (area) code for International PRA (for CLID).
- PFX2	xxxx	Central Office Prefix for IPRA (for CLID).
- HNPA	100-999	Home Number Plan Area code (for CLID).
- HNXX	100-999	Prefix for Central Office (for CLID).
- HLOC	100-9999	Home Location Code (ESN) (for CLID).
- LSC	xxxx	Local steering code (for CLID).
- CNTP	(PDN) LDN	Default for Calling Line ID (CLID).
- RCNT	0-(5)	Maximum inter-node hops in a network redirection call.

Table 233: LD 16 - Enable ISL for each route.

Prompt	Response	Description
REQ	NEW	Add a PRI trunk route.
TYPE	RDB	Route Data Block.
CUST	0-99	Customer number as defined in LD 15.
ROUT	0-511	Route number for Large System and CS 1000E system.
TKTP	ISL	Trunk route type.
ESN	(NO) YES	ESN signaling.
BRIP	(NO) YES	ISDN BRI packet handler route.
ISDN	(NO) YES	ISDN PRI option.
- MODE	ISLD	Mode of D-Channel that controls the route.
- DCHI	1-15	DCHI port number Prompted if Mode = ISLD.
- PNI	0-32700	Customer Private ID (unique to a customer).
- IFC	aaa	Interface machine type for this PRI route.
- NCNA	YES	Network call name allowed.
- NCRD	YES	Network Call Redirection.
- TRO	(NO) YES	Trunk optimization allowed (denied) on the route.
- NSF	(NO) YES	Network Service Facility.
- TIER	xxx	Incoming TIE route number.
- CHTY	(BCH) ABCH	Signaling type for B-channel digital routes.
- CTYP	aaa	Call type for outgoing direct dialed TIE route.
- INAC	(NO) YES	Insert ESN access code to incoming private network call.
- NCOS	(0)-99	Network Call Of Service.
- CLS	aaa	Class of service restriction for TIE route.
- TGAR	0-31	Trunk group access restrictions.
- IEC	000-999	Inter exchange carrier ID For Large Systems.
PTYP	aaa	Port type at far end.
ICOG	IAO ICT OGT	Incoming and Outgoing trunk. Incoming only trunk. Outgoing only trunk.
SIGO	aaa	Signaling arrangement.
NEDC	ORG ETH	Near end disconnect control. Originating end control. Either end control.
FEDC	(ORG)	Far end disconnect control.

Prompt	Response	Description
	ETH FEC JNT	Originating end control. Either end control. Far end control. Joint control.
CPDC	(NO) YES	The system is the only controlling party on incoming calls.
DLTN	(NO) YES	Provide dial tone to the far-end.
PSEL	(DMDM) TLNK	T-link or DM-DM protocol selection.
EQAR	(NO) YES	Enable Equal Access on this route.
- GCR	(YES) NO	Use General Carrier Restriction for Equal Access calls.
- NTOL	(DENY) ALOW	Allow or deny North American toll calls (1+ calls).
- ITOL	(DENY) ALOW	Allow or deny international toll calls (011+ calls).
- SCR	(NO) YES	Use Selective Carrier Restriction for Equal Access calls.
AUTH	(NO) YES	Authcode to be prompted for incoming TIE callers.

Table 234: LD 14 - Assign a channel identification to each trunk with the ISL option.

Prompt	Response	Description
REQ	NEW	Add new data.
TYPE	TIE	Trunk type allowed.
TN	l s c u	Terminal Number for Large System and CS 1000E
RTMB		Route number and Member Number
	0-511 1-4000	Range for Large System and CS 1000E system.
CHID	1-382	Channel identifier for ISL channels (remove with Xnn) — must be coordinated with far end (no default value).

Feature operation

No specific operating procedures are required to use this feature.

Index

T

W

X