

CHAPTER

Network Features

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The Cisco BTS 10200 Softswitch supports network features as described in the following sections:

- Interoperability
- New Network Features for Release 4.5.x
- Numbering Plans and Dialing Procedures—This section includes information on digit manipulation, E.164 dialing plan, casual dialing (dial around), dial 1 options, directory services, easily recognizable codes, Information service calls (900 and 976), n11 support (211, 311, 411, 511, 611, 711, 811). and NRUF reporting.
- Emergency Services (911)
- Lawful Intercept Interface
- Operator Services—This section includes information on Busy Line Verification and Operator Interrupt.
- Network Services—This section includes information on 8XX (Toll-Free Calling), Active Call Information Display, Alerting Notification to Third-Party Feature Server, Calling Party Number Options for Outgoing SETUP Messages, Dialing Parity (IntraLATA Toll Presubscription), Local Number Portability (LNP), Split-NPA, and T.38 Fax Relay.
- Trunk and Line Testing

In general, Cisco BTS 10200 Softswitch features delivered via gateway clients behave identically to their public switched telephone network (PSTN) counterparts.



For information on subscriber features, see Chapter 2, "Subscriber Features." For information on outgoing call restriction options (Class of Service and Outgoing Call Barring) see Chapter 3, "Class of Service Restrictions and Outgoing Call Barring Features."

Some features can be accessed and controlled by the subscriber using a handset and vertical service codes (VSCs). VSCs are provisionable by the service provider (any valid unique ASCII string up to five characters long), and the customary values are country specific. The VSC values used throughout this chapter are for illustration purposes. For convenience, some VSC values are preprovisioned in the Cisco BTS 10200 Softswitch.

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The valid formats for VSC ASCII strings are listed in the *VSC table specification* in the *Cisco BTS 10200* Softswitch Command Line Interface Reference Guide. The preprovisioned VSC values are listed in the Vertical Service Code appendix of the same document. To provision VSCs, see the VSC provisioning procedure in the Cisco BTS 10200 Softswitch Provisioning Guide.

Typically, the system responds to user handset actions by providing an appropriate announcement. However, if an announcement is not provisioned or cannot be played, an alternate tone (for example, reorder tone) is played. Announcements are listed in the *Cisco BTS 10200 Softswitch Provisioning Guide*, and tones are listed in the *Cisco BTS 10200 Softswitch Operations and Maintenance Guide*.

Interoperability

The Cisco BTS 10200 Softswitch interworks with a wide range of network elements (NEs), but there are certain limitations. Cisco recommends that you keep the following caution in mind as you prepare to purchase and use NEs for your network.



Some features involve the use of other NEs deployed in the service provider network, for example, gateways and media servers. See the Component Interoperability section of the *Release Notes* for a complete list of the specific peripheral platforms, functions, and software loads that have been used in system testing for interoperability with the Cisco BTS 10200 Softswitch Release 4.5.x software. Earlier or later releases of platforms of the *Release Notes* certifies only that the required interoperation of these platforms, the functions listed, and the protocols listed have been successfully tested with the Cisco BTS 10200 Softswitch.

New Network Features for Release 4.5.x

The following network features are new for Release 4.5.x:

- Dial 1 Options for Local, Toll, and InterLATA Calls
- NRUF Reporting for NANPA Audit Support
- Active Call Information Display
- Alerting Notification to Third-Party Feature Server
- Testing Capability for 911 FGD-OS Trunks

The following network features were enhanced in Release 4.5.x:

- Emergency Services (911)
- Calling Party Number Options for Outgoing SETUP Messages
- T.38 Fax Relay
- Network Loopback Test for NCS/MGCP Subscriber Endpoints
- 1xx Test Line Support

In addition, information has been updated for Release 4.5.x in several other sections of this chapter.

Numbering Plans and Dialing Procedures

The Cisco BTS 10200 Softswitch supports the numbering plans and dialing procedures listed in Table 1-1. These features are described in the sections that follow.

Note

For additional details on the rules used in the numbering plans and dialing procedures, see the *Cisco BTS 10200 Softswitch Dial Plan Guide*.

Feature Description	Reference	
Digit Manipulation		
E.164 Dialing Plan Implementation	ITU-T Recommendation E.164	
Casual Dialing (Dial Around)		
Dial 1 Options for Local, Toll, and InterLATA Calls		
Directory Services (411, 555-1212, 0+ Listing Services)	GR-532-CORE FSD-30-17-0000	
Easily Recognizable Codes	GR-2892-CORE SR-2275, Sec. 3.3	
Information Service Calls (900 and 976)		
n11 support (211, 311, 411, 511, 611, 711, 811)	GR-532-CORE FSD-30-16-0000	
NRUF Reporting for NANPA Audit Support		

Table 1-1 Support for Numbering Plans and Dialing Procedures

Digit Manipulation

The Digit Manipulation (DIGMAN) feature provides the ability to modify both calling number and called number for both incoming and outgoing calls within the Cisco BTS 10200 Softswitch.



The calling party number is also known as ANI (automatic number identification). The called party number is also known as DNIS (dialed number identification service).

In addition to modifying the calling number and the called number, the digit manipulation tables can be used to modify the nature of address (NOA) of ANI and/or DNIS numbers. This feature provides the following benefits in the service provider network:

- Dial plans for both North American Numbering Plan (NANP) and ITU-T E.164 numbering plan
- Flexible call processing
- ANI- or DNIS-based routing

For additional standards information, see the following industry sources:

- NANP—See http://www.nanpa.com
- ITU-T Recommendation E.164, The International Public Telecommunication Numbering Plan

The Cisco BTS 10200 Softswitch performs digit manipulation by matching and replacing digits in the digit string that is being processed.

E.164 Dialing Plan Implementation

The Cisco BTS 10200 Softswitch implements a dialing plan based on ITU-T Recommendation E.164, *The International Public Telecommunication Numbering Plan*, a standard for numbering and routing. This dialing plan uses a generic numbering scheme for number evaluation. The Cisco BTS 10200 Softswitch performs digit manipulation on ANI data of the calling party, and on DNIS data of the called party.

National Number

In the E.164 numbering scheme, there are three parts to any national number (number that terminates within the country):

• National destination code (NDC)—A region of the country (1 to 6 digits, typically 3).



Provisioning of the NDC is optional. Some countries do not use NDCs in the national number.

- Exchange code (EC)—An area served by a single central office (CO) switching facility (1 to 6 digits, typically 4).
- Dialing number (DN)—The specific digits that identify a subscriber line (1 to 4 digits, typically 4).



The combination [EC + DN] is called the subscriber number (SN). The combination [NDC + EC + DN], or [NDC + SN], is called the national number (NN).



[NDC + EC + DN] is interpreted as [NPA + NXX + XXXX] in NANP, where NPA (numbering plan area) = 200 to 999, NXX (office code) = 200 to 999, and XXXX = 0000 to 9999. The Cisco BTS 10200 Softswitch applies the NANP interpretation if the NANP-DIAL-PLAN flag is set to Y (yes) in the DIAL-PLAN-PROFILE table.

A user originates a call by dialing as follows:

- To place a call to a phone in the same EC (served by the same CO), dial the SN. In most cases, this is considered a local call.
- To place a call to a phone in another EC, but within the same region (same NDC), dial the SN. In most cases, this is considered a local toll call.
- To place a call to a phone in another region (different NDC), dial the national (trunk) prefix and the NN. The national prefix varies from country to country. In most cases, this type of call is considered a national toll call.

Examples of national prefixes include:

- 0 in China
- 1 and 0 within NANP
- 9 in Finland and Spain

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– 16 in France

Note

For countries that do not use NDCs, it is not necessary to provision any value for the NDC parameter in the Cisco BTS 10200 Softswitch.

International Number

The international number is the number dialed from one country to another to reach a subscriber. Each country is assigned a country code (CC). The international number is the combination [CC + NN], or [CC + NCD + EC + DN]. Table 1-2 lists several examples.

Country	City	CC	NDC	EC	DN Group	Complete International Number
Belgium	Bruxelles	32	02	123	XXXX	32-02-123-xxxx
China	Chengdu	86	28	8293	XXXX	86-28-8293-xxxx
Germany	Dusseldorf	49	211	12	XXXX	49-211-12-xxxx
Canada	Montreal	1	514	870	XXXX	1-514-870-xxxx
United Kingdom	London	44	71	248	XXXX	44-71-248-xxxx

Table 1-2Examples of International Numbers

To place a call to a phone in another country, the caller must dial an international prefix and then the international number. Thus, the complete digit string to dial is [international prefix + CC + NN]. The international prefix varies from country to country. Examples of international prefixes include:

• 00 in China

Example of a call from China to Montreal: 00-1-514-870-xxxx

• 011, 01 in NANP

Example of a call from the United States to Bruxelles: 011-32-02-123-xxxx

In some countries, two or more international prefixes may be used

- To reach different groups of countries
- To reach countries within a group

Casual Dialing (Dial Around)

Casual dialing, also known as dial around, specifies whether the carrier supports 101XXXX calls. The digit map CLI command tokens provide the digit pattern. The digit pattern specifies all possible acceptable patterns. An example of a casual digit pattern is 1010321 or 1010220. The digit map table tells the media gateway (MGW) how to collect and report dialed digits to the Call Agent (CA). Subscribers can prefix their toll, interLATA, or international calls with 101XXXX. Casual dialing supports the following casual calls:

- 101XXXX + 0/1 + NPA + NXX-XXXX
- 101XXXX + 0/00

• 101XXXX + 011/01 + CC + NN

Dial 1 Options for Local, Toll, and InterLATA Calls

The service provider can provision the system to control the use of prefix 1 for specific types of calls and for specific subscribers. Local, toll, and interLATA call types can each be independently provisioned in the subscriber-profile table as follows:

- Require that the number be dialed with a prefix 1—If the system is provisioned this way, and the caller attempts to dial the number without using a prefix 1, the system rejects the call and provides an appropriate announcement (Release Code 10).
- Require that the number be dialed without a prefix 1—If the system is provisioned this way, and the caller attempts to dial the number using a prefix 1, the system rejects the call and provides an appropriate announcement (Release Code 9).
- Prefix 1 optional—Allow call processing to proceed whether a prefix 1 is dialed on not.



For service access code (SAC) calls such as 500, 700, 800, and 900, the user must dial the prefix 1. The flags LOCAL-PFX1-OPT, INTERLATA-PFX1-OPT, and TOLL-PFX1-OPT in the Subscriber table do not affect these types of calls.

Note

For a list of the specific provisioning parameters, see the Subscriber Profile table in the *Cisco BTS 10200* Softswitch CLI Guide. For a complete list of release cause codes, see the Appendix of the Cisco BTS 10200 Softswitch Provisioning Guide.

Directory Services (411, 555-1212, 0+ Listing Services)

The Cisco BTS 10200 Softswitch supports the directory services access feature as specified in Telcordia document GR-352-CORE, LSSGR: *Interface To Directory Assistance Systems (FSD 30-17-0000)*.

Directory services allows a subscriber to obtain the listed telephone number for a given name and address. The caller dials a specific service number to reach directory services, also referred to as directory assistance (DA). When a subscriber dials one of the following digit patterns, the Cisco BTS 10200 Softswitch routes the call to the applicable directory services in the PSTN:

- 411 or 555-1212 (DA)
- 1+411, 1+555-1212 (toll DA)
- 1-NPA-555-1212 (mostly for out-of-town/state numbers)
- 1-8XX-555-1212 (toll-free numbers)
- 0+ listing services

The service to the caller can be provided manually by a live operator, automated via a voice or dual tone multifrequency (DTMF) recognition system, or by a combination of these. The volume level from an automated voice-response unit, however, should be comparable to that of a live operator. Different network operators can employ different systems in providing directory services.

A typical directory services request requires that the caller first give the name of the town and city. The caller then provides the name of the person or business that the caller wants to call, including the spelling of unusual names. Finally, the caller states if the request is for residence or business. Additional services include handling multiple requests made during the same call and automatic connection to the person (or business) the caller wants to call.

Easily Recognizable Codes

The Cisco BTS 10200 Softswitch supports selected easily recognizable codes (ERCs) as described in document SR-2275, *Telcordia Notes On the Network*, Section 3.3. The supported ERCs are

- 500 personal communications services (PCS)—See the Alliance for Telecommunications Industry Solutions (ATIS) document INC-95-0407-009, *Personal Communication Services N00NXX. Code* Assignment Guidelines, for a PCS description.
- 700 service access calls (SAC)—Range of codes used by interexchange carriers (IXCs) to provide services on the network.
- Toll-free service call features (8XX)—See the "8XX (Toll-Free Calling)" section on page 1-15 for a description.
- 900/976 information service calls—See the "Information Service Calls (900 and 976)" section on page 1-7 for a description.

Other Telcordia reference documents include:

- SR-2275, Telcordia Notes On the Network
- GR-2892-CORE, Switching and Signaling Generic Requirements for Toll-Free Service Using AIN

Information Service Calls (900 and 976)

Information service calls (ISCs) provide a variety of announcement-related services on a national or local basis. There are two general categories of this service:

- Public announcement services (PAS)—Weather, sports, horoscope, and so forth
- Media-stimulated calling (MSC)—Telephone voting, radio station call-ins, and so forth

National calls are dialed as 1-900-xxx-xxxx and local calls are dialed as NPA-976-xxxx.

n11 support (211, 311, 411, 511, 611, 711, 811)

Note

911 service is covered in the "Emergency Services (911)" section on page 1-9.

This section describes Cisco BTS 10200 Softswitch support for n11 services. The typical relationship between the n11 codes and the nature of dial (NOD) values is as follows. The NOD values are listed in the *Cisco BTS 10200 Softswitch Command Line Reference Guide*.

n11 Code	NOD Value
211	INFO
311	NON-EMG

n11 Code	NOD Value		
411	DA		
511	TRAFFIC		
611	REPAIR		
711	RELAY		
811	BUSINESS		

For additional information on n11 calling, see the following industry documents:

- Telcordia document GR-352-CORE, LSSGR: Service Codes N11 (FSD 30-16-000)
- The NANPA web site, http://www.nanpa.com/number_resource_info

Community Information and Referral Services (211)

The 211 service provides access to information from government service agencies and certain public charity groups.

Nonemergency Services (311)

Some city governments offer 311 service to provide nonemergency information to the community. The caller dials 311 and the Call Agent translates this to the closest nonemergency access office.

The Cisco BTS 10200 Softswitch supports nonemergency services (311) for routing calls to a specified route type and identification. Routes for all nonemergencies (311) are allocated through the destination table by defining the call type (call-type=NON-EMG) and the routing information for the dialed digits.

Directory Assistance (411)

The 411 service provides directory assistance. See the "Directory Services (411, 555-1212, 0+ Listing Services)" section on page 1-6.

Traffic and Transportation Information (511)

The 511 service provides access to information about local traffic conditions.

Repair Service (611)

The 611 service connects to the local telephone repair service (if the service provider offers this service).

Telecommunications Relay Services (711)

The 711 service provides access to telecommunications relay services (TRS).

Local Billing Services (811)

The 811 service connects to the local telephone billing office.

NRUF Reporting for NANPA Audit Support

Numbering Resource Utilization and Forecast (NRUF) reporting provides North American Numbering Plan Administration (NANPA) audit data based on provisioned values in the dn2subscriber table. For FCC-required NANPA audit compliance, the report input is NPANXX. In markets outside of NANPA, the input can be based on either the combination of the national destination code (NDC) and the exchange code (EC), or just the EC.

The data for NRUF reporting is generated based on either the NDC or the EC. The service provider can use the **report dn-summary** command to generate the following reports:

- Report on all DNs belonging to a specific NDC and EC.
- Report on a thousands group within a specific NDC and EC.



For additional details of this feature, see the NRUF chapter of the *Command Line Interface Reference Guide*.

Emergency Services (911)

The Cisco BTS 10200 Softswitch supports emergency services (911) as specified in Telcordia document GR-529-CORE, *LSSGR: Basic 911 Emergency Service (FSD 15-01-0000)*.

Other Telcordia reference documents include

- SR-4163, E9-1-1 Service Description
- GR-350-CORE, E911 Public Safety Answering Point: Interface Between a 1/1A ESS Switch and Customer Premises Equipment

Description

The digit string 911 is typically used in the U.S. Other digit strings are used elsewhere in the world.

Emergency service is a public safety feature providing emergency call routing to a designated Emergency Service Bureau (ESB), normally called the public safety answering point (PSAP) in the United States. The 3-digit 911 number is assigned for public use in many areas of the United States and Canada for reporting an emergency and requesting emergency assistance. Depending on municipal requirements and procedures, an ESB attendant can transfer the call to the proper agency, collect and relay emergency information to the agency, or dispatch emergency aid directly for one or more participating agencies.

911 calls are location dependent and must be selectively routed to the appropriate PSAP depending on where the call originates. The routing process is part of the Enhanced 911 (E911) feature set and works as follows:

- **1.** In the PSTN, the local serving end office routes the call to the designated E911 tandem for that serving area.
- 2. The E911 tandem then routes the call to the proper PSAP.

Once the caller is connected to the PSAP attendant, the PSAP system typically displays the caller's directory number to the PSAP attendant. Additional data (such as the subscriber's name, address and closest emergency response units) may also be retrieved from the local carrier automatic location identification (ALI) database and displayed to the PSAP attendant.



The service provider can provision a flag for each subscriber to specify which number to send with emergency calls—the subscriber directory number or the subscriber billing number.

Special emergency functions can be provided via a channel-associated signaling (CAS) trunking gateway (TGW) that supports ESB trunks or emergency service line (ESL) trunks with MF signaling. Examples of special emergency functions include:

- Operator callback—Allows the PSAP to automatically ring back the caller.
- End-to-end called-party hold—The Cisco BTS 10200 Softswitch keeps the connection active even if the caller goes on hook.
- Operator disconnect—Allows the PSAP to terminate the call even though the caller has not gone on hook.

Important Provisioning Requirements

For service providers in the U.S., it is typical to provision the Destination table with call-type=EMG for the digit string 911, and call-subtype=NONE (default), because 911 is a central dispatch point for all emergency, ambulance, fire, and police calls.



On the Cisco BTS 10200 Softswitch, for a call to be considered an emergency, it must be provisioned as call-type EMG. If you are using separate DNs for ambulance, fire, and police service (typically applies to networks outside the U.S.), Cisco strongly recommends that you provision these as call-type EMG and call-subtype <AMBULANCE or FIRE or POLICE> in the Destination table. This is the only way to be sure that they will be given all the treatment of the EMG call-type.

Depending on the region of the world, the provisionable timers may require different values, or may not be needed, and they can be turned off. The called-party control feature, typically used in the United States, can also be turned off. All other functions of the emergency number are the same as for the 911 feature.

The emergency service feature can be made available to all subscriber lines connected to a Cisco BTS 10200 Softswitch using the default office service ID, or to all subscribers in a specific POP using the office service ID. See the "Office Service ID and Default Office Service ID" section on page 2-124 for a general description of this provisionable service.

Feature Interactions

The following feature interactions apply to emergency calls (call-type=EMG):

- During a 911 call from a subscriber line, the call waiting (CW) and three-way calling (TWC) features are automatically disabled for the subscriber line.
- There is an interaction when a Centrex subscriber invokes call hold (CHD) and places a call to an emergency number:
 - When the emergency operator answers the call, a two-party call is active between the subscriber and the emergency operator. The on-hold party remains on hold.
 - When the subscriber presses the Flash button or hookswitch, a three-way call is established among the subscriber, the emergency operator, and the previously on-hold party.

- It is not possible to place the emergency operator on hold.

Feature Provisioning Commands

Provisioning commands are available in the Cisco BTS 10200 Softswitch Provisioning Guide.

<u>}</u> Tip

To provision this feature, see the 911 provisioning procedure in the Cisco BTS 10200 Softswitch Provisioning Guide.

Lawful Intercept Interface

This section describes the lawful intercept interface supported by the Cisco BTS 10200 Softswitch.

General Description of Lawful Intercept Implementation

The Cisco BTS 10200 Softswitch supports the call data interface and call content function for lawful intercept, along with the provisioning interface required to configure a wiretap. The Cisco BTS 10200 Softswitch provides support for lawful intercept using two different, industry-developed architectures: PacketCable and the Cisco Service Independent Intercept (SII). Depending on their network type, service providers may choose to configure their networks consistent with either of these architectures in their effort to meet their obligations related to lawful intercept. Given the constantly evolving nature of industry-developed standards, service providers must recognize that the features and functionality of the Cisco BTS 10200 Softswitch described below may also be subject to change over time.



Each country controls its own laws applicable to lawful intercept. For example, in the United States, one of the applicable laws is referred to as the Communications Assistance for Law Enforcement Act (CALEA).

Lawful Intercept Provisioning Interface

The Cisco BTS 10200 Softswitch supports a secure provisioning interface to process wiretap requests from law enforcement agencies through a mediation device. The service provider can limit viewing and provisioning of these parameters to selected authorized personnel. The applicable parameters (entered via CLI) include the DN, tap type, and call data channel for data transmission. The tap type specifies whether the tap order is a pen register (outgoing call information), a trap and trace (incoming call information), a pen and trace (incoming and outgoing call information), or an intercept (bidirectional plus the call content).

Lawful Intercept Call Data Interface

The Cisco BTS 10200 Softswitch provides the PacketCable EMS/RADIUS interface for the transmission of call identifying information to the lawful intercept delivery function (DF) server as required by Appendix A, PCES Support, in PKT-SP-EM-I08-040113, *PacketCable Event Messages Specification (EMS)*, January 13, 2004.

Full call-identifying information (call data) is shipped to a DF server from the Cisco BTS 10200 Softswitch for the subject under surveillance for various call types (for example, basic call and call forwarding).

Lawful Intercept Call Content Function

The call content function provides for capturing voice in a replicated Real-Time Transport Protocol (RTP) stream. The Cisco BTS 10200 Softswitch can be configured to operate with simultaneous support for PacketCable intercept and Cisco SII, or with Cisco SII only.

Simultaneous support for PacketCable intercept and Cisco SII is achieved as follows: During the call-setup phase, the Cisco BTS 10200 Softswitch searches for a PacketCable-compliant call-content intercept access point (IAP) in the call path. If the Cisco BTS 10200 Softswitch determines that there is no such IAP available in the call path, it falls back to Cisco SII.



An intercept access point (IAP) is a point within a communication system where some of the communications or call identifying information of an intercept subject's equipment, facilities, and services are accessed.

Additional information about each type of intercept is provided below:

• PacketCable intercept—In PacketCable intercept, a replicated RTP stream is sent to the DF server by an aggregation router or a trunking gateway upon request from the Cisco BTS 10200 Softswitch. The Cisco BTS 10200 Softswitch requests the relevant IAP (aggregation router or trunking gateway) to duplicate and transport the RTP stream to a predefined IP address and port number.

The Cisco BTS 10200 Softswitch uses Common Open Policy Service (COPS) protocol when sending the above request to an aggregation router, and Media Gateway Control Protocol (MGCP) when sending the request to a trunking gateway.

• Cisco Service Independent Intercept—In Cisco SII, a replicated RTP stream is sent to the DF server by an aggregation router or a trunking gateway upon request from the DF server. The DF server uses SNMPv3 as the control protocol to send the intercept request to the appropriate IAP.

Feature Provisioning Commands

Provisioning commands are available in the Cisco BTS 10200 Softswitch Provisioning Guide.

Tip

To provision this feature, see the *CALEA provisioning procedure* in the *Cisco BTS 10200 Softswitch Provisioning Guide*.

Operator Services

The Cisco BTS 10200 Softswitch supports operator services as specified in Telcordia Requirement FR-271, *Operator Services Systems Generic Requirements (OSSGR)*.

Operator services is a call-processing function whereby callers can access either a live operator or an automated function to complete calls or gain access to information. The service provider can provide this feature or outsource it to a third-party vendor. Some additional functions accomplished by operator services include automatic call distribution, billing detail recording, and information retrieval.

This section includes the following additional topics:

- Numbers Used to Access Operator Services, page 1-13
- Types of Services, page 1-13
- Busy Line Verification (BLV) and Operator Interrupt (OI) Services, page 1-14

Numbers Used to Access Operator Services

The following numbers are commonly used to access operator services:

- 0—Local operator support
- 00—Operator support outside the "local" calling area, using a presubscribed interexchange carrier (PIC)
- 0+ area code and number—Operator support when the destination number is known (that is, for collect calls, calling card calls, person-to-person calls, and so forth, using PIC
- CAC+0+—Operator services, using a dialed carrier access code (CAC)
- 01+CC+NN—International operator services, using PIC
- CAC+01+CC+NN—International operator services, using a dialed CAC

Types of Services

Operator services provided to callers typically include:

- Assistance
- General information
- Directory assistance
- Dialing instructions
- Rate information
- Credit recording
- Trouble reporting
- Call completion
- Alternate billing services (ABS)
- Calling card calls
- Collect calls
- Third-number calls
- Handling options
- Person-to-person
- Conference calls
- Call transfer
- Real-time rating
- Rate quotes
- Time and charges

Notify

Busy Line Verification (BLV) and Operator Interrupt (OI) Services

This section describes busy line verification (BLV) and operator interrupt (OI) services. OI is also referred to as emergency interrupt (EI). BLV and OI services are based on GR-1176 (FSD 80-01-0300), *Busy Line Verification*, part of Telcordia OSSGR requirements (FR-271).

Description and Operation

BLV service permits the user to obtain operator assistance to determine if a called line is in use. The user dials 0, waits for the operator to pick up the line, and requests BLV service. OI service permits the operator to speak directly with the busy party. The service provider can deny BLV service to any subscriber by setting type=denied for fname=BLV in the subscriber-feature-data table (see the BLV provisioning link listed below). Note that denying BLV also denies OI.

BLV and OI services work as follows:

- 1. The user calls the operator and requests BLV service regarding a specific called line.
- 2. The operator provides the BLV service.
- 3. For OI, the operator interrupts the conversation in progress and relays a message.
- 4. If the interrupted party at the called line is willing to hang up, they do so.
- 5. The user can originate a new call to the called DN.



Note At the user's request, the operator has the option to directly connect the user to the called line.

The BLV feature can be made available to all subscriber lines connected to a Cisco BTS 10200 Softswitch using the default office service ID, or to all subscribers in a specific POP using the office service ID. See the "Office Service ID and Default Office Service ID" section on page 2-124 for a general description of this provisionable service.

Feature Interactions

The following feature interactions are applicable to the BLV and OI services:

- When the operator attempts BLV, if the verified party is engaged in a call and has features currently invoked, the operator might receive a busy tone and might not be able to perform an interrupt on the call. In this section, "currently invoked" means that another feature has already been triggered in the call. There are a few exceptions, such as Cancel Call Waiting (CCW) and Do Not Disturb (DND); for example, BLV can be successfully performed even if CCW or DND is currently invoked on the call.
- If the verified party (terminating subscriber) has call forwarding unconditional (CFU) activated, the operator will receive a busy tone and will not be able to perform an interrupt on the call.



To provision this feature, see the *BLV provisioning procedure* in the *Cisco BTS 10200 Softswitch Provisioning Guide.*

Network Services

The Cisco BTS 10200 Softswitch supports the network services listed in Table 1-3.

Table 1-3 Support for Network Services

Feature Description	Reference	
8XX (Toll-Free Calling)	SR-2275, Sec. 14.6	
Active Call Information Display		
Alerting Notification to Third-Party Feature Server		
Calling Party Number Options for Outgoing SETUP Messages		
Dialing Parity (IntraLATA Toll Presubscription)	FSD-20-24-0040 TR-TSY-000693	
Local Number Portability (LNP)	ATIS/T1S1 T1.TRQ-02-2001	
Split-NPA	INC97-0404-016	
T.38 Fax Relay	IETF RFC 2833 ITU-T Recommendation T.38	

8XX (Toll-Free Calling)

The purpose of the toll-free feature is to have the called party, rather than the calling party, charged for the call. These calls are prefixed with the 1+8XX service access codes. The seven digits following the 8XX codes are used for routing the call. For an inbound/outbound 8XX call, the Cisco BTS 10200 Softswitch checks the local toll-free database first. If the corresponding DN is not found in the local toll-free database, the system sends a query to the service control point (SCP) to request the corresponding DN.

All aspects of toll-free calling are transparent to the caller. A caller expects to dial 1-8XX-NXX-XXXX to reach the desired destination. The company that translates the number to a specific DN, and the company that routes the call, must appear transparent to callers. Most callers are not aware that their dialed 8XX number is changed into a specific DN. What matters to the callers is that they reach what they perceive to be the called number, and they are not billed for the call.

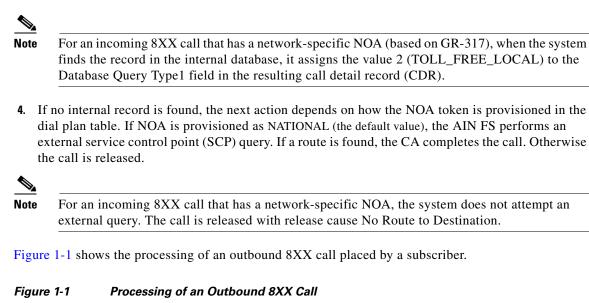


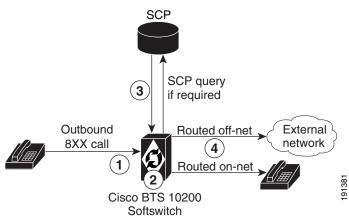
These toll-free (8XX) features can be made available to all subscriber lines connected to a Cisco BTS 10200 Softswitch using the default office service ID, or to all subscribers in a specific POP using the office service ID. See the "Office Service ID and Default Office Service ID" section on page 2-124 for a general description of this provisionable service.

8XX Call Processing

The system processes outbound 8XX calls as follows:

- 1. The CA signals the AIN FS to perform an 8XX query.
- 2. The AIN FS performs an internal database query.
- **3.** If an internal record is found for the 8XX number, the AIN FS provides the routing information to the CA and the call is attempted.



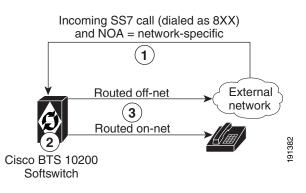


Notes for Figure 1-1

- 1. A subscriber dials an 8XX call.
- 2. The system attempts to translate the 8XX call to a DN in its local database.
- **3.** If there is no record in the local database, the system sends a query to the SCP and receives a translated DN.
- **4.** The system routes the call to the appropriate subscriber (on-net call) or external network (off-net call).

Figure 1-2 shows the processing of an 8XX call received from the network with a network-specific NOA.

Figure 1-2 Processing of an 8XX Call with a Network-Specific NOA



Notes for Figure 1-2

- 1. The system receives an incoming SS7 call with a toll-free (8XX) dialed DN and with NOA=network-specific.
- 2. The system attempts to translate the 8XX call to a DN in its local database.
- **3.** The system routes the call to the appropriate subscriber (on-net call) or external network (off-net call).

Local Toll-Free Database

This section explains how the system uses information from the local toll-free database.

The Cisco BTS 10200 Softswitch provides the ability to translate inbound/outbound 8XX numbers at the Feature Server (FS) using a local 8XX database. The 8XX service supports the following features:

- Origin-dependent routing
- Time-of-day routing
- Percentage-based routing
- Information digit-based screening
- Black/white list screening

The Cisco BTS 10200 Softswitch also supports optional DNIS service. In an 8XX DNIS service, when a call is terminated to a PBX (call center), 4 digits are outpulsed to the PBX to identify the originally dialed 8XX number. In case of custom DNIS, up to 22 digits can be outpulsed with additional information such as:

- Original 8XX number dialed
- Automatic number identification (ANI)
- Originating line information of the calling party

When a translated number (for an original 8XX call) is received, the Analyzed Info DP triggers the FS. The Cisco BTS 10200 Softswitch looks up the DNIS and TG information for the call. The DNIS information is then outpulsed to the PBX. If an overflow condition is encountered, the call is routed to the overflow trunk. The overflow trunk can be a PSTN trunk.

See SR-2275, *Telcordia Notes on the Network*, Section 14.6 for additional information on toll-free database services.

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SCP-Based Toll-Free Services

This section explains how the system uses information from the external toll-free database.

The Cisco BTS 10200 Softswitch communicates with an SCP-based database called the toll-free database service, which contains information for routing the call. The database service provides information about the network service provider selected to complete the call, and information for translating the toll-free number to a specific 10-digit directory number (DN). The routing of the call can vary depending on the arrangements made between the toll-free subscriber and the network service provider. These arrangements can include selective routing based on the time of day, day of week, and location from which the call originates.

Provisioning Commands

To provision this feature, see the 8XX (Toll-Free Calling) provisioning procedure in the Cisco BTS 10200 Softswitch Provisioning Guide.

Active Call Information Display

The Cisco BTS 10200 Softswitch allows the service provider to display information about an active (in-progress) call (originating or terminating). The implementation is based on Telcordia document *GR-529-CORE, Call Tracing (FSD 15-03-000)*. The display is performed using a CLI query command.

The input parameter can be any subscriber-specific or trunk-specific information including:

- Subscriber-specific information:
 - DN
 - TSAP address of the residential gateway (RGW)
 - MLHG ID and terminal
 - Centrex group ID and extension
 - Termination ID
- Trunk-specific information:
 - SIP call ID
 - H.323 call ID
 - Trunk group ID and trunk ID (for SS7, ISDN or CAS)

The system output (display) includes available information about the calling and called parties (for example, calling number, called number, call state, trunk type, call ID, charge number, and redirected number). The system supports both brief and verbose display modes.



For additional information about required inputs and available outputs, see the "Displaying Active Call Information" section in the *Operations and Maintenance Guide* and the "Query" chapter in the *Command Line Interface Reference Guide*.

Alerting Notification to Third-Party Feature Server

The Cisco BTS 10200 Softswitch can be provisioned to deliver alerting notification (notification of power ringing or call-waiting tone on the subscriber line) and call data to a third-party feature server (3PTYFS). This feature is available for both off-net-to-on-net calls and on-net-to-on-net calls. It can be assigned globally (for all subscribers on the switch), on a per-POP basis, or for individual subscribers. The service provider can use appropriately designed and configured feature servers to make use of this notification and data to provide value-added services to subscribers; for example, delivery of caller ID on a subscriber television or computer screen.

Note

Throughout this document, this feature (alerting notification to 3PTYFS) is referred to as Alerting Notification.



This document does *not* describe the messaging interface specs or call-data fields provided by the Cisco BTS 10200 Softswitch. Developers of applications for the 3PTYFS who are interested in the interface specifications should contact their Cisco account team.

Background Information

Alerting Notification was originally introduced in Release 3.5.4. In Release 4.5.x, this feature supports advertising of an externally addressable FQDN to an external third-party feature server when necessary. Previously, the FQDN advertised in Contact and Via headers in a SIP INVITE message resolved to an internal management network IP address. In addition, Release 4.5.x supports the provisioning of this feature on a per-POP basis.

Call Flow

The call flow works as follows:

- 1. The Cisco BTS 10200 Softswitch receives signaling for an incoming call, and sets up the call to the subscriber line.
- At the CALL_ACCEPTED trigger detection point (TDP) in the call, the Cisco BTS 10200 Softswitch generates a CALL_ACCEPTED_NOTIFY trigger, and sends a SIP Invite message directed to a 3PTYFS. The SIP Invite message includes a Feature Control Protocol (FCP) attachment containing the call data.
- 3. The 3PTYFS receives the SIP message and executes any actions that have been programmed into it.

Alerting Notification has no impact on the setup or progress of the call. The Cisco BTS 10200 Softswitch continues with normal call processing regardless of any response from the 3PTYFS.

Prerequisites

The Cisco BTS 10200 Softswitch locates the 3PTYFS via a TSAP address that is provisioned in the Cisco BTS 10200 Softswitch database. If the TSAP address is a domain name, the domain name must also be configured in the service provider domain name system (DNS) server.

The Cisco BTS 10200 Softswitch advertises its own TSAP address to the 3PTYFS. There are specific requirements for entering this information during initial software installation. For details, see the "Installation Considerations" section on page 1-20.

The 3PTYFS should be provisioned to support this feature in accordance with the applicable product documentation. The Cisco BTS 10200 Softswitch does not send any provisioning or status/control commands to the 3PTYFS.

The data in the FCP attachment generated by the Cisco BTS 10200 Softswitch is plain ASCII text and is not encrypted. The security of the connection between the Cisco BTS 10200 Softswitch and the 3PTYFS is the responsibility of the service provider.



It is the responsibility of the 3PTYFS to honor the presentation privacy restrictions, and control any usage or display of this information based on those restrictions.

Restrictions and Limitations

CALL_ACCEPTED_NOTIFY Trigger Sent for Incoming Calls to Subscribers Only

The system sends the CALL_ACCEPTED_NOTIFY trigger to the 3PTYFS only if the called party is a subscriber on the Cisco BTS 10200 Softswitch. This is true even if the feature is provisioned globally (at the office level) on the Cisco BTS 10200 Softswitch.

Limitations when Calling Party Is Using Certain SIP and H.323 Devices

Some calling-party devices (certain SIP- and H.323-based endpoints) may not send an explicit alerting indication (180 Ringing for SIP and Alerting for H.323). In these cases, the Call Agent does not report the CALL_ACCEPTED_NOTIFY trigger to the 3PTYFS.

Subsequent Triggers

The Cisco BTS 10200 Softswitch does not send updated information to the 3PTYFS based on subsequent triggers in the call (following the CALL_ACCEPTED TDP). For example, if a user hangs up while another call is on hold (in call-waiting mode) and the phone is rung again, the Cisco BTS 10200 Softswitch does not report a trigger and does not send any data.

NAPTER and SRV Record Lookup Not Supported

This feature does not support the use of the Naming Authority Pointer (NAPTR) or DNS services (SRV) records for lookup of the 3PTYFS domain name. The DNS server must be populated with the address (A) record for the fully qualified domain name (FQDN) specified in the TSAP address of the 3PTYFS.

Status and Control Commands

The **status feature-server** command reports the status of feature server components internal to the Cisco BTS 10200 Softswitch. However, the Cisco BTS 10200 Softswitch does not send any status or control commands to the 3PTYFS.

Installation Considerations

For Alerting Notification to function correctly, specific data must be entered into the opticall.cfg file at the time of initial Cisco BTS 10200 Softswitch software installation. The choice of data depends on whether the 3PTYFS will be deployed in the same private (internal) management network as the Cisco BTS 10200 Softswitch, or in a public network.

• If the 3PTYFS is deployed in the same private (internal) management network as the Cisco BTS 10200 Softswitch, the 3PTYFS can obtain the IP address of the Cisco BTS 10200 Softswitch from the DNS server. That DNS entry will resolve correctly to the private IP address.

• If the 3PTYFS is deployed in a public network (outside the private management network of the Cisco BTS 10200 Softswitch), the 3PTYFS must reach the Cisco BTS 10200 Softswitch by using an external IP address. In this case, you must populate opticall.cfg and the DNS server with the external IP address for the Cisco BTS 10200 Softswitch.

This installation data also affects the provisioning requirements for the 3PTYFS in the Feature Server table:

- If the 3PTYFS is deployed in the private management network, the EXTERNAL-FEATURE-SERVER parameter must be set to N.
- If the 3PTYFS is deployed in a public network, the EXTERNAL-FEATURE-SERVER parameter must be set to Y.



For additional information, including provisioning steps, see the *Alerting Notification to Third-Party Feature Server* document.



Installation of the 3PTYFS and peripheral devices is outside the scope of this document. Those devices and software should be installed according to the applicable product documentation.

Calling Party Number Options for Outgoing SETUP Messages

The Cisco BTS 10200 Softswitch provides options for controlling the calling party number (CPN) data sent in the outbound SETUP message on calls outbound or redirected from the Cisco BTS 10200 Softswitch to the PSTN.

Option to Send Billing DN as CPN for Outgoing Calls

The system has a provisionable option for sending a subscriber billing DN (or main DN of a PBX subscriber) as the CPN in outgoing SETUP messages on outgoing nonemergency calls. This is provisionable using the SEND-BDN-AS-CPN token in the Subscriber table.

- If SEND-BDN-AS-CPN is set to Y, the system sends the billing DN of the subscriber as the CPN in the outgoing setup message. If the billing DN is not provisioned, the system sends the value of the subscriber directory number (DN1).
- If SEND-BDN-AS-CPN is set to N, the system sends the subscriber DN1 as the CPN in the outgoing setup message. For PBX, the system sends the individual subscriber DN (received in the SETUP message) as the CPN in the outgoing setup message.

Note

The sending of the subscriber name and number is subject to the provisioning of the PRIVACY token in the Subscriber table.

Option to Send Billing DN as CPN for Emergency Calls

The system has a provisionable option for sending a subscriber billing DN (or main DN of a PBX subscriber) as the CPN in outgoing SETUP messages on outgoing emergency calls. This is provisionable using the SEND-BDN-FOR-EMG token in the Subscriber table.



In this document, emergency calls are calls to DNs that are provisioned as call-type=EMG in the Destination table.

- If SEND-BDN-FOR-EMG is set to Y, the system sends the billing DN of the subscriber as the CPN in the outgoing setup message. If the billing DN is not provisioned, the system sends the value of the subscriber directory number (DN1).
- If SEND-BDN-FOR-EMG is set to N, the system sends the subscriber DN1 as the CPN in the outgoing setup message. For PBX, the system sends the individual subscriber DN (received in the SETUP message) as the CPN in the outgoing setup message.

Option to Send Redirecting Number as CPN for Redirected Calls

This feature allows the service provider to control the CPN data sent in the outbound SETUP message on redirected calls outbound from the Cisco BTS 10200 Softswitch to the PSTN.

The CPN option is provisionable (via CLI commands) using the SEND-RDN-AS-CPN token in the TRUNK-GRP table:

- If this token is set to Y (yes), the system overwrites the existing CPN with the redirecting number (RDN) and includes the new value in the outbound SETUP message.
- If this token is set to N (no), the system does not change the existing CPN data. N is the default value.

This feature is applicable to the following scenarios:

- Redirection by a subscriber phone
- Redirection of a basic or Tandem call

Figure 1-3 shows an example of the networks and phones involved in redirection by a subscriber phone. Table 1-4 explains how to provision the SEND-RDN-AS-CPN token for various call-redirection scenarios and results.

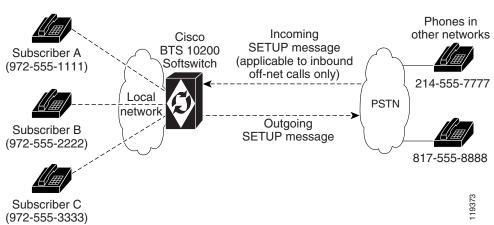


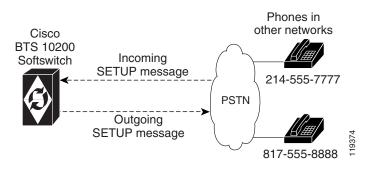
Figure 1-3 General Network View for Redirection by a Subscriber Phone

Scena	rio (See Figure 1-3)	Existing CPN and RDN Data (Example)	Value Provisioned for SEND-RDN- AS-CPN	Effect On Outbound SETUP Message	Content of Outbound SETUP Data (Example)
On-ne	t to off-net call (either of the following):	CPN=	Y	Overwrite CPN	CPN=
A -> I	3 -> fwd -> C -> fwd -> off-net	972-555-1111		with RDN	972-555-3333
A -> 0	C -> fwd -> off-net	RDN= 972-555-3333			RDN= 972-555-3333
			N	Do not change CPN	CPN= 972-555-1111
					RDN= 972-555-3333
Off-ne	et to on-net to off-net call:	CPN=	Y	Overwrite CPN	CPN=
Inbound off-net call -> B -> fwd -> off-net		214-555-7777		with RDN	972-555-2222
Note	In this example, the existing RDN (from the incoming SETUP message)	RDN= 817-555-8888			RDN= 972-555-2222
	is 817-555-8888. The new RDN is the DN of the forwarding	(from incoming	N	Do not change CPN	CPN= 214-555-7777
	phone, Subscriber B—972-555-2222.	SETUP message)			RDN= 972-555-2222

Table 1-4 Provisioning SEND-RDN-AS-CPN Token in TRUNK-GRP Table for Redirection by a Subscriber Phone

Figure 1-4 shows an example of the networks and phones involved in redirection of a basic or Tandem call. Table 1-5 explains how to provision the SEND-RDN-AS-CPN token for call-redirection scenarios and results. Note that the content of the outbound SETUP message depends on whether the RDN is available in the incoming SETUP message.

Figure 1-4 General Network View for Redirection of Basic or Tandem Call



Scenario (See Figure 1-4)	Existing CPN and RDN Data (Example)	Value Provisioned for SEND-RDN- AS-CPN	Effect On Outbound SETUP Message	Content of Outbound SETUP Data (Example)
Off-net to off-net call (basic or Tandem) with RDN available in the incoming SETUP message: off-net call -> Cisco BTS 10200 Softswitch -> off-net	CPN= 214-555-7777 RDN= 817-555-8888	Y	Overwrite CPN with RDN	CPN= 817-555-8888 RDN= 817-555-8888
	(from incoming SETUP message)	N	Do not change CPN	CPN= 214-555-7777 RDN= 817-555-8888
Off-net to off-net call (basic or Tandem) with RDN <i>not available</i> in the incoming SETUP message: off-net call -> Cisco BTS 10200 Softswitch	CPN= 214-555-7777 RDN not available	Y	Do not change CPN	CPN= 214-555-7777 RDN not available
-> off-net	(from incoming SETUP message)	N	Do not change CPN	CPN= 214-555-7777 RDN not available

Table 1-5 Provisioning SEND-RDN-AS-CPN Token in TRUNK-GRP Table for Redirection of Basic or Tandem Call

<u>)</u> Tip

To view all the tokens in this table, see the Trunk Group Table information in the Cisco BTS 10200 Softswitch Command Line Interface Reference Guide.

Dialing Parity (IntraLATA Toll Presubscription)

The Cisco BTS 10200 Softswitch supports this feature in accordance with Telcordia document GR-693-CORE, LSSGR: Presubscription Indication (FSD 20-24-0000).

Dialing parity—also known as intraLATA toll presubscription—allows subscribers to select a telecommunications company for intraLATA calls (local toll calls) in the same way they select a long-distance provider. With dialing parity, subscribers are able to dial the number they want and have a preselected carrier—a competitive local exchange carrier (CLEC), incumbent local exchange carrier (ILEC), or a long-distance carrier—automatically handle the call if it is a local (intraLATA) toll call. Preselecting a local toll carrier eliminates the need for dial-around service for local toll calls (101XXXX numbers). Prior to implementation of dialing parity, long-distance carriers provided intraLATA service by dialing around an ILEC or CLEC via 101XXXX numbers (carrier access codes—CACs).

Local access and transport areas (LATAs) were created after the breakup of the AT&T system. LATAs are also known as called service areas or local toll calling areas. CLECs and ILECs provide two types of calls to their subscribers within the LATA:

- Local calls
- Local toll calls

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Local toll calls are typically calls to places more than 16 miles from the subscriber location in urban areas and more than 13 miles in rural areas. Local toll calls do not qualify as either local or long distance—they are between the two and are subject to different rates.

Local Number Portability (LNP)

The Cisco BTS 10200 Softswitch supports local number portability (LNP) for North American and ITU-based systems. For general information, see *Number Portability Switching Systems*, T1.TRQ-02-2001, which provides unofficial agreement within T1S1. T1S1 is the ATIS accredited body for signaling. This document is available at http://www.atis.org.

LNP permits subscribers who change their local phone company to keep their existing telephone numbers. An FCC order requires this feature in the 100 top metropolitan service areas in the United States. LNP permits calls to be routed to the subscriber's new local switch without any particular per-call action required of either the calling or called party. Each switch contains a database of the office codes (NPA-NXXs) associated with subscriber numbers that have been ported in and ported out.



The LNP feature can be made available to all subscriber lines connected to a Cisco BTS 10200 Softswitch using the default office service ID, or to all subscribers in a specific POP using the office service ID. See the "Office Service ID and Default Office Service ID" section on page 2-124 for a general description of this provisionable service.

The Cisco BTS 10200 Softswitch supports the LNP function as follows:

- Ranges/blocks of ported numbers are provisionable in the Cisco BTS 10200 Softswitch, with block size granularity from 100 to 10,000 DNs per block.
- During the call processing, if the dialed digits/destined digits match 3 to 10 contiguous digits of a
 ported NPA-NXX-XXXX at the Info_Analyzed/ Collected_Info trigger detection point (TDP), a
 query is initiated to an external database using the AIN Info_Analyzed message. This LNP trigger
 is also known as the public office dialing plan (PODP) trigger.
- The Cisco BTS 10200 Softswitch processes the received response (Analyze_Route) from the TCAP query and determines whether the dialed digits have been translated to a location routing number (LRN):
 - If the CalledPartyID received from the Analyze_Route differs from the dialed digits (that is, the LRN comes back), the call is routed based on the received CalledPartyID as the ISUP IAM CalledPartyNumber and sets the ForwardCallIndicator parameter to "Number translated". The ISUP IAM also includes the ISUP GenericAddress Parameter (GAP) set to the dialed digits.
 - If the CalledPartyID received from the Analyze_Route is the same as the dialed digits (that is, no LRN comes back), the call is routed based on the received CalledPartyID as the ISUP IAM CalledPartyNumber and sets the ISUP ForwardCallIndicator (FCI) parameter to "Number translated".
 - When the LNP query results in an error, the call is routed based on the dialed digits/destination digits, and does not include the ISUP GAP, and the FCI is set to "Number not translated".

For a comprehensive description of LNP functions and provisionable parameters, see the *Cisco BTS* 10200 Softswitch Dial Plan.

To provision LNP options, see the appropriate procedures from the list below:

- Local Number Portability (LNP) for ANSI/North America (in the Provisioning Guide)
- Local Number Portability (LNP) ITU Local BTS Database Query (in the Provisioning Guide)

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- *ITU Local Number Portability (LNP)* (a feature module)
- SS7 Provisioning (in the Provisioning Guide)

Split-NPA

When DNs are exhausted within an NPA, an additional NPA is assigned to the region. The new NPA may be allocated as an overlay over the existing NPA, in which case there is no major impact to the Cisco BTS 10200 Softswitch. However, when the new NPA is assigned based on a geographical split of the region, there are significant impacts. The assignment of the new NPA based on a geographical split is referred to as split-NPA.

The split-NPA feature impacts both provisioning (EMS) and call processing subsystems in the Cisco BTS 10200 Softswitch. Several provisioning tasks must be performed to introduce a new NPA into a region, including:

- Duplicate records (tasks to be performed before permissive period)
- Update ANI records (tasks to be performed during permissive period)
- Cleanup (tasks to be performed after permissive period)



Permissive period is the time frame where both old NPA and the new NPA can be dialed to reach the subscribers affected due to the split-NPA feature.

Before the permissive period begins, subscribers affected due to the split-NPA can be reached only via the old NPA. Duplicate records for both old and new NPA are created before the permissive period begins.

During the permissive period, both old and new NPA can be dialed (10-digit dialing is required to reach a subscriber in the affected NPA). The subscriber (ANI) and subscriber feature data records are updated to the new NPA during the permissive period.

Once the permissive period ends, subscribers affected due to the split-NPA can be reached only via the new NPA. This is referred to as the mandatory dialing period for the new NPA. The duplicate records created before the permissive period are cleaned up after the mandatory period begins.

For additional information on split-NPA, see the ATIS document INC97-0404-016, NPA Code Relief Planning & Notification Guidelines.



The split-NPA feature does recognize a leading digit (example: 9) that has been provisioned to represent a centrex subscriber DN. The only leading digit that the split-NPA feature takes into account is the leading "1" digit.



To provision this feature, see the *Split NPA provisioning procedure* in the *Cisco BTS 10200 Softswitch Provisioning Guide*.

T.38 Fax Relay

In previous releases, the Cisco BTS 10200 Softswitch partially supported T.38 fax relay call agent controlled mode on MGCP and H.323 interfaces. Release 4.5.x supports the PacketCable environment (eMTA subscribers), as well as extending T.38 fax support to SIP, NCS, and TGCP interfaces. The support is for both the line and trunk side of SIP, and interworks with the existing T.38 support over MGCP line and trunk side, and H.323 trunk side.

The Cisco BTS 10200 Softswitch supports ITU-T T.38 procedures on the following interfaces:

- NCS MTA subscribers
- MGCP subscribers
- MGCP (or TGCP) trunking gateways (SS7, ISDN)
- H.323 trunks

T.38 fax is supported for the following H.323 configurations:

- H.323 trunk using fast connect procedure (fast start)
- H.323 trunk using non-fast connect procedure (slow start)
- H.323 trunk using gatekeeper (H.225 RAS messaging)
- H.323 trunk not using gatekeeper (direct trunks)
- H.323 trunk with and without H.245 tunneling enabled
- SIP trunks
- RFC 3261-compliant SIP endpoints (Release 4.5.1 only)

Following is a list of industry references for T.38 fax relay

- ITU-T Recommendation T.38 (06/98) Procedures for Real-Time Group 3 Facsimile Communication Over IP Networks.
- ITU-T Recommendation T.38 Annex D SIP/SDP Call Establishment Procedures.
- F. Andreasen, draft-andreasen-mgcp-fax-04.txt, August 2004, FXR: MGCP Fax Package.
- H.323 Specification Annex D Version 2 (also incorporated into H.323 Version 4).

List of T.38 Fax Topics Covered in this Section

This section covers the following system capabilities:

- MGCP/NCS Interface—Fax Modes Supported
- SDP Attributes Support for T.38 Fax Relay
- MTA DQOS Support for T.38 Fax Relay
- Fallback to Audio Media on T.38 Negotiation Failure
- Audio Restore After Successful T.38 Fax Transmission
- T.38 Glare Handling
- SDP Attributes Encoding Formats
- SIP Support for Call Legs
- Protocol Interworking
- Control Configuration

- Restrictions and Limitations
- Feature Provisioning Commands

MGCP/NCS Interface—Fax Modes Supported

The system supports the following fax procedures from the MGCP or NCS endpoint:

- T.38 loose mode (as defined by the FXR package)
- Cisco-proprietary gateway mode (not using the FXR package)
- Fax using existing audio media (fax pass-through)—The system does not support audio codec upspeeding in this case.

SDP Attributes Support for T.38 Fax Relay

The system supports the exchange of media path (SDP or H.245) attributes between the Cisco BTS 10200 Softswitch managed end devices used to support T.38. These attributes are:

- Capabilities attributes defined in RFC-3407 (which indicate the endpoint is T.38 capable)
- T.38 attributes defined in ITU-T T.38 Annex D (which negotiates T.38 properties between endpoints)

MTA DOOS Support for T.38 Fax Relay

There are DQOS considerations for NCS MTA subscribers when switching between audio and T.38 codec. The Cisco BTS 10200 Softswitch follows the DQOS flow characteristics for T.38 sessions described in PacketCable specification "pkt-sp-codec."

For DQOS, the Cisco BTS 10200 Softswitch creates new gates for the T.38 codec when switching to T.38 media, because endpoints can change their media IP and ports when switching between audio and T.38.

Fallback to Audio Media on T.38 Negotiation Failure

The system supports audio fallback on all interfaces supporting T.38 fax. Failure to negotiate T.38 media occurs when the non-fax detecting endpoint fails a request to modify the connection (MDCX) to T.38, and the MDCX contained a T.38 remote connection descriptor (SDP) from the fax-detecting end. When this occurs, a counter increments to track the event occurrences. Any other T.38 procedure fail scenarios do not trigger the fall back procedure. Instead, the call is cleared by the endpoint.

Audio fall back on T.38 negotiation failure requires a collaborative support by the endpoints. The MGCP/NCS endpoints requirements are:

- Non-Fax Detecting Endpoint
- Fax Detecting Endpoint

Non-Fax Detecting Endpoint

When the non-fax detecting endpoint fails the MDCX request to modify the connection to T.38, and this MDCX contained a T.38 remote connection descriptor (SDP) from the fax-detecting end, then the non-fax detecting endpoint continues with its current audio media settings.

Fax Detecting Endpoint

When the non-fax detecting endpoint fails a switch to T.38, the fax-detecting endpoint is still in T.38 media mode. The Cisco BTS 10200 Softswitch sends an MDCX to the fax detecting endpoint to abort T.38 procedures and revert back to previous audio media. The fax detecting endpoint responds with an audio SDP, and the Cisco BTS 10200 Softswitch exchanges this SDP with the remote end.

The Cisco BTS 10200 Softswitch applies the following recommendation, forecast as an update, to the MGCP FXR Package. However, it does not send the "off" Fax LCO to abort T.38 procedures. Instead, it sends an "L:revious codec>" LCO.

For remote SIP endpoints or gateways sending a Re-Invite with T.38 SDP to switch to T.38 media receiving a fail response should fall back to previous SDP settings.

For H.323 calls, if the non-H.323 endpoint fails to switch to T.38 fax while the H.323 side is already switched to T.38 fax, then the H.323 side reapplies H.245 procedure to return to audio codec.

Audio Restore After Successful T.38 Fax Transmission

The Cisco BTS 10200 Softswitch can restore audio media after successful T.38 fax transmission. However, this feature is beyond the scope of the FXR package, and requires collaborative support by the endpoints.

For the Cisco BTS 10200 Softswitch to provide this feature, the MGCP/NCS endpoints must complete the following:

- Once the Cisco BTS 10200 Softswitch is notified of the completion of T.38 fax transmission, it sends an MDCX to the notifying endpoint to revert back to previous audio media. The notifying endpoint responds with an audio SDP, and the Cisco BTS 10200 Softswitch exchanges this SDP with the remote end.
- Remote SIP endpoints or gateways must send a Re-Invite message containing audio media SDP to restore audio.

T.38 Glare Handling

The Cisco BTS 10200 Softswitch applies a call agent controlled switch to T.38 fax media when initiated by either the originating or terminating endpoint. This includes the scenario in which both endpoints initiate the switch, causing a glare condition at the Cisco BTS 10200 Softswitch.

For details about glare handling, contact your Cisco support representative.

SDP Attributes Encoding Formats

SDP capability attributes can be formatted two ways when sent from the Cisco BTS 10200 Softswitch out of the network using MGCP, NCS and SIP protocols.

• RFC 3407 based encoding method

In this method, SDP is encoded:

• Cisco proprietary method

In this method, SDP is encoded:

a=X-sqn:0 a=X-cap: 1 audio 0 18 96 a=X-cpar: a=fmtp:96 0-16,32-35 a=X-cap: 2 image udptl t38

The Cisco BTS 10200 Softswitch selects the encoding method based on following guidelines:

- For the SIP interface, the Cisco BTS 10200 Softswitch always performs encoding using the RFC 3407-based encoding method.
- For MGCP/NCS endpoints, the Cisco BTS 10200 Softswitch uses a provisionable field to choose between the encoding methods.
- The H.323 interface does not use SDP, so this section is not applicable to H.323.

SIP Support for Call Legs

The system supports T.38 fax relay call agent controlled mode to SIP lines and trunks where one or both call legs are SIP. In addition to passing the SDP advertisement of T.38 fax codec, the Cisco BTS 10200 Softswitch issues a re-invite or similar SIP method to change the codec of an established session to a T.38 fax relay codec session once the fax event is detected (such as operating in a call agent controlled mode).

This has a dependency on SIP CPEs and media gateways. Cisco recommends that the T.38 session is initiated by the terminating side of endpoints.

For more information, refer to ITU T.38 03/2002.

Protocol Interworking

The Cisco BTS 10200 Softswitch supports T.38 for both SIP lines and SIP trunks, and interworking with the following protocols as either terminating or originating:

- MGCP line and trunk
- NCS (for cable)
- H.323 trunk

After the fax is done, the call falls back to a voice call.

The detailed protocol interworking matrix is shown in Table 1-6.

Table 1-6 Protocol interworking matrix

	H.323 Trunk	MGCP line (eMTA using NCS) and trunk	SIP line and trunk ¹
H.323 trunk	X	X	X
MGCP (eMTA using NCS)	X	X	X
SIP line and trunk ¹	X	X	X

1. SIP line support for T.38 fax is provided in Release 4.5.1 only.

Figure 1-5 shows an example of MGCP and SIP interworking.

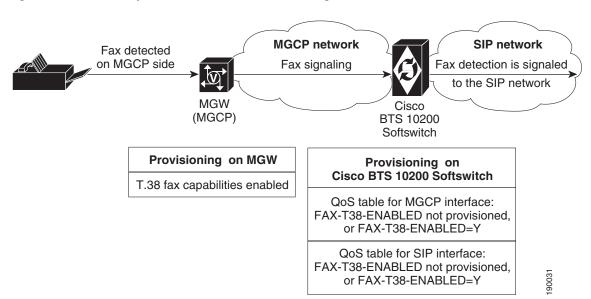


Figure 1-5 Example of MGCP and SIP Interworking for T.38 Fax

Control Configuration

The Cisco BTS 10200 Softswitch can be configured with either call agent controlled T.38 mode or gateway controlled T.38 mode.

Restrictions and Limitations

T.38 fax relay has some limitations in Release 4.5.x, including:

- Cisco BTS 10200 Softswitch Unsupported T.38 Fax Transport Methods
- Cisco BTS 10200 Softswitch Unsupported T.38 Interface
- MGCP/NCS Interface—T.38 Fax Modes Unsupported
- Relationship Between T.38 Fax Transmission and Call Features on the Cisco BTS 10200 Softswitch
- End-to-End SDP Exchange for T.38 Media and the H.323 Interface
- Internet Fax Terminal Endpoint Types—Not Supported
- Cisco BTS 10200 Handling of T.38 Failure Event Notification from an MGCP or NCS Endpoint

Cisco BTS 10200 Softswitch Unsupported T.38 Fax Transport Methods

The Cisco BTS 10200 Softswitch does not support following T.38 fax transport methods from ITU-T T.38:

- TCP
- Fax over RTP

Cisco BTS 10200 Softswitch Unsupported T.38 Interface

Support for ITU-T T.38 procedures are not provided on the following interfaces managed by the Cisco BTS 10200 Softswitch (including the interworking between supported and unsupported interfaces):

- RFC 3261-compliant SIP endpoints are supported in Release 4.5.1 but are not supported in Release 4.5.0.
- H.323 subscribers.
- CAS trunks.

MGCP/NCS Interface—T.38 Fax Modes Unsupported

The Cisco BTS 10200 Softswitch does not support the following MGCP FXR package defined T.38 fax modes from the MGCP/NCS endpoint:

- T.38-Strict mode
- Gateway mode (Though Cisco-proprietary Gateway mode is supported)
- Off mode

If the Cisco BTS 10200 Softswitch does not signal any FXR fax mode in the Local Connection Options, including "Off" mode, the gateway can engage in Gateway mode. If this occurs, the Cisco BTS 10200 Softswitch does not receive any Gateway mode notification events from the endpoint because it does not request them. The Cisco BTS 10200 is not notified of the gateway mode activity. The Cisco BTS 10200 Softswitch honors the FXR package requirements during gateway mode by not interfering with the gateway procedures in this case.

Relationship Between T.38 Fax Transmission and Call Features on the Cisco BTS 10200 Softswitch

The Cisco BTS 10200 Softswitch makes no distinction between calls using audio and calls using fax transmission with respect to call features.

Cisco recommends that Cisco BTS 10200 Softswitch subscribers disable the Call Waiting feature before making a call for fax transmission. This protects against any interruptions during fax transmission.

In certain multi-party call feature scenarios, such as three-way calling where a user has engaged the three-way call feature on Cisco BTS 10200 Softswitch and one party attempts a switch to T.38 fax, the endpoint fails to switch the call to T.38. The party may be either disconnected or reverted back to audio depending on the endpoints capabilities.

End-to-End SDP Exchange for T.38 Media and the H.323 Interface

The H.323 protocol must negotiate T.38 fax connection attributes (example: bit rate, maximum buffer size) during the voice call establishment using Terminal Capability Set (TCS) messages. However, for SIP and MGCP, the endpoint does not report T.38 fax connection attributes until the fax actually starts. When this occurs from an interworking H.323 endpoint to a SIP/MGCP interface, and the H.323 endpoint is ready to send TCS message during voice call establishment phase, the T.38 fax attributes are not available from the SIP/MGCP interface.

To overcome this interworking limitation, all Cisco IOS gateways assume the defaults for these attributes while exchanging TCS messages. Cisco BTS 10200 follows the same philosophy for H.323 to/from MGCP/NCS and H.323 to/from SIP calls. Cisco BTS 10200 assumes following defaults:

• Maximum Bit Rate = 14.4 kbps (This field can be configured in T38_MAX_BIT_RATE field in the CA_CONFIG table)

- Fill Bit Removal = false
- MMR Transcoding = false
- JBIG Transcoding = false
- Data Rate Management Method = transferredTCF
- Maximum Buffer Size = 200 (This field can be configured in T38_MAX_BUFFER_SIZE field in CA_CONFIG table)
- Maximum Datagram Size = 72 (This field can be configured in T38_MAX_DATAGRAM_SIZE field in the CA_CONFIG table)
- Error Correction = t38UDPRedundancy

To overcome other interworking limitations with SIP, IOS H.323 gateways send the fax UDP port in H.245 Open Logical Channel (OLC) messages. A provisioning field (REMOTE_FAX_PORT_ RETRIEVAL_MSG) is added in h323-tg-profile and h323-term-profile, enabling the H.323 interface to read remote endpoint's fax UDP port either from OLC message or from OLC Ack message.



This does not apply to T.38 fax transmissions across H.323 to H.323 calls on the Cisco BTS 10200 Softswitch. In this case, the H.245 messages are exchanged directly through the Cisco BTS 10200 Softswitch.

Internet Fax Terminal Endpoint Types—Not Supported

The Cisco BTS 10200 Softswitch does not support endpoints that negotiate for T.38 media on initial call setup. These endpoints include internet fax terminals or internet-aware fax devices, and internet telephony gateways that only support T.38 real-time fax communications (by design or by configuration), or are statically configured to support T.38 fax calls only.

Cisco BTS 10200 Handling of T.38 Failure Event Notification from an MGCP or NCS Endpoint

The Cisco BTS 10200 Softswitch releases the call if the MGCP or NCS fax-detecting endpoint issues a "t38(failure)" event. This event is sent by the endpoint when it encounters some kind of problem with the T.38 fax relay procedure as stated in the MGCP FXR package.

Feature Provisioning Commands

Provisioning commands are available in the Cisco BTS 10200 Softswitch Provisioning Guide.

To provision this feature, see the *T.38 Fax Relay provisioning procedure* in the *Cisco BTS 10200 Softswitch Provisioning Guide*.

Trunk and Line Testing

This section describes trunk and line testing features, and includes the following topics:

- Trunk Testing
- Testing Capability for 911 FGD-OS Trunks
- Network Loopback Test for NCS/MGCP Subscriber Endpoints

• Network Loopback Test for ISDN PRI Trunks (Release 4.5.1, MR1)



For general troubleshooting procedures, see the Troubleshooting Guide.

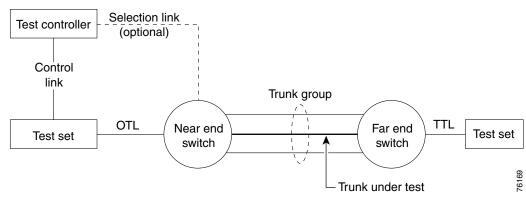
Trunk Testing

Trunk testing is used to evaluate the transmission quality of the shared trunks that interconnect switching systems. Trunk testing is extremely important in monitoring system health, because it is the only practical way to objectively evaluate the performance of individual trunks.

Trunk testing requires the following equipment and test lines. (Some additional types of equipment and lines may also be used.) A basic system setup is shown in Figure 1-6.

- Test controller
- Test set(s)
- Originating test line (OTL)
- Terminating test line (TTL)





Near End Test Origination Test Calls

The Cisco BTS 10200 Softswitch supports calls used to test individual trunks that connect a local gateway with a gateway or PSTN switch at a remote office. The Cisco BTS 10200 Softswitch supports OTL and TTL capability. User-provided test equipment and, optionally, test controllers may be connected to the test lines. Proper selection of test equipment and test functions helps to ensure interoperability between different carriers.



The processes described in this section are applicable to the Cisco BTS 10200 Softswitch. The processes may work differently on other switches.

The process for testing a Cisco BTS 10200 Softswitch OTL is as follows:

1. The user verifies that the remote CO has the desired 1xx test line available.

- 2. The user sets up a test device on a CAS TGW that is connected to the local Cisco BTS 10200 Softswitch.
- **3.** The user provisions the CAS-TG-PROFILE table, setting TEST-LINE = YES. (Provisioning commands are described in the *Cisco BTS 10200 Softswitch Command Line Interface Reference Guide*.)
- 4. On the test device at the CAS TGW side, the user enters digits representing the circuit to be tested and the test to be performed:
 - TG, for example 0003
 - Trunk number, for example 0018

The complete trunk address in this example is 00030018.

- Test type (10x), for example 104

The technician dials KP-00030018-104-ST.

5. The Cisco BTS 10200 Softswitch automatically inserts either 9581 or 9591 in front of the test type digits to create a dialing string.

The complete test string in this example is PREFIX | 00030018 | 9581104 | END.

Nata

Note Alternatively, with the Cisco BTS 10200 Softswitch, the user can dial the test type with the 9581 or 9591 included: *KP-00030018-9581104-ST*.

- **6.** The Cisco BTS 10200 Softswitch selects the trunk to be tested based on the user-defined trunk address.
- 7. The TGW outpulses the digits to the remote switch over the designated trunk.

1xx Test Line Support

When the Cisco BTS 10200 Softswitch is the near-end switch, the following process takes place at the remote switch:

- 1. The remote switch recognizes the trunk test prefix (9581 or 9591) on the incoming signal, and the test type is used to route the test to the appropriate test line.
- 2. The appropriate tests are performed on the test set.
- 3. Additional test processes may occur, depending on the specific test configuration.

When the Cisco BTS 10200 Softswitch is supporting the TTL capability (test call originated at another switch), the process is as follows. The Cisco BTS 10200 Softswitch receives the 958 or 959 call, recognizes the 958 or 959 type, and routes the test to the appropriate test line.

With Release 4.5.x, the Cisco BTS 10200 Softswitch supports the capability for a TDM-based testing device to perform continuity testing over an MF CAS TDM trunk interface. This capability requires that an MGCP-based trunking gateway is present in the test path. The TDM test type is the traditional 1xx test type, with an additional enhancement in Release 4.5.x—the ability to route the test call to a specified DN on a given trunk circuit.

T108 Test Line Support

The T108 test line feature determines the performance of trunks connecting digital exchange switches, including voice over packet (VoP) softswitches. Cisco BTS 10200 Softswitch incoming trunks requesting other 1xx-type test lines are routed to shared test lines for the requested tests, regardless of

which gateway terminates the trunk or which gateway/IAD terminates the test line. The T108 test line feature requests a test to be performed within the same gateway where the trunk under test (TUT) is terminated, and provides a digital loopback within the gateway. The T108 test line feature supports manual and automated testing.

The T108 test line sequence is as follows:

- The near-end switch originates the test sequence by placing a test call, identifying the trunk to be selected, and the test line number. A digital test pattern generator is used in the test setup shown in Figure 1-6.
- **2.** The near-end switch uses the trunk identifier to override normal call processing and select only the requested trunk.
- **3.** The far-end switch responds to the destination number and connects to the T108 test line. The T108 test line enables a digital loopback.
- **4.** When the near-end switch receives answer supervision, it conducts digital test sequences to ascertain trunk performance.
- 5. Once the test sequences are completed, the near-end switch releases the test call and both switches release the trunk connection.
- **6.** The far-end switch can detect if the test connection exceeds a preset time, and releases the test connection if the preset time is exceeded.

Note

The T108 test line is also used for trunk redirection (wholesale dial) for Internet services where the carrier modem termination is integrated into the trunk gateway. In this case, the integral digital stored program (DSP) normally supports modem-only transmissions.

Testing Capability for 911 FGD-OS Trunks

When turning up 911 Feature Group D Operator Services (FGD-OS) trunks, there is an exchange of Off-hook/On-hook signaling and the passing of tone back and forth without a complete call setup. Signaling for this function is based on the MGCP MO package.

Upon receiving a CLI command or Test Access request, Cisco BTS 10200 Softswitch sends the request to the gateway via MGCP signaling to trigger the test capability on 911 trunk at the gateway (not part of a call setup sequence). The Cisco BTS 10200 Softswitch reports the result to the operator upon receiving the notification from the gateway (for example, receiving off-hook or on-hook notification). Once this gateway test capability on a 911 trunk is in place, it can be invoked remotely across the MGCP interface associated with the Cisco BTS 10200 Softswitch.

Note

In order to support this functionality, the gateway itself must be able to provide the test capability to send and monitor the reception of the signaling and passing tone without a call setup involvement.

Network Loopback Test for NCS/MGCP Subscriber Endpoints

This feature supports the capability for a testing device to perform network loopback tests from any line-side NCS/MGCP Residential Gateway or Media Termination Adapter (MTA). The loopback tests can be initiated from designated test endpoints (subscribers) controlled by the Cisco BTS 10200

Softswitch. The procedure for setting up the test includes configuring the test lines as subscriber terminations and provisioning the MGW parameters. The system allows NCS/MGCP endpoints in a trunk group to be provisioned as a test trunk group with specific test attributes.

Restrictions and Limitations

The following restrictions and limitations apply:

- The testing and tested devices must be configured on same Call Agent. The system cannot perform network loopback test calls that originate from another switch.
- The system does not provide the ability to perform network loopback testing across H.323 or SIP networks.
- You cannot perform the Network Loopback Test if the status of the subscriber to be tested is **unequipped** (**UEQP**) or **operational-out-of-service** (**OOS**).
- Although you can test this feature using a regular MTA as the testing device by configuring the endpoints as subscriber terminations in Cisco BTS 10200 Softswitch, you need appropriate test equipment to perform voice-quality testing.

Configuring and Operating

The procedures for configuring the lines and gateways, and the procedure for performing the tests, are described in the *Network Loopback Test* section of the *Cisco BTS 10200 Softswitch Troubleshooting Guide*.

Network Loopback Test for ISDN PRI Trunks (Release 4.5.1, MR1)

This feature allows operators to conduct network loopback testing originating from shared ISDN PRI trunks. The shared test trunk group accepts both normal and test calls. Test calls are identified by provisioning the call-type and call-subtype tokens in the Destination table. For detailed requirements and procedures for running this type of trunk test, see the *Cisco BTS 10200 Softswitch Troubleshooting Guide*.

