CHAPTER 1

Network Features

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

- Interoperability
- 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)
- 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), SIP Triggers, Split-NPA, and T.38 Fax Relay.
- Trunk and Line Testing

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

Note For information on subscriber features, see Chapter 3, "Subscriber Features." For information on outgoing call restriction options (Class of Service and Outgoing Call Barring) see Chapter 4, "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 BTS 10200. The valid formats for VSC ASCII strings are listed in the Vertical Service Code commands in the *Cisco BTS 10200 Softswitch CLI Database*. To view the current VSC values provisioned on your system, use the **show vsc** CLI command. 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 BTS 10200 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.

Caution 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 BTS 10200 Release 5.0 software. Earlier or later releases of platform software might be interoperable and it might be possible to use other functions on these platforms. The list in 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 BTS 10200.

Numbering Plans and Dialing Procedures

The BTS 10200 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*.

Table 1-1 Support for Numbering Plans and Dialing Procedures

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	

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 BTS 10200.

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

E.164 Dialing Plan Implementation

The BTS 10200 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 BTS 10200 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 BTS 10200 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
  - 16 in France

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

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.

Table 1-2 Examples of International Numbers

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

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.

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 BTS 10200 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 BTS 10200 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 BTS 10200 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 1-13 for a description.
- 900/976 information service calls—See the "Information Service Calls (900 and 976)" section on page 1-6 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 1-8.

This section describes BTS 10200 support for n11 services. The typical relationship between the n11 codes and the nature of dial (NOD) values is as follows.

n11 Code	NOD Value
211	INFO
311	NON-EMG
411	DA
511	TRAFFIC
611	REPAIR
711	RELAY
811	BUSINESS

For a complete list of NOD values, see the Nature of Dial command in the *Cisco BTS 10200 Softswitch CLI Database*. To view the current NOD values provisioned on your system, use the **show nod** CLI command.

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 BTS 10200 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 1-5.

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 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 NDC and the 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 BTS 10200 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*

This section covers the following topics:

- “Description” section on page 1-8
- “Important Provisioning Requirements” section on page 1-9
- “911 Overflow Announcement” section on page 1-10
- “Emergency 911 Trunk Connection Loss Alarm” section on page 1-10
- “Feature Interactions” section on page 1-10
- “Feature Provisioning Commands” section on page 1-10

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 BTS 10200 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.

Caution On the BTS 10200, 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 BTS 10200 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 3-134 for a general description of this provisionable service.

911 Overflow Announcement

The system plays an announcement when all circuits to the emergency center are busy and the emergency call cannot be completed to the emergency center. An example of an announcement for this feature is, "We are experiencing 911 difficulties. Please hang up and dial 0 to reach an operator for emergency assistance." The announcement is applied when the announcement resource is available and applicable. For the specific cause code and announcement ID, see "Appendix A, Cause Code to Announcement ID Mappings" in the *Cisco BTS 10200 Softswitch Provisioning Guide*.

Emergency 911 Trunk Connection Loss Alarm

The BTS 10200 is capable of generating a critical alarm of when an emergency trunk resource becomes remotely or locally blocked. This alarm will be raised when any of the following events occurs:

- The gateway becomes unreachable.
- The emergency trunk termination is administratively made OOS through CLI commands on the BTS 10200.
- The emergency trunk termination is remotely or locally blocked.

This feature is applicable only to emergency trunks of type CAS, SS7 and ISDN. The EMERGENCY-TRUNK-GROUP token in the applicable trunk group table must be provisioned to support this feature. For CAS trunk groups, the E911 / EMERGENCY-TRUNK token must also be provisioned.

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

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

Operator Services

The BTS 10200 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 1-11
- Types of Services, page 1-11
- Busy Line Verification (BLV) and Operator Interrupt (OI) Services, page 1-12

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 subscribers lines connected to a BTS 10200 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 3-134 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.

Feature Provisioning Commands

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

Network Services

The BTS 10200 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 Outbound SETUP Messages	
Dialing Parity (IntraLATA Toll Presubscription)	FSD-20-24-0040 TR-TSY-000693
Local Number Portability (LNP)	ATIS/T1S1 T1.TRQ-02-2001
RFC 2833 DTMF Relay—Call Agent Controlled Mode	Section 7.1.1 of PKT-SP-NCS1.5-I01-050128 Section 7.1.1 of PKT-SP-TGCP1.5-I01-050128
SIP Triggers	
Split-NPA	INC97-0404-016
T.38 Fax Relay, Modem, and TDD Handling	IETF RFC 2833 ITU-T *Recommendation T.38*
ENUM Capability	

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 BTS 10200 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.

Note These toll-free (8XX) features can be made available to all subscriber lines connected to a BTS 10200 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 3-134 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.

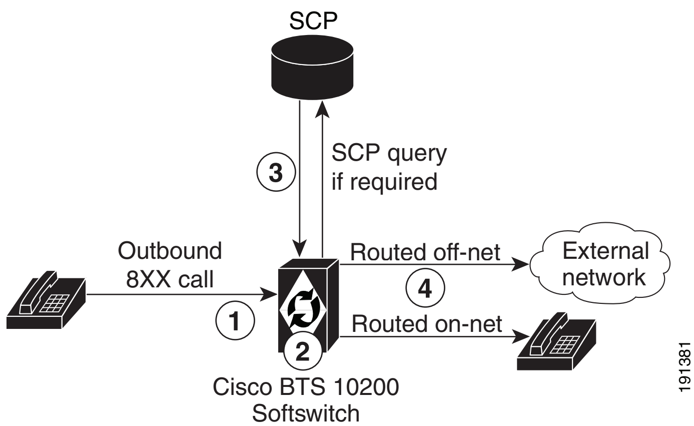
Note For an incoming 8XX call that has a network-specific NOA (based on GR-317), when the system finds the record in the internal database, it assigns the value 2 (TOLL_FREE_LOCAL) to the Database Query Type1 field in the resulting call detail record (CDR).

4. If no internal record is found, the next action depends on how the NOA token is provisioned in the dial plan table. If NOA is provisioned as NATIONAL (the default value), the AIN FS performs an external service control point (SCP) query. If a route is found, the CA completes the call. Otherwise the call is released.

Note For an incoming 8XX call that has a network-specific NOA, the system does not attempt an external query. The call is released with release cause No Route to Destination.

Figure 1-1 shows the processing of an outbound 8XX call placed by a subscriber.

Figure 1-1 Processing of an Outbound 8XX Call

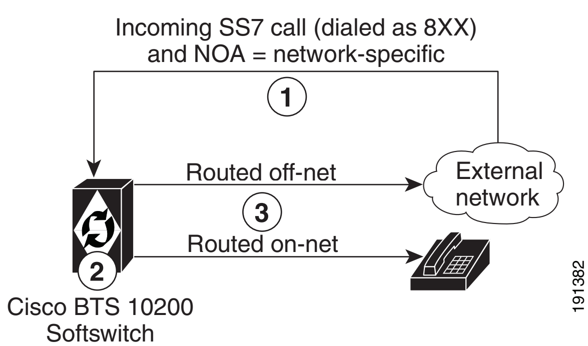


Notes for Figure 1-1

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 BTS 10200 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 BTS 10200 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 BTS 10200 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.

SCP-Based Toll-Free Services

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

The BTS 10200 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 BTS 10200 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 "Viewing Active Calls" information in the *Cisco BTS 10200 Softswitch Operations and Maintenance Guide*.

Alerting Notification to Third-Party Feature Server

The BTS 10200 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 feature supports advertising of an externally addressable FQDN to an external third-party feature server when necessary. 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 BTS 10200. Developers of applications for the 3PTYFS who are interested in the interface specifications should contact their Cisco account team.

Call Flow

The call flow works as follows:

1. The BTS 10200 receives signaling for an incoming call, and sets up the call to the subscriber line.
2. At the CALL_ACCEPTED trigger detection point (TDP) in the call, the BTS 10200 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 BTS 10200 continues with normal call processing regardless of any response from the 3PTYFS.

Prerequisites

The BTS 10200 locates the 3PTYFS via a TSAP address that is provisioned in the BTS 10200 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 BTS 10200 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 1-18.

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

Caution The data in the FCP attachment generated by the BTS 10200 is plain ASCII text and is not encrypted. The security of the connection between the BTS 10200 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 BTS 10200. This is true even if the feature is provisioned globally (at the office level) on the BTS 10200.

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 BTS 10200 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 BTS 10200 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 BTS 10200. However, the BTS 10200 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 BTS 10200 software installation. The choice of data depends on whether the 3PTYFS will be deployed in the same private (internal) management network as the BTS 10200, or in a public network.

- If the 3PTYFS is deployed in the same private (internal) management network as the BTS 10200, the 3PTYFS can obtain the IP address of the BTS 10200 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 BTS 10200), the 3PTYFS must reach the BTS 10200 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 BTS 10200.

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.

Note 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.

Feature Provisioning Commands

For provisioning steps, see the Alerting Notification provisioning procedure in the *Cisco_BTS 10200 Softswitch Provisioning Guide*.

Calling Party Number Options for Outbound SETUP Messages

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

Option to Send Billing DN as CPN for Outbound Calls

The system has a provisionable option for sending a subscriber billing DN (or main DN of a PBX subscriber) as the CPN in outbound 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 outbound 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 outbound SETUP message. For PBX, the system sends the individual subscriber DN (received in the SETUP message) as the CPN in the outbound SETUP message.

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 outbound SETUP messages on outgoing emergency calls. This is provisionable using the SEND-BDN-FOR-EMG token in the Subscriber table.

Note 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 outbound 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 outbound SETUP message. For PBX, the system sends the individual subscriber DN (received in the SETUP message) as the CPN in the outbound setup message.

The system sends the main subscriber number, if it is provisioned, according to the following rules:

- If an ISDN PBX has main-sub-id provisioned with send-bdn-for-emg=N, send-bdn-as-cpn=N, and it is an EMG call, then the DN1 of the main-sub-id is sent as the calling party number.
- If an ISDN PBX has main-sub-id provisioned with send-bdn-for-emg=Y, send-bdn-as-cpn=N, and it is an EMG call, then the billing-dn of the main-sub-id is sent as the calling party number.

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 BTS 10200 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, page 1-21
- Redirection of a Basic or Tandem Call, page 1-22

Redirection by a Subscriber Phone

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

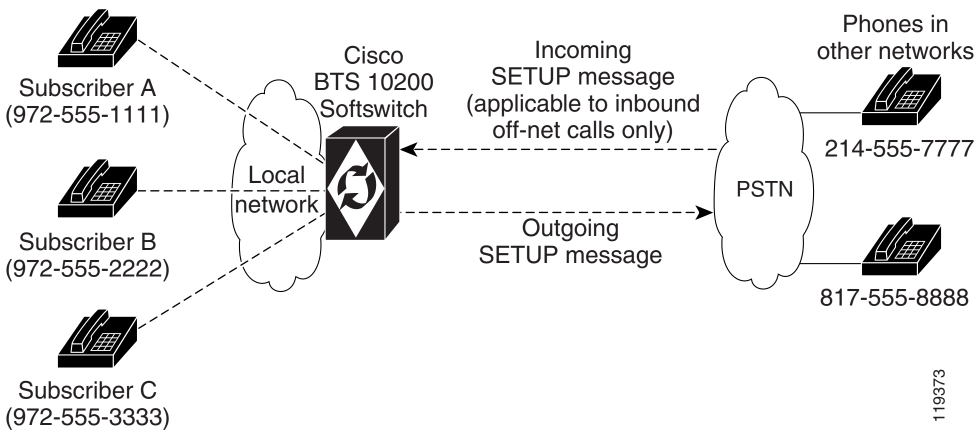


Table 1-4 Provisioning SEND-RDN-AS-CPN in TRUNK-GRP Table for Redirection by Subscriber Phone

Scenario (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-net to off-net call (either of the following): A -> B -> fwd -> C -> fwd -> off-net A -> C -> fwd -> off-net	CPN= 972-555-1111 RDN= 972-555-3333	Y	Overwrite CPN with RDN	CPN= 972-555-3333 RDN= 972-555-3333
		N	Do not change CPN	CPN= 972-555-1111 RDN= 972-555-3333

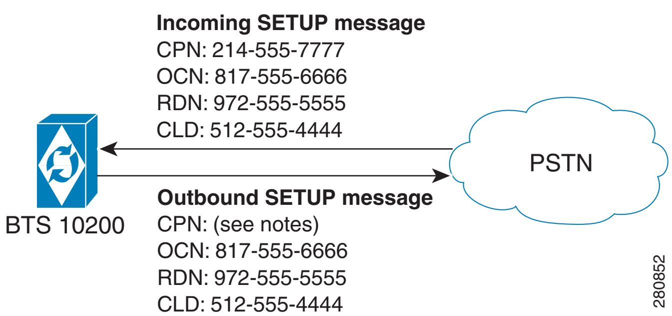
Table 1-4 Provisioning SEND-RDN-AS-CPN in TRUNK-GRP Table for Redirection by Subscriber Phone (continued)

Scenario (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)
Off-net to on-net to off-net call: Inbound off-net call -> B -> fwd -> off-net **Note** In this example, the existing RDN (from the incoming SETUP message) is 817-555-8888. The new RDN is the DN of the forwarding phone, Subscriber B—972-555-2222.	CPN= 214-555-7777 RDN= 817-555-8888 (from incoming SETUP message)	Y	Overwrite CPN with RDN	CPN= 972-555-2222 RDN= 972-555-2222
		N	Do not change CPN	CPN= 214-555-7777 RDN= 972-555-2222

Redirection of a Basic or Tandem Call

Figure 1-4 illustrates the redirection of a basic or Tandem call.

Figure 1-4 Example of Redirection of Basic or Tandem Call



Notes for Figure 1-4

- CPN = calling party number
- OCN = original called number
- RDN = redirecting number
- CLD = called number
- In this scenario, The SETUP message for the incoming call has a CLD (called number) 512-555-4444. If this number does not correspond to a subscriber, the BTS 10200 routes the call back to the PSTN according to provisioning in the dial plan. By default, the data in the outbound SETUP message (the message sent from the BTS 10200 to the PSTN) is the same as the data in the incoming SETUP message. However, if the SEND-RDN-AS-CPN parameter in the TRUNK-GRP table is set to Y, and the RDN is available in the incoming SETUP message, the system replaces the CPN value with the RDN value. Table 1-5 explains how to provision the SEND-RDN-AS-CPN token for call redirection.

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

Scenario (See Figure 1-4)	Content of Incoming SETUP Data (Example)	Value Provisioned for SEND-RDN-AS-CPN (Default = N)	Effect On Outbound SETUP Message	Content of Outbound SETUP Data (Example)
Basic or Tandem call with RDN available in the incoming SETUP message	CPN= 214-555-7777 RDN= 972-555-5555	Y	Overwrite CPN with RDN	CPN= 972-555-5555 RDN= 972-555-5555
		N	Do not change CPN	CPN= 214-555-7777 RDN= 972-555-5555
Basic or Tandem call with RDN *not available* in the incoming SETUP message	CPN= 214-555-7777 *RDN not available*	Y	Do not change CPN	CPN= 214-555-7777 RDN not available
		N	Do not change CPN	CPN= 214-555-7777 RDN not available

Dialing Parity (IntraLATA Toll Presubscription)

The BTS 10200 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

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 BTS 10200 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.

Note The LNP feature can be made available to all subscribers lines connected to a BTS 10200 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 3-134 for a general description of this provisionable service.

The BTS 10200 supports the LNP function as follows:

- Ranges/blocks of ported numbers are provisionable in the BTS 10200, 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 BTS 10200 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 Generic Address 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 Guide*.

To provision LNP options, see the appropriate procedures in the *Cisco BTS 10200 Softswitch Provisioning Guide*:

- Local Number Portability (LNP) for ANSI/North America
- Local Number Portability (LNP) ITU Local BTS Database Query
- SS7 Provisioning

RFC 2833 DTMF Relay—Call Agent Controlled Mode

The system supports Call Agent (CA) controlled mode for dual tone multifrequency (DTMF) relay based on RFC 2833. During call setup, the CA (the BTS 10200) can authorize an embedded multimedia terminal adapter (eMTA) or media gateway (MGW) to invoke RFC 2833 DTMF relay procedures.

Note Prior to Release 5.0, RFC 2833 DTMF relay was not controlled by the Call Agent (the BTS 10200), therefore invocation of RFC 2833 DTMF relay was based on the configuration in the individual eMTA or MGW.

This feature affects MGCP, NCS, TGCP, SIP, and H.323 gateways and endpoints connected to the BTS 10200. The implementation is compliant with the following requirements from industry standards:

- Section 7.1.1 of PKT-SP-NCS1.5-I01-050128
- Section 7.1.1 of PKT-SP-TGCP1.5-I01-050128

Interfaces

The following interfaces are supported for this feature:

- NCS-based eMTA subscribers
- MGCP-based and TGCP-based TGWs (SS7, ISDN, CAS)
- MGCP-based subscribers

Protocol Interworking

This feature can be invoked on the following types of calls:

- MGCP(subscribers and trunks) <-> NCS(subscribers)
- MGCP(subscribers and trunks) <-> H.323(endpoints and trunks)
- NCS(subscribers) <-> H.323(endpoints and trunks)
- MGCP(subscribers and trunks) <-> SIP(subscribers and trunks)
- NCS(subscribers) <-> SIP(subscribers and trunks)
- SIP(subscribers and trunks) <-> H.323(endpoints and trunks)
- MGCP(subscribers and trunks) <-> MGCP(subscribers and trunks)
- H.323(endpoints and trunks) <-> H.323(endpoints and trunks)
- SIP(subscribers and trunks) <-> SIP(subscribers and trunks)
- NCS(subscribers) <-> NCS(subscribers)

The BTS 10200 sends the RFC 2833 DTMF relay parameters to the MGW or endpoint when setting up the initial call and also when setting up features involved in the call, for example forwarded calls.

In addition to configuring the BTS 10200, you must also configure the gateways and endpoints appropriately for the desired features.

Availability of Legacy Behavior

An endpoint can send RFC 2833 DTMF parameters in the Session Description Protocol (SDP) message based on the configuration in the endpoint and without control of the CA. This legacy behavior can be preserved by accepting the default values of the following parameters in the BTS 10200 database:

- In the qos table, dtmf-telephone-event-enabled defaults to N.
- In the h323-tg-profile and h323-term-profile tables, rfc2833-payload defaults to 98.

- In the mgw-profile table, dtmf-telephone-event-supp defaults to AUTO.

Limitations

This section discusses limitations on the implementation of this feature.

Limitations Based on Interactions with Out-Of-Band Digit Transmission

The BTS 10200 assigns a higher priority to the authorization of RFC 2833 DTMF relay than to the authorization of out of band (OOB) digits. If both RFC 2833 and OOB support are enabled in the qos and mgw-profile tables, the system sends RFC 2833 authorization but not OOB authorization. However, the actual implementation of RFC 2833 or OOB implementation depends on the configuration of the two endpoints in the call. If an endpoint specifically requests the use of OOB digits (for example, a SIP subscribe/notify case), the BTS 10200 requests OOB digits also.

Limitations on CA Control

The implementation of RFC 2833 DTMF relay in a specific call depends on payload negotiation by the two endpoints in the call. The following two cases illustrate the limitations on the control exercised by the CA:

- If the BTS 10200 sends authorization during call setup, but the two endpoints cannot successfully negotiate payload parameters, RFC 2833 DTMF relay is not performed.

  For MGCP, NCS, and SIP endpoints, payload negotiation is done through SDP exchange. For H.323 endpoints it is done through H.245 messages.
- If the BTS 10200 *does not send* RFC 2833 DTMF relay authorization during call setup, but one of the endpoints sends SDP messages containing RFC 2833 attributes, the BTS 10200 passes the attributes through to the other endpoint. In such a case. the two endpoints might negotiate payload parameters, and RFC 2833 DTMF relay could be performed.

Limitations on Interoperability with Other NEs

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

Caution Some features involve the use of other network elements (NEs) deployed in the service provider network, for example, gateways, media servers, SIP devices, and eMTAs. See the "Component Interoperability" section of the *Cisco BTS 10200 Softswitch 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 BTS 10200 Release 5.0 software. Earlier or later releases of platform software might be interoperable and it might be possible to use other functions on these platforms. The list in 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 BTS 10200.

Conditions for Sending LCO for RFC 2833 Telephone-Events

The following conditions apply to BTS 10200 messaging for RFC 2833 telephone-events to the originating and terminating sides in the call. The implementation of these capabilities depends upon the provisioning of certain parameters in the database.

- If the originating side does not support RFC 2833, the BTS 10200 does not send telephone-events to the terminating side even if the terminating side supports RFC 2833.
- If the originating side supports RFC 2833, the BTS 10200 sends telephone-events to the terminating side before checking whether the terminating side supports RFC 2833.
- If the originating side supports RFC 2833, but the DTMF flag in the mgw-profile table is set to N, the BTS 10200 does not send telephone-events to either side.

If the originating endpoint is NCS, MGCP, or TGCP based, the BTS 10200 sends a create-connection (CRCX) message containing the local connection option (LCO) for RFC 2833 telephone-events if the originating endpoint is provisioned in the BTS 10200 with:

- qos table—dtmf-telephone-event-enabled=y
- mgw-profile table—One of the following is true: (1) dtmf-telephone-event-supp=y, *or* (2) dtmf-telephone-event-supp=auto and the endpoint reports the telephone-event codec in the AUEP ACK message.

The system can also send a CRCX message containing the LCO for RFC 2833 telephone-events if the terminating and originating endpoints are both MGCP based, or if both endpoints are NCS or TGCP based, and the following parameters are provisioned in the BTS 10200:

- Both endpoints are provisioned with codec-neg-supp=y in the mgw-profile table.
- The terminating endpoint is provisioned with:
  - qos table—dtmf-telephone-event-enabled=y
  - mgw-profile table—One of the following is true: (1) dtmf-telephone-event-supp=y, *or* (2) dtmf-telephone-event-supp=auto and the endpoint reports the telephone-event codec in the AUEP ACK message.

SIP Triggers

This section explains the scope of the SIP Triggers feature and provides a technical description of it.

Scope for Release 5.0

Release 5.0 of the BTS 10200 supports the following SIP triggers:

- Off-hook trigger with provisionable delay (OHD):
  - Supported for MGCP and NCS subscribers
  - *Not* supported for SIP endpoints (SIP subscribers)
- Termination Attempted Trigger 1 (TAT_1) and Termination Attempted Trigger 2 (TAT_2):
  - Supported for MGCP and NCS subscribers.
  - Release 5.0, MR1 and later. Supported for SIP endpoints (SIP subscribers) when the incoming call is based on a DN.

Feature Limitations

SIP triggers are not supported for Centrex subscribers.

The system does not support the OHD trigger for SIP subscribers.

The BTS 10200 does not invoke any SIP triggers for SIP subscribers on incoming calls that are based on an address of record (AOR).

Technical Description of SIP Triggers

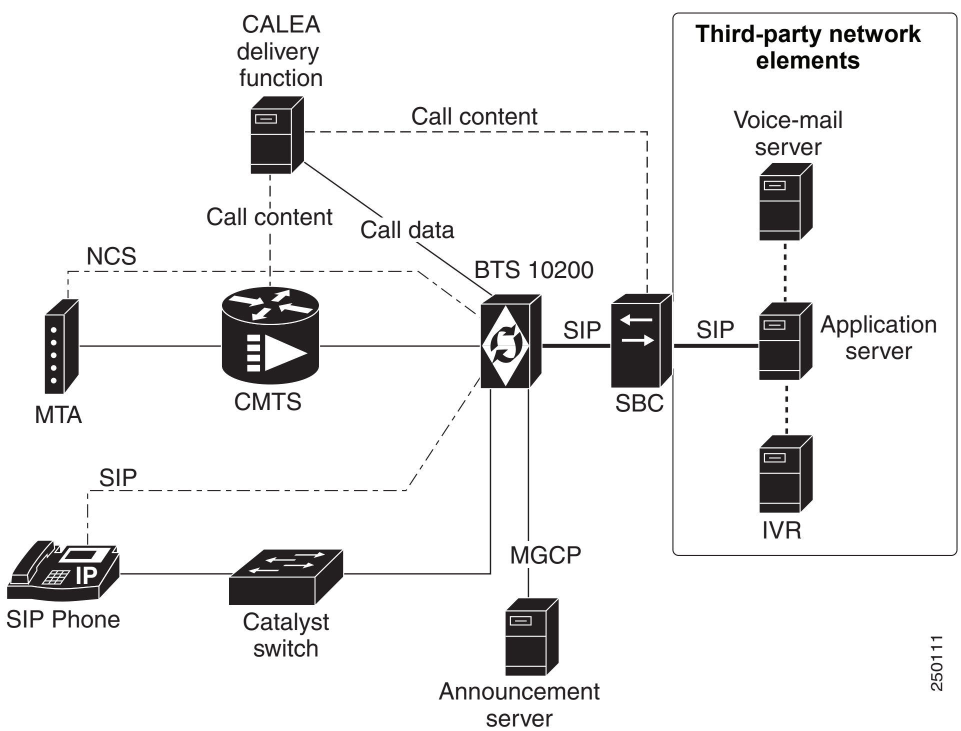
The SIP Triggers feature uses the SIP protocol, with some extensions, to enable the BTS 10200 to interoperate with third-party application servers so that Multi-Service Operators (MSOs) can provide customers with enhanced features and services. The triggers can be used by the third-party servers to provide originating services (such as voice dial) when a subscriber places a call, and enhanced terminating services (such as TV caller ID and custom ringback) when a subscriber receives a call.

The BTS 10200 supports multiple application servers. Application servers are provisioned per subscriber origination and subscriber termination. This section describes the triggers that enable this interoperation with the third-party application servers.

You can provision SIP triggers on an individual subscriber level on the BTS 10200.

Figure 1-5 shows a typical network architecture for SIP triggers, including the connection between the BTS 10200 and the third-party application server. The capability to provide TAT_1 and TAT_2 SIP triggers to SIP phones is available in Release 5.0 MR1 and later.

Figure 1-5 Typical Network Architecture for SIP Triggers



Acronyms for Figure 1-5

CALEA—Communications Assistance for Law Enforcement Act

NCS—Network-based call signaling

SBC—Session border controller

CMTS—Cable modem termination system

IVR—Interactive voice response

Terminology Used in This Section

The following terminology is used in this section:

- TAT_1 and TAT_2—Termination attempt triggers, collectively referred to as TAT in this document. These triggers occur at different points in the call; TAT_1 occurs before TAT_2, and can be used by an external application server to provide specific services at specific points in the call. These triggers are provisioned as fname=TAT_1 and fname=TAT_2.
- OHD—Off-hook trigger with provisionable delay. The OHD trigger for each subscriber can be designated as off-hook immediate (OHI) or off-hook delayed (OHD). You provision this through the offhook-trigger-type parameter in the Subscriber table:
  - OHD occurs when a provisionable timer runs out after the caller goes off-hook.
  - OHI occurs immediately after the caller goes off-hook.

Tip There is an OHD trigger provisioned as fname=OHD in the Feature table, and there is a delayed or immediate designation provisioned as offhook-trigger-type=OHD or OHI in the Subscriber table.

Off-Hook Trigger (Delayed and Immediate)

The OHD trigger occurs either immediately after the user goes off-hook (if offhook-trigger-type=OHI) or after a delay set through a configurable timer (if offhook-trigger-type=OHD).

- If off-hook immediate is provisioned, the BTS_10200 establishes a connection to the external application server provisioned for that combination of subscriber and trigger and sends the call to the server immediately after the caller goes off-hook.
- If off-hook delayed is provisioned and the user goes off-hook, dial tone is provided to the subscriber for the configured number of seconds. When the delay timer expires, dial tone is stopped, and the BTS 10200 establishes a connection to the external application server provisioned for that combination of subscriber and trigger and sends the call to the server. If the user starts dialing before the delay timer expires, the BTS 10200 allows digit collection to be completed before establishing the connection to the application server. Depending on the service invoked (for example, dial by name), the application server determines the desired called party and sends the call back to the BTS 10200 to continue originating processing.

The BTS 10200 inserts two route headers in the INVITE that is sent to the application server:

- Topmost route header—Intended for the application server so that it can identify the logic to be executed on the application server. This header can be provisioned on the BTS 10200 per subscriber or set with a default for all OHD subscribers.
- Return route header—Identifies the necessary call session and processing information when the INVITE is returned to the BTS 10200. This route header must be returned unchanged to the BTS 10200 in the INVITE that is sent from the application server to the BTS 10200.

Termination Attempt Triggers (TAT_1 and TAT_2)

The TAT trigger occurs when a call terminates to a BTS 10200 subscriber who has one or both TAT triggers (TAT_1 and TAT_2) enabled. The BTS 10200 sends the terminating call to the external application server before ringing the subscriber. The application server may provide services such as screening or custom ringback. If the application server determines that the call should be offered to the subscriber, it sends the call back to the BTS 10200 to continue termination processing.

The TAT triggers operate as follows:

- The TAT_1 trigger takes precedence over all other BTS 10200 terminating features at the TERMINATION_ATTEMPT_AUTHORIZED trigger detection point. However, if the CNAM TCAP query is provisioned for a subscriber, it is performed before the TAT_1 trigger, and the name is provided to the application server. The BTS 10200 honors the calling name it receives from the application server and does not launch a CNAM query if the name is presented by the application server.
- The TAT_2 trigger is the last feature invoked at the TERMINATION_ATTEMPT_AUTHORIZED trigger detection point.

A TAT trigger is not disabled even if the user is in an Emergency call. However, a return INVITE from the application server fails at the BTS 10200 if the user is in an Emergency call.

As it processes a TAT, the BTS 10200 manages the following events:

- Successful application server invocation
- Unsuccessful application server invocation
- Application server terminates call
- Application server sends call back to the BTS 10200 for termination processing

The system populates two Route headers in the INVITE it sends to the application server. The contents of the Route headers are controlled by provisioning. The provisioning of the first Route header is determined by the requirements of the application server and that of the second is determined by the requirements of the BTS 10200. The following examples illustrate the format of the Route field, with both the first Route header and return Route header shown in the example:

Examples—Route Message:

```
Route: <topmost route>,<return route>;service-ref=SCM0579081256

Route: <sip:TAT_1-app@APP_SERVER1.serviceprovider.com;lr>,
<sip:TAT_1@bts1.serviceprovider.com;lr>;service-ref=SCM0579081256

Route: <sip:TAT_2-app@APP_SERVER1.serviceprovider.com;lr>,
<sip:TAT_2@bts1.serviceprovider.com;lr>;service-ref=SCM0537491333
```

Note The meaning of lr in the route message is loose routing. See RFC 3261 for a description.

Subscriber Features

The following are examples of services that could be offered in conjunction with the TAT and OHD triggers if there ia an appropriate application server in the network:

- Voice menu (OHD trigger)
- Voice dialing (OHD trigger)

- Multi-party voice dialing (OHD trigger)
- TV caller ID, with or without picture (TAT trigger)
- Custom ring-back tone (TAT trigger)
- Enhanced voicemail UI (OHD trigger)
- Message status (OHD trigger)
- Missed call status (OHD trigger)
- Voicemail screening (specific digit string (vertical service code))
- Smart call forward (TAT trigger)
- Smart call return (OHD trigger)
- Dialpad sound effects (specific digit string)
- Multi-ring call forward (TAT trigger)
- Click to dial (third-party call control [3PCC] mechanisms)

Note Support of subscriber services is not limited to those identified in the preceding list.

Failover Behavior

During transitions (for example, while the call to the application server is being placed or while the call from the application server is placed to the destination), a failover of the call agent or feature server is likely to cause the call to be dropped.

If a call is connected to the application server, or if a call is connected end to end, a failover of the call agent or feature server does not disrupt the call.

Feature Interactions for OHD Trigger

The OHD trigger is supported for MGCP and NCS subscribers, but not for SIP subscribers.

Features that are provided by the application server interwork with the features provided by the BTS 10200. Table 1-6 describes the feature interactions for the OHD trigger.

Table 1-6 Feature Interactions for OHD Trigger

Feature	Abbreviation	Interaction
Vertical Service Codes (VSCs): • Activation, deactivation, and interrogation features, including the use of VSCs, for example AC_ACT, AC_DEACT, CBLK, CCW, COT, CFUA, CFBI, CNAB, CNDB, DND_ACT, and DND_DEACT. • VSCs dialed to access feature management functions, for example PS_MANAGE, PS_O, CIDSD, CIDSS, VM_ACCESS. • Initial VSCs entered by the handset user to invoke AC and AR.	(various)	If the user dials the VSC during the dial tone, the BTS 10200 collects the VSC digits and then forwards the call to the application server. If the user dials the VSC after the BTS 10200 establishes the call to the application server, the application server collects the VSC digits. The application server is configured to send the REFER back to the BTS 10200 with the VSC if it is not capable of handling this VSC. When the REFER is received by the BTS 10200, the BTS 10200 provides origination processing. During origination processing, if more digits must be collected, the BTS 10200 does so by playing appropriate tones directly at the endpoint. After the process (activation, deactivation, interrogation, or feature management) is complete, the BTS 10200 does not invoke SIP triggers to the application server again.
Toll Free	8xx	OHD trigger takes precedence.
Emergency Service	911	Emergency calls are routed to the application server if the BTS 10200 Call Agent Configuration table is provisioned to do so: `add ca-config type=EMG-ROUTE-TO-AS; datatype=BOOLEAN; VALUE=Y;` If EMG-ROUTE-TO-AS is set to N (default), the BTS 10200 processes the call locally and does not invoke the application server. If EMG-ROUTE-TO-AS is set to Y and the dialed digits match the digit string provisioned in the **emergency-number-list** table, the BTS 10200 can route the call to the application server. The actual treatment of the call is as follows: • If the user dials the emergency number before the call is connected to the application server, the BTS 10200 processes the call locally and does not invoke the application server. • If the user dials the emergency number after the call is connected to the application server, the application server is expected to collect the digits and send a REFER or return INVITE to the BTS 10200. When the BTS 10200 receives the REFER or return INVITE, it makes the call to the emergency number.
Automatic Callback	AC	When the subscriber enters the VSC on the handset, the BTS 10200 invokes the OHD trigger. When the BTS 10200 calls back the subscriber, the callback does not invoke OHD. When the BTS 10200 sets up the final call to the remote phone, it treats this as a new call and invokes the OHD trigger to the application server.
Automatic Recall	AR	Same as AC.

Table 1-6 Feature Interactions for OHD Trigger (continued)

Feature	Abbreviation	Interaction
Class of Service	COS	OHD takes precedence, and COS screening is applied after the return INVITE is received at the BTS 10200. Authorization Code and Account Code features are already being supported by use of IVR. Therefore, these features, if needed, can be configured on the BTS 10200 to be supported by IVR.
Call Transfer	CT	A second origination attempt also goes through the application server.
Hotline	HOTLINE	If HOTLINE and OHD are active, the hotline number is expanded and the number is included in the request URI user part sent to the application server.
Hotline Variable	HOTV	Same as Hotline.
Limited Call Duration	LCD	OHD takes precedence and the process of checking the prepaid server occurs after the return INVITE is received from the feature server.
Local Number Portability	LNP	OHD takes precedence and a local number portability query is invoked after the return INVITE is received from the feature server.
Outgoing Call Barring	OCB	OHD takes precedence and OCB is applied after the application server sends the call back to the BTS 10200.
Speed Call - 1 /2 digit	SC1D SC2D	The call is routed to the application server with the speed-dial digits. When the return INVITE is received from the application server, the BTS 10200 provides the speed call function.
Three Way Call	TWC	The second origination attempt goes to the application server.
Three Way Call Deluxe	TWCD	Same as TWC.
Usage Sensitive Three Way Call	USTWC	Same as TWC.
Warmline	WARMLINE	If warm line and OHD are provisioned, the warmline number is included in the INVITE message (user part of request URI) to the application server. **Note** If a subscriber is provisioned for both warmline and OHD, the delay for OHD should not be used. Either the subscriber dials before the warmline timeout and the dialed number is provided to the application server, or the warmline timeout occurs, and the warmline number is provided to the application server. The application server does not collect digits. Off Hook Immediate is not available. Speak to Dial is not available. Similarly, the offhook delay timer for OHD is not used if hotline is also active. If both OHD and hotline are active, the BTS 10200 immediately sends an INVITE to the application server that includes the provisioned hotline number. The user does not receive an initial dial tone.

Feature Interactions for TAT_1 and TAT_2 Triggers

TAT_1 and TAT_2 triggers are supported for MGCP and NCS subscribers in the initial release of Release 5.0, and also supported for SIP subscribers in Release 5.0 Maintenance Release 1 (MR1) and later.

Features that are provided by the application server interwork with the features provided by the BTS 10200. Table 1-7 describes the feature interactions for the TAT_1 and TAT_2 triggers.

Caution It is the responsibility of the application server to honor the privacy settings of the calling party based on the information received in the message from the BTS 10200 for the TAT_1 and TAT_2 triggers.

Table 1-7 Feature Interactions for TAT_1 and TAT_2 Triggers

Feature	Abbreviation	Interaction for MGCP and NCS Subscribers	Interaction for SIP Subscribers (Release 5.0 MR1 and Later)
Emergency-Service	911	If the called user is already in an emergency call, and the BTS 10200 receives an additional incoming call for this user, the TAT trigger is invoked, and the BTS 10200 sends the new call information to the application server. The application server might send a return INVITE. If the user is in the emergency call when the BTS 10200 receives the return INVITE, the BTS 10200 fails the new incoming call.	Same as for MGCP/NCS
Anonymous Call Rejection	ACR	The order of precedence for the triggers is TAT_1, ACR, TAT_2. If the call is rejected by ACR, TAT_2 is not invoked.	If the feature is provided by the SIP endpoint, the TAT_1 and TAT_2 triggers are invoked normally. If the feature is provided by the BTS 10200, the interaction is the same for SIP subscribers as for MGCP/NCS subscribers.
Busy Line Verification	BLV	BLV takes precedence and the TAT trigger is not invoked.	BLV is not supported for SIP subscribers.
Call-Forwarding-Busy/ Call-Forward-No-Answer/ Call Forwarding Combined/ Call-Forwarding-Unconditional - Invocation	CFB/CFNA/ CFC/CFU Invocation	The order of precedence for the triggers is TAT_1, CFU, TAT_2, [CFB/CFNA/CFC].	Same as for MGCP/NCS **Note** The reminder ring is not provided for SIP subscribers, but this does not change the feature interaction.
Calling Identity Delivery on Call Waiting	CIDCW	The TAT takes precedence over CIDCW, and the return INVITE from application server results in CW.	For SIP subscribers, CIDCW is a SIP endpoint feature, not a feature of the BTS 10200.

Table 1-7 Feature Interactions for TAT_1 and TAT_2 Triggers (continued)

Feature	Abbreviation	Interaction for MGCP and NCS Subscribers	Interaction for SIP Subscribers (Release 5.0 MR1 and Later)
Calling Name Delivery	CNAM	The CNAM query is performed before the TAT is invoked.	Same as for MGCP/NCS
Calling Number Delivery	CND	The application server is a trusted entity; therefore, the calling number is always delivered to the application server.	Same as for MGCP/NCS
Call Waiting	CW	The TAT takes precedence over CW, and the return INVITE from application server results in CW.	For SIP subscribers, CW is a SIP endpoint feature, not a feature of the BTS 10200.
Call Waiting Deluxe	CWD	Same as CW.	Same as CW.
Do Not Disturb	DND	TAT_1 takes precedence and the BTS 10200 invokes DND after it receives the return INVITE from the application server. **Note** DND takes precedence over TAT_2.	If the feature is provided by the SIP endpoint, the TAT_1 and TAT_2 triggers are invoked normally. If the feature is provided by the BTS 10200, the interaction is the same for SIP subscribers as for MGCP/NCS subscribers.
Distinctive Ringing Call Waiting	DRCW	The order of precedence for the triggers is TAT_1, TAT_2, DRCW.	Same as for MGCP/NCS
No Solicitation Announcement	NSA	The order of precedence for the triggers is TAT_1, NSA, TAT_2.	Same as for MGCP/NCS
Privacy Screening	PS	The order of precedence for the triggers is TAT_1, PS, TAT_2.	Same as for MGCP/NCS
Remote Call Forwarding	RCF	The TAT_1 trigger takes precedence over RCF.	Same as for MGCP/NCS
Selective Call Acceptance/ Selective Call Forwarding/ Selective Call Rejection	SCA/SCF/SCR	TAT_1 takes precedence, and the BTS 10200 invokes SCA/SCF/SCR after it receives the return INVITE from the application server. **Note** SCA/SCF/SCR takes precedence over TAT_2.	Same as for MGCP/NCS
Voicemail	VM	The order of precedence for the triggers is TAT_1, VM, TAT_2.	Same as for MGCP/NCS
Voicemail Always	VMA	The TAT_1 trigger takes precedence over VMA.	Same as for MGCP/NCS

Billing

The following call detail block (CDB) billing fields report the SIP originating and terminating context IDs. Details of these CDBs are provided in Appendix A of the *Cisco BTS 10200 Softswitch Billing Guide*.

- CDB field 232, SIP Originating Context Id
- CDB field 233, SIP Terminating Context Id

The following call data fields are provided for SIP triggers. Details regarding the information in these fields are provided in Chapter 2, "Feature Server Derived Call Data," of the *Cisco BTS 10200 Softswitch Billing Guide*.

- SIP Off-Hook Trigger
- SIP Termination Attempt Trigger

Provisioning Commands

The provisioning steps for this feature are based on the provisioning steps for SIP triggers for NCS endpoints in the *Cisco BTS 10200 Softswitch Provisioning Guide*, with the following exception—SIP triggers for SIP endpoints include only the terminating attempt triggers (TAT_1 and TAT_2), not the off-hook trigger (OHD). When you provision SIP endpoints, include all of the provisioning for TAT-related parameters, and exclude any provisioning for OHD-related parameters.

Error Handling

This section describes how the BTS 10200 operates when it receives 4xx, 5xx, and 6xx responses over the IP Multimedia Subsystem (IMS) Service Control (ISC) interface from the application server.

Table 1-8 lists the response codes and indicates how the BTS 10200 operates when it receives 4xx, 5xx, and 6xx responses over the ISC/application server interface.

Table 1-8 uses the following terms:

- Continue—Indicates that the BTS 10200 processes the call locally. In the case of the OHI or OHD trigger with no digits, the BTS 10200 prompts the user, collects the address digits, and then processes the call locally.
- Error treatment—Indicates that the BTS 10200 connects the user to a configured tone or announcement and then disconnects the call.

Table 1-8 BTS 10200 Management of Error Response Codes

Response Code	OHI or OHD with No Digits	OHD with Digits, TAT
400—Bad Request	Continue	Continue
401—Unauthorized	Continue	Continue
402—Payment Required	Continue	Continue
403—Forbidden	Continue	Error treatment
404—Not Found[1]	Continue	Continue
405—Method Not Allowed	Continue	Continue
406—Not Acceptable	Continue	Continue
407—Proxy Authentication Required	Continue	Continue

Table 1-8 BTS 10200 Management of Error Response Codes (continued)

Response Code	OHI or OHD with No Digits	OHD with Digits, TAT
408—Request Timeout	Continue	Continue
410—Gone	Continue	Error treatment
413—Request Entity Too Large	Continue	Continue
414—Request URI Too Large	Continue	Continue
415—Unsupported Media Type	Continue	Continue
416—Unsupported URI Scheme	Continue	Continue
420—Bad Extension	Continue	Continue
421—Extension Required	Continue	Continue
423—Interval Too Brief	Continue	Continue
480—Temporarily Unavailable	Continue	Continue
481—Call/Transaction Does Not Exist	Continue	Continue
482—Loop Detected	Continue	Error treatment
483—Too Many Hops	Continue	Continue
484—Address Incomplete	Continue	Error treatment
485—Ambiguous	Continue	Continue
486—Busy Here	Continue	Error treatment
487—Request Terminated	Continue	Error treatment
488—Not Acceptable Here	Continue	Continue
491—Request Pending	Continue	Continue
493—Undecipherable	Continue	Continue
500—Server Internal Error	Continue	Continue
501—Not Implemented	Continue	Continue
502—Bad Gateway	Continue	Error treatment
503—Service Unavailable	Continue	Continue
504—Server Timeout	Continue	Continue
505—Version Not Supported	Continue	Continue
513—Message Too Large	Continue	Continue
600—Busy Everywhere	Continue	Error treatment
603—Decline	Continue	Error treatment
604—Does Not Exist Anywhere	Continue	Error treatment
606—Not Acceptable	Continue	Error treatment

1. The application server might send a 404 or 305 response code if the user is not subscribed to the requested service (feature). If a 305 is returned, the FQDN of the contact header received in the 305 must be of the receiving BTS 10200. Receiving the 305 prompts the BTS 10200 to continue to process the call locally without reengaging the application server. If the BTS 10200 receives a 404 response code, it continues to process the call locally. The 404 also might be generated by a downstream node; but, the BTS 10200 is not able to differentiate if the 404 is generated by the application server or the downstream node. Therefore, attempting to process the call locally might fail again. The number of times that the BTS 10200 reattempts this "failed call" can depend on the number of triggers executed before the 404 error is encountered. In the worse case, if a user is configured with OHD, TAT_1, and TAT_2, the BTS 10200 can reattempt the call four times before the caller is connected to error treatment.

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 BTS 10200. 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 BTS 10200. 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)

Note The 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*.

Note 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.

Feature Provisioning Commands

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

T.38 Fax Relay, Modem, and TDD Handling

This section describes the BTS 10200 fax, modem, and Telecommunications Devices for the Deaf (TDD) handling feature.

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

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).

This section covers the following topics:

- Understanding the Fax, Modem, and TDD Handling Feature
- 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
- TDD Handling
- Control Configuration
- Restrictions and Limitations
- Feature Provisioning Commands

Understanding the Fax, Modem, and TDD Handling Feature

Voice Over Internet Protocol (VoIP) converts sounds into data; when converting, tones are lost. To fix the issue of lost tones the ITU developed the T.38 standard. T.38 converts sounds into data without losing these tones. Fax, modem, and TDD communications do not need the same kind of treatment voice calls do. Upspeed carries fax, modem, and TDD calls in-band over a voice call without the overhead of a G.729 codec and voice-enhancing features.

BTS 10200 Call Agents (CAs) manage calls and control media gateways (MGWs). A gateway can be the BTS 10200 itself or another device, like a Multimedia Terminal Adapter (MTA) or an Integrated Access Device (IAD). Using different modes on the BTS 10200, you can set gateways to do the following upon receiving fax, modem, or TDD tones:

- Use (upspeed) the G.711 codec
- Turn off silence suppression, echo cancellation (EC), and voice activity detection (VAD)

Table 1-9 describes the modes supported by the BTS 10200.

Table 1-9 Modes for Fax, Modem, and TDD Handling

Mode	Functions
T.38 Loose	Uses User Defined Protocol Transport Layer (UDPTL) packets to carry fax data; UDPTL is a UDP-based protocol not used outside of T.38 and is the most common transport for T.38. or Uses Transmission Control Protocol (TCP).
T.38 Strict	Uses UDPTL packets to carry fax data; UDPTL is a UDP-based protocol not used outside of T.38 and is the most common transport for T.38. or Uses TCP Requires remote devices (MTAs, IADs) to report their support of T.38 in the Session Description Protocol (SDP) message to the BTS 10200.
GW	Excludes CA; gateway controls how to handle calls. Always supported, even if the gateway does not do anything special for fax. Prevents CA from sending conflicting commands during gateway’s special fax handling.

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 DQOS 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:<previous 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:

```
a=sqn: 0
a=cdsc: 1 audio RTP/AVP 0 18 96
a=cpar: a=fmtp:96 0-16,32-35
       a=cdsc: 2 image udptl t38
```

- 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 1-10.

Table 1-10 Protocol interworking matrix

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

Figure 1-6 shows an example of MGCP and SIP interworking.

Figure 1-6 Example of MGCP and SIP Interworking for T.38 Fax

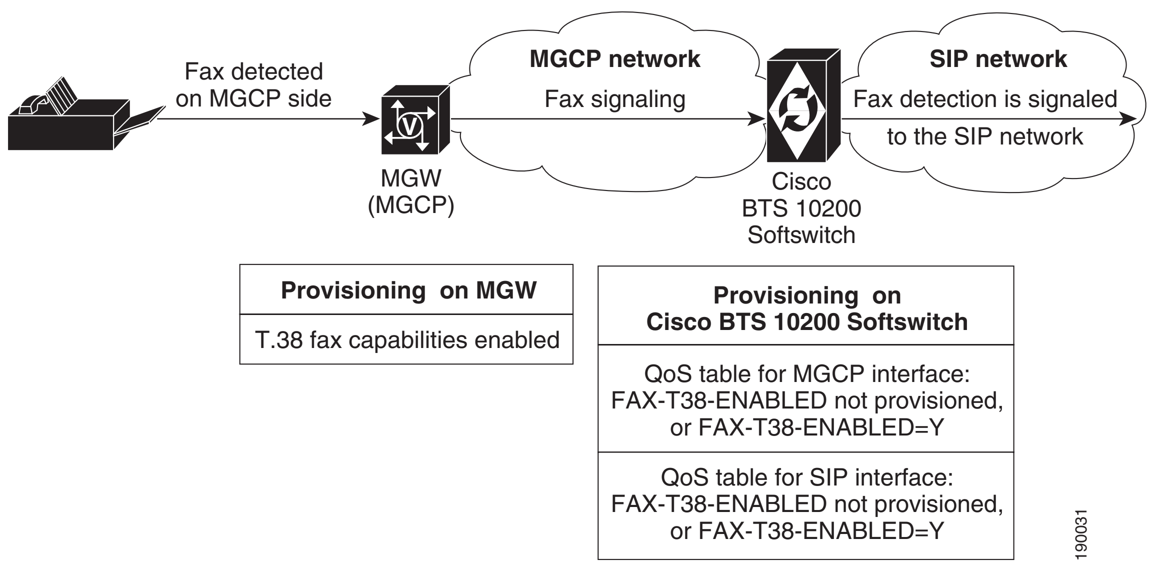


Table 1-11 illustrates how the BTS 10200 uses the T.38 fax protocol when a fax is detected. Usage depends on the value of the QoS fax_t38_enabled token for each endpoint involved in the call, as well as the protocol type of each endpoint. The symbols used in the table are:

- T38—The Cisco BTS 10200 Softswitch uses the T.38 protocol for fax transmission.
- X—The Cisco BTS 10200 Softswitch does not use the T.38 protocol for fax transmission.

- T38*—Since one of the field values in this combination is set to N, the MGCP endpoint involved in this call does not receive the local connection option (L:fxr:fx/t38) in the initial CRCX request from Cisco BTS 10200 Softswitch. However, if the endpoint receives a T.38 SDP from the remote end detecting fax, then it is assumed to support the switch to T.38 media connection.

Table 1-11 T.38 Fax Usage based on QoS Fax-t38-enabled Value

Interface/Protocol (Value)	SIP (Y)	SIP (N)	H.323 (Y)	H.323 (N)	MGCP (Y)	MGCP (N)
SIP (Y)	T38	T38	T38	X	T38	T38*
SIP (N)	T38	T38	X	X	T38*	T38*
H323 (Y)	T38	X	T38	T38	T38	X
H323 (N)	X	X	T38	T38	X	X
MGCP(Y)	T38	T38*	T38	X	T38	X
MGCP (N)	T38*	T38*	X	X	X	X

TDD Handling

Each TDD call has two call legs, one from the hearing user to the relay operator and one from the relay operator to the non-hearing user. The hearing user and the relay operator need the settings involved with a regular voice call. The non-hearing user and the relay operator need the settings that support a TDD device, so the BTS 10200 upspeeds the call between the relay operator and the non-hearing user. To provision TDD handling, set the following options in the MGW Profile table:

- TDD_TONE_SUPP to Y
- MODEM_TONE_SUPP to Y
- FAX_INBAND_METHOD to FT_UPSPEED

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 the following limitations:

- 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 BTS 10200
- 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):

- 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:

- Gateway mode (though the Cisco-proprietary Gateway mode is supported)
- Off mode

If the BTS 10200 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 BTS 10200 does not receive any Gateway mode notification events from the endpoint because it does not request them. The BTS 10200 is not notified of the gateway mode activity. The BTS 10200 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 BTS 10200

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

Cisco recommends that the BTS 10200 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 the BTS 10200 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.

Note 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

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

CMTS Discovery Using the Static Subnet Table

The CMTS Discovery Using the Static Subnet Table feature uses the statically provisioned Subnet table in the BTS 10200 system. The service providers must configure all subnets handled by every CMTS using the Subnet table. The BTS 10200 uses the IP address of the multimedia terminal adapter (MTA) and the Subnet table information to determine the CMTS (AGGR) handling the MTA.

For more information on this feature and the related provisioning procedure, refer to Provisioning CMTS Discovery Using the Static Subnet Table section in the *Cisco BTS 10200 Softswitch PacketCable and Event Message Guide*.

Aggregation Id Subnet

This feature allows a service provider to use the Subnet table to statically configure all subnets handled by every Cable Modem Termination System (CMTS). The Cisco BTS 10200 Softswitch uses the IP address of the embedded Multimedia Terminal Adapter (eMTA) and Subnet table to determine the CMTS handling of a particular eMTA. An eMTA is a residential gateway. A CMTS is an aggregation device for multiple eMTAs.

An effective aggr-id is the aggr-id in effect for a particular eMTA. It identifies the CMTS to which the Cisco BTS 10200 Softswitch sends Dynamic Quality of Service (DQoS) requests for that eMTA. A manual aggr-id is an aggr-id that is provisioned by a service provider. If an aggr-id is provisioned in the Media Gateway table or Subnet table, it is a manual aggr-id.

For more information on the provisioning procedure for this feature, refer to Provisioning the Aggregation ID Subnet section in *Cisco BTS 10200 Softswitch PacketCable and Event Message Guide*.

ENUM Capability

This section describes the Cisco BTS 10200 Softswitch ENUM capability feature introduced in Release 5.0 MR1. This section specifies the feature functions and operations, and contains the following:

- ENUM Support for Routing
- ENUM Routing Use Cases
- BTS 10200 Routing Operations
- Operational Recommendations for Porting Procedures
- ENUM Clients Operation
- Planning
- Operating
- Troubleshooting

Feature Description

ENUM provides a simple query-based mechanism for applications to retrieve data associated with a particular E.164 address. The Cisco BTS 10200 Softswitch supports ENUM queries for on-net routing and LNP data. The following sections describe how the Cisco BTS supports ENUM for SIP routing and LNP queries.

ENUM Support for Routing

The ENUM Capability feature maps the called DN to the Uniform Resource Identifier (URI) to determine the switch on which the called DN resides. It then routes the call on-net. In order to use this feature, the service provider must deploy private or carrier ENUM servers that hold E.164-to-URI mapping for all DNs owned by the service provider.

Performing the ENUM Query

When receiving a call from a subscriber or from another network element, the BTS 10200 determines if it is the final destination of the call. If the BTS 10200 is not the final destination, it performs an ENUM query to determine the destination and obtains the URI to route the call.

The following rules apply to the BTS 10200 ENUM capability feature:

- The BTS 10200 launches the ENUM query after it receives the TCAP LNP query results, unless the same ENUM server is configured as LNP capable.
- The BTS 10200 does not perform ENUM queries for operator calls and casual calls.
- If the service provider does not want to perform ENUM lookup for an NPA-Nxx or wants to use different top-level domains for the ENUM queries based on a different NPA-Nxx, the BTS 10200 can be controlled on a configuration basis at the destination table level.

Routing Upon Receipt of the ENUM Response

This section describes the BTS 10200 routing operation that is based on the response from the ENUM server. In general, responses from the ENUM servers fall into two categories for routing operations:

- The BTS 10200 receives a URI indicating the final destination of the E.164 number.
- The BTS 10200 does not receive any URI information.

Call Routing Based on the Received URI

If the called DN is an on-net subscriber (that is, if it is in the VoIP domain), the ENUM server returns the URI associated with the DN, indicating the address of the destination switch that can terminate the call to the subscriber. The BTS 10200 uses the domain portion of the received URI to determine the on-net path where the call should be routed.

The BTS 10200 retrieves the domain portion from the returned URI and attempts to find the on-net route configured on the BTS 10200 (in the domain2route table). For example, if the ENUM query is launched for DN1 and the ENUM response is DN2@btsX.sp.net, the BTS 10200 ignores the DN2 and uses domain btsX.sp.net to route the call. The BTS 10200 also uses the same domain in the Req-URI field of the outgoing INVITE message (for this example, Req-URI is set to DN1@btsX.sp.net).

Because the BTS 10200 performs a longest match for the received domain with configured routes, service providers may configure only substrings in the domain2route table to keep the number of routing entries to a minimum. For example, the user can configure the BTS 10200 to use one on-net route for bts10.region1.sp.com and another for region1.sp.com (that is, all BTS 10200 nodes in region1 except BTS10).

The BTS 10200 can also specify different policy-based routing features for each domain returned from the ENUM server. For example, Percentage Based Routing and Time of Day Routing can be applied against the domain by specifying **ROUTE-TYPE = ROUTE-GUIDE**.

The user can also specify multiple routes (that is, multiple softswitch trunks) against the received domain and use the BTS 10200 Route Advance feature to select an alternate on-net route in case the first on-net route cannot be used to route the call to its final destination.

The BTS 10200 can use a destination-based route (typically pointing to the PSTN interface, such as an SS7 or SIP trunk group, toward the MGC) for a particular domain received from the ENUM server. This capability can be used in various situations, such as:

- The ENUM server returns the domain of a switch for which a direct IP route does not exist.
- There are no business arrangements for routing the call on-net between two VOIP service providers.

Finally, the BTS10200 can block the call based on the received domain. By specifying **ROUTE-TYPE=NO-ROUTE** in the domain2route table, this feature can be used in cases where information received from the ENUM server points to the domain of the BTS that launched the ENUM query.

Call Routing If the BTS Does Not Receive a URI or Receives a No Response message

If the called DN is not an on-net subscriber (is not in the VoIP domain), the ENUM server does not return a URI. The BTS 10200 performs the existing routing operation and chooses the route specified against the destination (for example, the SS7 route or the SIP route toward the MGC) to send the call towards the terminating switch.

The same behavior applies in the following situations when the BTS 10200 might not have a URI available to route the call on-net:

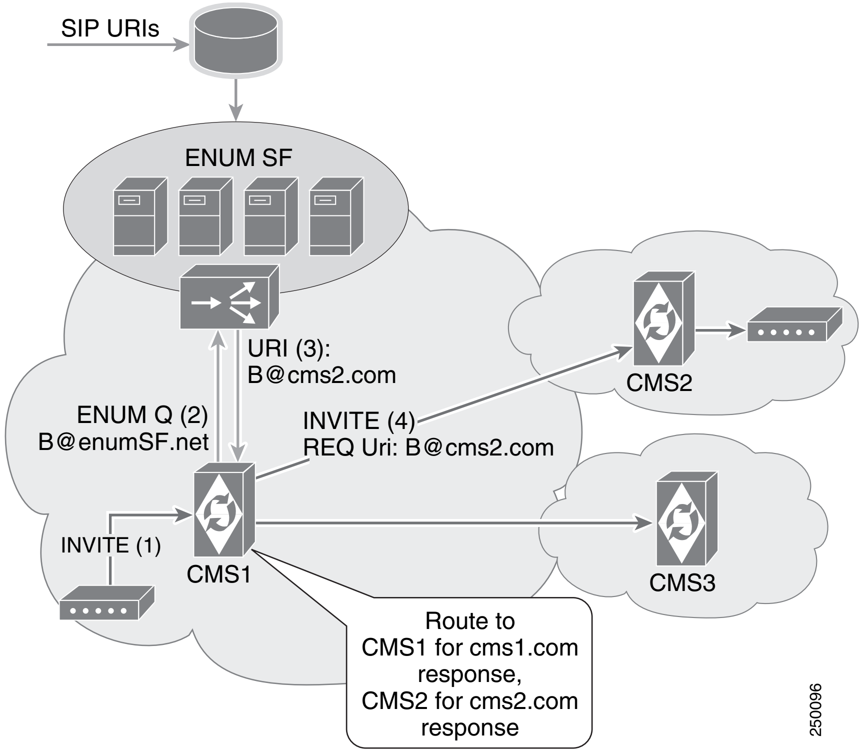
- There is no response from the ENUM server within the configured time-limit.
- No match is found for the route specified for the domain returned from the ENUM server.
- The internal BTS10200 resources are not available to perform the ENUM query.

ENUM Routing Use Cases

Selecting Inter-CMS Trunks

Figure 7 shows a typical BTS 10200 configuration for deployment to a network without a SIP route proxy. The service provider can create an inter-CMS trunk group and specify the routing policy for a particular route based on the domain portion of the return URI.

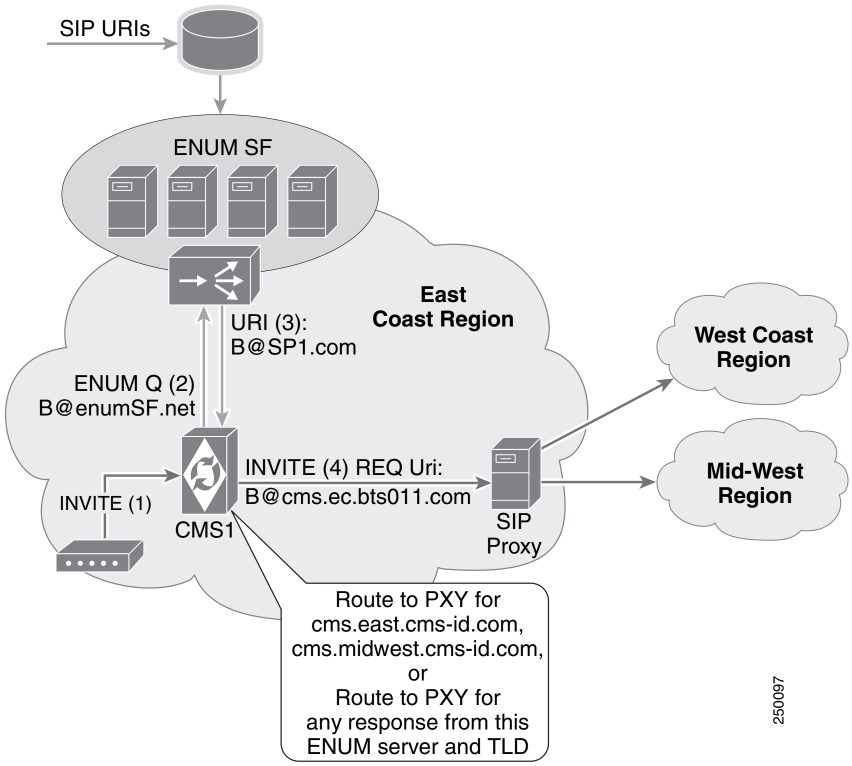
Figure 7 Inter-CMS Trunks: Network Without SIP Route Proxy



Selecting a SIP Trunk to a Route Proxy

Figure 8 shows a typical BTS 10200 configuration for deployment to a network with a SIP route proxy. The service provider can create one trunk group to go toward the SIP proxy and specify the routing logic for choosing the route, regardless of the domain of the returned URI. The domain portion of the returned URI is used in the Request URI for the outgoing SIP INVITE. The SIP proxy can direct the calls as appropriate based on this value.

Figure 8 Inter-CMS Trunks: Network With SIP Route Proxy

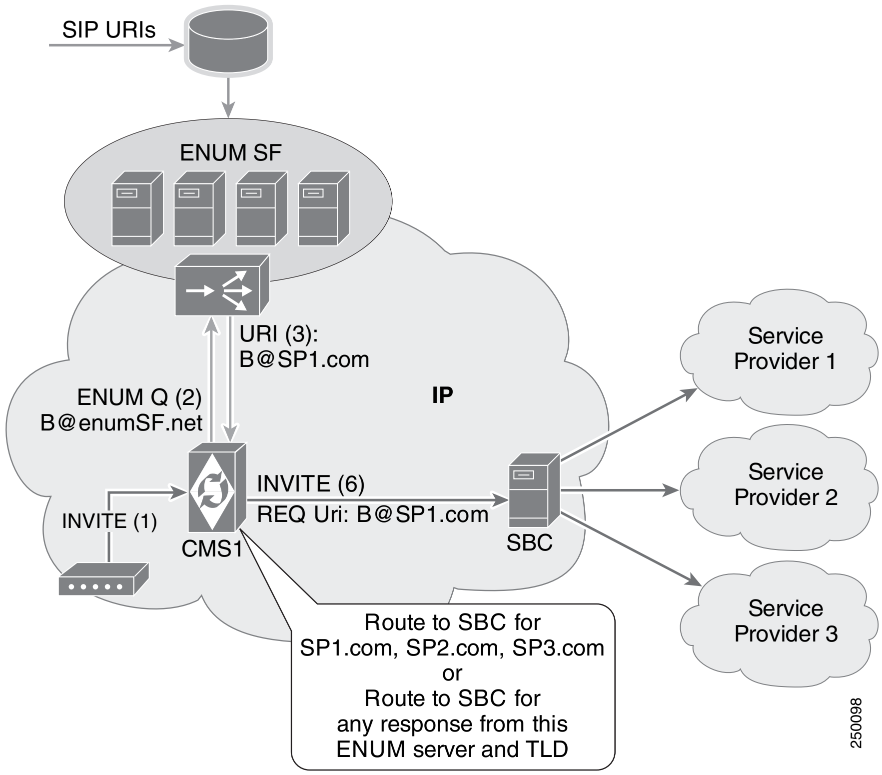


Selecting a SIP Trunk to an SBC

Figure 9 shows a typical configuration for routing calls for on-net subscribers in a different service provider domain when a peering arrangement to route on-net calls is established. A service provider typically uses a session border controller to route calls to or receive calls from a different service provider.

The service provider can direct one trunk group toward its own Session Border Controller (SBC) and specify the routing logic so it chooses a particular route, regardless of the domain in the returned URI. The domain portion of the returned URI is used in the Request URI for the outgoing SIP INVITE. The SBC can direct the calls as appropriate.

Figure 9 Routing a SIP Trunk to an SBC



ENUM Support for LNP Data

The Call Management Server (CMS) typically retrieves LNP information by sending a TCAP query toward a Signaling Control Point (SCP) database. RFC 4769 specifies another approach in which similar information can be specified in the ENUM server (E2U+PSTN NAPTR records) and retrieved through an ENUM query rather than a TCAP query. The BTS 10200 allows the user to disable the TCAP query mechanism and retrieve LNP information using the ENUM query mechanism.

The BTS 10200 retrieves the E2U+PSTN records from the ENUM server when LNP criteria match and LNP information is required.

Note The BTS 10200 will not use the domain portion specified against the rn field in the E2U+PSTN record for on-net routing described above.The BTS 10200 retrieves only the npdi flag (if available) and the user part of the rn field in the E2U+PSTN record.

BTS 10200 Routing Operations

The BTS 10200 uses the Local Routing Number (LRN) as well as the URI to make routing decisions. The following subsections describe the BTS 10200 routing. That behavior depends on information received by the BTS 10200 from the ENUM server.

BTS 10200 Receives the NPDI Indicator and LRN Information

The following table specifies the conditions that occur when the BTS 10200 receives the Number Portability Dip Indicator (NPDI) and LRN information:

If...	...Then
The URI is not available to the BTS 10200,	The BTS 10200 performs the routing operation based on the LRN. In addition, the BTS 102000 includes the NPDI indicator and LRN in the outgoing IAM/INVITE message.
The received LRN is for the same switch,	The BTS 10 2000 attempts to terminate the call to the subscriber.
The received LRN is a cluster LRN,	The BTS 10200 attempts to use the cluster dial plan routing to route the call.

BTS 10200 Receives Only Npdi Indicator

The BTS 10200 performs the routing operation based on the URI (if available) or the destination table. In addition, the BT S10200 includes the NPDI indicator in the outgoing IAM/INVITE message to indicate that the LNP query is already performed.

BTS 10200 Receives the NPDI Indicator, LRN and URI

The BTS 10200 always attempts to select SIP routes against the domain portion of the returned URI. However, if no routing policy is specified for the returned domain or the call cannot be routed on-net due to network conditions, the BTS 10200 tries to route the call based on the LRN information it received.

The user can also configure the BTS 10200 to use the ENUM functionality to retrieve LNP data but not perform domain-based routing. To do that, the user does not specify a domain-based routing policy.

Operational Recommendations for Porting Procedures

The following sections specify how customers should configure their server when a DN is ported into a VoIP network or ported out from a VoIP network.

Porting In a DN

A DN from a different network, such as a PSTN network, is ported in to the service provider network. During the transition phase

- No update is required in the LNP database.
- No update is required in the on-net routing database.

After porting is complete, the customer should do the following:

- Specify the E2U+SIP records for the ported-in number so all the switches within the VoIP network can use the on-net routing functionality.
- Update the LNP database with the LRN or cluster LRN of the destination switch to which the subscriber is ported. This information is typically used by nodes outside the VoIP domain to route the call.

Porting Out a DN

A DN from the service provider network is ported out on a switch in a different network, such as a PSTN network. During the transition phase

- No update is required in the LNP database.
- No update is required in the on-net routing database.

After porting is complete, the customer should do the following:

- Update the LNP database with the LRN or cluster LRN from the destination switch to which the subscriber is ported out.
- Remove the E2U+SIP records in the ENUM server so all VoIP nodes can use the LRN-based routing functionality.

Porting a DN to a Different Node

If a DN is moved from Node A to Node B within the same service provider network, modifying the E2U+SIP record in the ENUM server is sufficient as long as both nodes share the same LRN or cluster LRN. If the nodes have different LRNs or cluster LRNs, we recommend updating the LNP database so calls that originate in the PSTN network are efficiently routed.

ENUM Clients Operation

This section describes the processing rules for ENUM.

Launching the ENUM query

The BTS 10200 launches the ENUM query using the E.164 number of another server. The user can specify the top-level domain and predefined digits used for the query before the BTS 10200 launches the query. Use the following steps to launch the ENUM query.

Step 1 Delete from the digit string as many leading digits as specified by DEL-DIGITS in the ENUM profile, ignoring any leading or intermediate nondigit characters. For example:

digit string = 954048

DEL-DIGTS = 1

Result = 54048

Step 2 Prefix the digits specified by PFX-DIGITS in the ENUM profile to the transformed digit string after any leading nondigits or characters.

digit string = 54048

PFX-DIGITS = 1-469-25

Result = 1-469-255-4048

Step 3 Remove all nondigit characters from the transformed digit string.

digit string = 1-469-255-4048

Result = 14692554048

Step 4 Insert dots ('**.**') between the digits of the transformed digit string.

digit string = 14692554048

Result = 1.4.6.9.2.5.5.4.0.4.8

Step 5 Reverse the transformed digit string.

digit string = 1.4.6.9.2.5.5.4.0.4.8

Result = 8.4.0.4.5.5.2.9.6.4.1

Step 6 Append the ENUM domain root specified by the TOP_LEVEL_DOMAIN of the ENUM profile to the transformed digit string.

After processing is complete, the BTS 10200 sends the query to the ENUM server specified in the profile. The BTS 10200 ENUM client provides the nonblocking querying behavior to the applications

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Filtering the response

After receiving the response from the ENUM server, the BTS 10200 extracts the URI used for on-net routing, according to the specifications in RFC 3761. The following steps describe how the BTS 10200 automatically filters the response.

Step 1 All NAPTR records in which the service field does not match the requested service are ignored.

Step 2 All nonterminal NAPTR records (those records in which the flags field is not u) are ignored.

Step 3 The remaining NAPTR records are sorted based on the order field (in decreasing order) and the preference field (in increasing order).

Step 4 All NAPTR records in which the regexp field contains malformed regular expressions are ignored.

Step 5 The NAPTR record at the top of the list sorted in Xref_Colorparanum is selected.

Step 6 The regular expression rules specified in the regexp field are applied to the digit string to obtain a URI.

Step 7 The required information is extracted from the URI. Any malformed URIs are ignored.

Step 8 One of the following occurs:

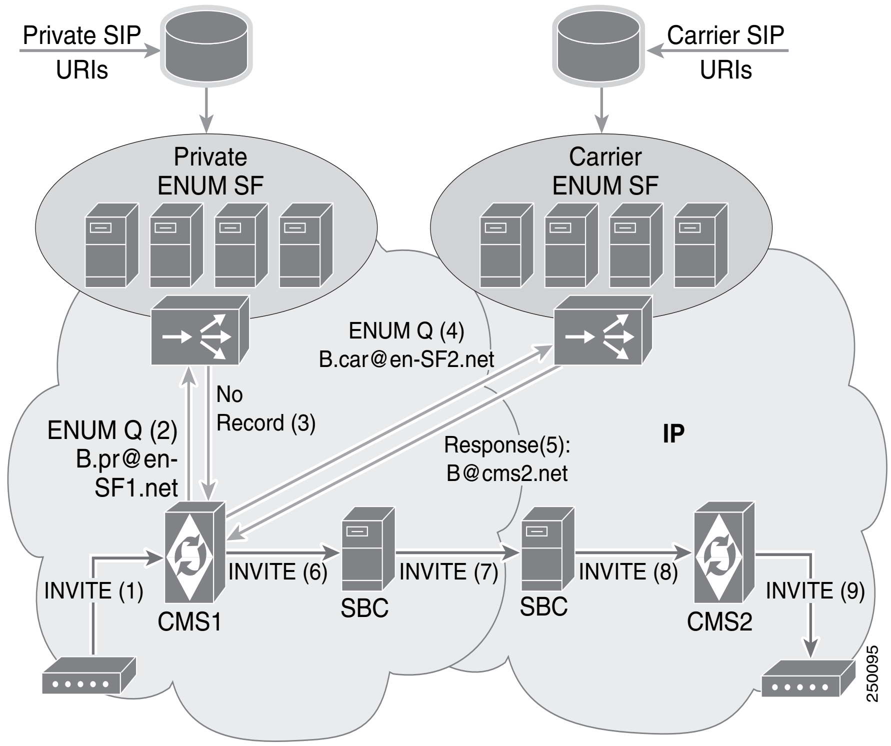
- If multiple URIs are present, the first in the list is used. Because the list has been sorted, the first in the list has priority.
- If only one URI is present, it is used for on-net routing.
- If no URIs are present, the DN is considered off-net and the call is routed using other routing rules.

Supporting Multiple Roots

The BTS 10200 can launch multiple ENUM queries toward the same set of ENUM servers or to a different set for the same call using different top-level domains. You can use this function when information for on-net subscribers is located in multiple ENUM servers, as when there is a carrier ENUM in addition to a private ENUM.

Figure 10 shows how the BTS 10200 supports multiple roots. In this example, it is configured to send an ENUM query to a private ENUM server farm. The private ENUM server farm responds with “no record found.” With the appropriate configuration, the BTS 10200 launches the query to the carrier ENUM server farm with a different top-level domain configured against the ENUM profile of carrier ENUM servers.

Figure 10 ***ENUM Queries for Multiple Roots***



Supporting Multiple ENUM Servers

The BTS 10200 can interface with multiple ENUM servers or server farms from which it retrieves data from the SRV records. It can also distribute the ENUM queries with a round-robin or priority order policy. Refer to "Configuring the DNS" section in the *BTS 10200 Softswitch Provisioning Guide* for additional details about the selection policy. The following procedure describes how the BTS 10200 retrieves data from the SRV records.

Step 1 Send an SRV query to the default DNS server for the logical service name obtained from the ENUM_SERVER_DOMAIN field specified in the ENUM profile table.

Step 2 Obtain and compile a list of servers from the target field of each SRV record returned by the query.

Step 3 Send an A query to the default DNS server for each server obtained in the previous steps.

Step 4 Obtain and compile a list of ENUM server IP addresses associated with each server in the A records returned by the query.

The cached IP addresses are reached at predefined intervals specified in the **ENUM_SERVER_DOMAIN_TTL** field in the ENUM profile. The query retrieves all changes in network configuration or query distribution policy.

Monitoring ENUM servers and Measuring Latency

The BTS10200 uses the actual ENUM queries and test queries to monitor the status of each ENUM server. The BTS10200 sends a test query to an ENUM server if no actual ENUM queries are sent to that server during the period specified by the TEST-QUERY-INTERVAL parameter in the ENUM profile table. The test query used to monitor the status of an ENUM server is an NAPTR query.

The BTS10200 uses the following logic to update the ENUM server availability status.

- If three consecutive queries (including test queries) time-out or indicate the server is unavailable, the BTS10200 marks that ENUM server as unavailable.
- If it receives three consecutive test query responses from an ENUM server that has been marked as unavailable, the BTS10200 marks that ENUM server as available.

The BTS10200 calculates the round-trip delay for each ENUM and test query to determine the average latency associated with the ENUM server.

The BTS10200 uses the latency and availability parameters to select the ENUM server to which the next ENUM query should be sent.

Planning

This section provides information on prerequisites applicable to this feature.

Prerequisites

The service provider must deploy a private or carrier ENUM server to make use of ENUM functionality on the BTS 10200. In addition, if the ENUM LNP feature is used, the same ENUM server must be configured with E2U+PSTN records for all ported DNs.

Operating

For information on the ENUM measurements and the effect of ENUM feature on the Billing fields, refer to the *Cisco BTS 10200 Softswitch Billing Interface Guide* and the *BTS 10200 Softswitch Operations and Maintenance Guide*.

Troubleshooting

For information on ENUM signaling events and alarms, refer to *BTS 10200 Softswitch Troubleshooting 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

For general troubleshooting procedures, see the *Cisco BTS 10200 Softswitch Troubleshooting Guide*.

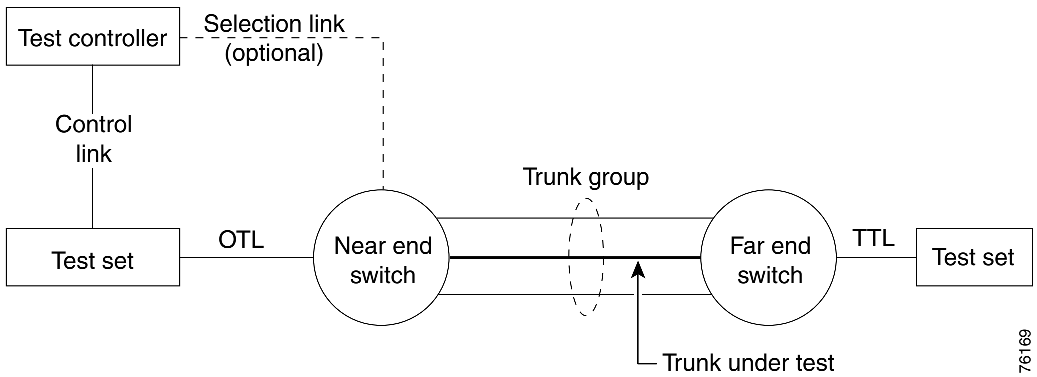
Trunk Testing

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 1-11.

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

Figure 1-11 Trunk Testing Setup



Near End Test Origination Test Calls

The BTS 10200 supports calls used to test individual trunks that connect a local gateway with a gateway or PSTN switch at a remote office. The BTS 10200 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 BTS 10200. The processes may work differently on other switches.

The process for testing a BTS 10200 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 BTS 10200.
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 BTS 10200 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.

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

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

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—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. BTS 10200 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:

1. 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 1-11.
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.

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, BTS 10200 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 BTS 10200 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 BTS 10200.

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 MTA. The loopback tests can be initiated from designated test endpoints (subscribers) controlled by the BTS 10200. 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 BTS 10200, 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

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*.