



CHAPTER 2

Supported Signaling Protocols

Revised: February 7, 2008, OL-11755-03

The Cisco BTS 10200 Softswitch supports the following types of external signaling protocols:

- Media Gateway Control Protocol (MGCP) line
- MGCP Channel-Associated Signaling (CAS)
- Integrated Services Digital Network (ISDN) primary rate interface (PRI)
- Signaling Transport (SIGTRAN) for SS7 applications, including ISDN user part (ISUP) support for several national ISUP variants
- H.323
- Session Initiation Protocol (SIP) and SIP-T
- PacketCable-based signaling protocols:
 - Network-Based Call Signaling (NCS) protocol
 - Trunking Gateway Control Protocol (TGCP)
 - DQoS/COPS query and response protocol
 - Remote authentication dial-in user service (RADIUS) authentication protocol (IETF RFC 2865)

The Cisco BTS 10200 Softswitch 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 signaling features involve the use of other NEs deployed in the service provider network, for example, gateways, media servers, announcement servers, eMTAs, H.323 endpoints, and SIP phones. See the “[Component Interoperability](#)” section of the *Release Notes* for a complete list of the specific peripheral platforms, functions, and software loads that have been used in system testing for interoperability with the Cisco BTS 10200 Softswitch Release 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 Cisco BTS 10200 Softswitch.

The signaling types are described in more detail in the sections that follow:

- [MGCP Line Signaling Support, page 2-2](#)
- [MGCP CAS Signaling Support, page 2-3](#)

- [SS7 Signaling Support Through SIGTRAN, page 2-4](#)
- [ISDN Signaling Support, page 2-10](#)
- [H.323 Signaling Support, page 2-11](#)
- [SIP and SIP-T Signaling Support, page 2-13](#)
- [PacketCable-Based Signaling Support, page 2-15](#)

MGCP Line Signaling Support

Media gateways (MGWs) provide bearer paths between voice and packet networks. MGWs also provide connection control, endpoint control, auditing, and status functions. These gateways are equipped with voice coders that convert voice into packets, and voice decoders that convert packets into voice. Connections are grouped in calls, which means that a call can have one or more connections. One or more Call Agents (CAs) set up the connections and calls.

The Cisco BTS 10200 Softswitch connects to a variety of MGWs using Media Gateway Control Protocol (MGCP), and provides voice over IP (VoIP) bearer-path control. This implementation is based upon the evolving industry standards for MGCP, including the following MGCP variants:

- MGCP (IETF Version 0.1, Draft 5, February 1999)
- MGCP (IETF RFC 2705, Version 1.0, October 1999)



Note

The MGCP-VERSION and MGCP-VARIANT parameters in the MGW-PROFILE table are used to identify the MGCP version and variant that an MGW supports.

General Functions of the MGCP Interface

The MGCP interface performs the following functions:

- Handles MGW initialization
- Provides endpoint auditing
- Provides MGW fault management
- Provides maintenance and administration of each termination, MGW operational states, and so forth
- Carries call-control signaling
- Carries media-path control signaling

Special Functions of MGCP Interface

The Cisco BTS 10200 Softswitch supports several special-purpose MGCP-based functions:

- Codec selection service—The process a CA uses to find a common codec (coder/decoder) type between an originating and terminating call leg so a call can go through. The preferred codec type for originating and terminating calls is provisioned by the service provider using the QoS table in the Cisco BTS 10200 Softswitch database. The QoS can be configured for a subscriber or trunk group (TG). The CA makes a decision on actual codec type based on a combination of the following conditions:

- Codec types available on the MGW—The MGW dynamic profile (list of supported codecs reported by MGW) or MGW static codec list (list of supported codecs configured in the Cisco BTS 10200 Softswitch).
- The codec type provisioned in the QoS table—If a certain codec type is provisioned in the QoS table but not available in the MGW dynamic profile or TG profile, that type cannot be used. When no matching code is found, default pulse code modulation mu-law (PCMU) codec is used.

Several codec types are supported, including:

- G.711 mu-law (PCMU)—Default value for codec type
 - G.711 A-law (PCMA)
 - G.723.1 High rate
 - G.723.1 Annex A High rate
 - G.723.1 Low rate
 - G.723.1 Annex A Low rate
 - G.729
 - Additional codecs—See the QoS table in the [Cisco BTS 10200 Softswitch CLI Database](#) for a complete list.
- Resource Reservation Protocol (RSVP)—An Internet Engineering Task Force (IETF) protocol for providing integrated services and reserving resources on the IP network. The service provider provisions the preferred reservation profile (guaranteed, controlled load, or best effort) in the QoS table. When a reservation is needed on a connection, the Cisco BTS 10200 Softswitch specifies the preferred reservation profile to the gateway. Whether or not RSVP is used depends on the configuration of the gateway as well as the preferred reservation profile specified by the Cisco BTS 10200 Softswitch. If the best-effort RSVP profile is specified, RSVP is not performed.
 - Announcement server—A media server that stores network-based announcements and plays them to a caller upon request from the Cisco BTS 10200 Softswitch. The announcement server interfaces with the Cisco BTS 10200 Softswitch using MGCP. Every Cisco BTS 10200 Softswitch in the network requires its own announcement server.
 - Dual tone multifrequency (DTMF) signaling—Signaling that is transported across the IP network under MGCP control.
 - Channel-Associated Signaling (CAS)—Signaling that is used with the MGCP interworking function.
 - Voice over ATM (VoATM) support—Configurable parameters that support ATM extensions (AAL1, AAL2, and AAL5) on MGCP.



Note

The ATM adaptation layer (AAL) is a standards-based layer that allows multiple applications to have data converted to and from an ATM cell. It uses a protocol that translates data for higher-layer services into the size and format of an ATM cell.

MGCP CAS Signaling Support

The Cisco BTS 10200 Softswitch supports the following MGCP CAS interfaces:

- Public safety answering point (PSAP) systems interface for 911 emergency services
- Operator services interface, including a legacy operator services interface that uses MF/T1 trunks

- PBX interfaces



Note CAS is used with the MGCP interworking function.

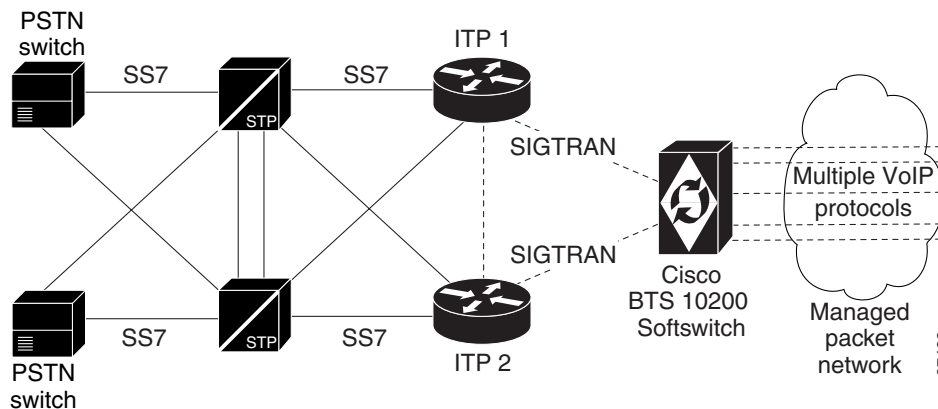
SS7 Signaling Support Through SIGTRAN

The Cisco BTS 10200 Softswitch communicates with Signaling System 7 (SS7)-based PSTN switches and service control points (SCPs) by using a SIGTRAN-based signaling gateway (SG). The SIGTRAN interface carries all SS7 messages encapsulated in IP packets. The Cisco IP Transfer Point (ITP) is one of the SGs used with the Cisco BTS 10200 Softswitch for this purpose.

Interface to the SS7 Network

The basic interface of the Cisco BTS 10200 Softswitch to the SS7 network is shown in [Figure 2-1](#).

Figure 2-1 Cisco BTS 10200 Softswitch Interface to the SS7 Network



Note For information on compatibility with specific Cisco ITPs, see the “Cisco ITP Signaling Gateways” section in the [Cisco BTS 10200 Softswitch Release Notes](#).

The Cisco BTS 10200 Softswitch can be configured to have multiple originating point codes (OPCs). For information on OPCs, network configuration options, and subsystems, see the [Cisco BTS 10200 Softswitch SS7 SIGTRAN Solution Guide](#).

For additional information, see the following standards and industry documents:

- ANSI T1.113, *Telecommunications Signaling System No. 7 (SS7)–Integrated Services Digital Network (ISDN) User Part (ISUP)*
- GR-317-CORE, *Switching System Requirements for Call Control Using the Integrated Services Digital Network User Part*
- GR-394-CORE, *Switching System Generic Requirements for Interexchange Carrier Interconnection Using the Integrated Services Digital Network User Part*
- GR-533-CORE, *LSSGR: Database Services Service Switching Points–Toll-Free Service*

- GR-1188-CORE, *LSSGR: CLASS Feature: Calling Name Delivery Generic Requirements*
- IETF RFC 2960, *Stream Control Transport Protocol (SCTP)*
- IETF draft-ietf-sigtran-sua-14.txt, *Signalling Connection Control Part User Adaptation Layer (SUA)*

Support for ISUP Variants

The Cisco BTS 10200 Softswitch supports the following ISUP variants:

- ANSI ISUP (for NANP region, based on Telcordia document GR-317)
- ITU93 White Book ISUP
- European Telecommunications Standards Institute (ETSI) v2 ISUP
- Q.761 Standard
- Q.761 China
- Q.761 Thailand
- Q.761 ETSI v3 Hungary
- Q.761 Standard 97
- Q.761 ETSI v3 France
- Q.761 ETSI v3 Poland
- Q.767 Standard
- Q.767 Mexico
- Q.767 Colombia

ISUP Transparency with the Cisco PGW 2200

ISUP transparency provides the capability for the Cisco BTS 10200 Softswitch to transfer Generic Transparency Descriptor (GTD) messages and information elements across an IP network to a Cisco PGW 2200. In the Cisco PGW 2200, the GTD messages are mapped to ISUP messages, repackaged, and sent out to the PSTN/SS7 network. ISUP transparency is important because it enables the transport of calls from a Session Initiation Protocol (SIP) network through an IP network and out to a PSTN network without any loss of signaling information. ISUP transparency is achieved with the use of the Cisco GTD mechanism. GTD provides a means to specify messages of various protocols used in the PSTN network in plain text format. In that format, they can be easily understood by the network elements (NEs) within the IP network or on the boundary between the PSTN and IP networks.



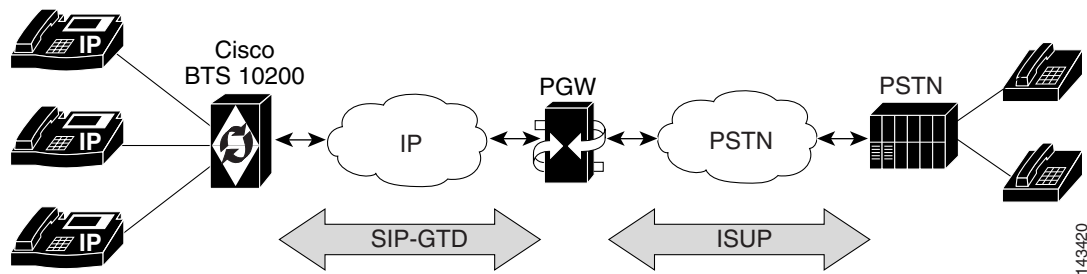
Note

This feature is supported in Cisco PGW Software Release 9.6(1) and Cisco BTS 10200 Softswitch Software Release 5.0.

The ISUP transparency function on the BTS-PGW interface, illustrated in [Figure 2-2](#), passes normalized parameters to expedite (1) mapping at the PSTN interconnect side and (2) any feature invocation necessary on either the Cisco PGW 2200 or the Cisco BTS 10200 Softswitch. It adds support for GTD attachments to SIP-T trunk messages, allowing the Cisco BTS 10200 Softswitch to interwork with the Cisco PGW 2200 for interconnection to the PSTN.

Figure 2-2 ISUP Transparency on the BTS-PGW Interface

SIP/MGCP phones



When the Cisco BTS 10200 Softswitch generates SIP messages to be sent out on SIP-T trunks, a GTD attachment is generated based on the GTD parameters defined in the GTD-PARMS token in the Softswitch Trunk Group Profile (softsw-tg-profile) table. The Cisco PGW 2200 decodes GTD attachments of incoming SIP messages, and converts all GTD parameter contents to the equivalent ISUP values in the appropriate information element on the outgoing PSTN side.

When the egress trunk is a SIP-T trunk, the system supports the mapping of Progress Indication messages from the Cisco BTS 10200 Softswitch to SIP INFO messages with GTD attachments containing Call Progress (CPG) messages. This supported feature applies only to SIP subscribers. When a SIP INFO or RE-INVITE message is received over a SIP-T trunk with a GTD attachment containing a CPG message, a Progress Indication message is generated and sent to the system.

In the deployment model, the Cisco PGW 2200 is the PSTN gateway, and the Cisco BTS 10200 Softswitch provides a residential or Centrex application platform.

PSTN Supplementary Services

The following PSTN supplementary services are enabled by the ISUP transparency feature:

- Number ID supplementary services
 - Direct Dial In (DDI)
 - Calling Line Identification Presentation (CLIP)
 - Calling Line Identification Restriction (CLIR)
- Call diversion supplementary services
 - Call Forwarding Busy (CFB)
 - Call Forwarding No Reply (CFNR)
 - Call Forwarding Unconditional (CFU)
 - Call Waiting (CW)
 - Call Hold (HOLD)
- Multiparty supplementary services
 - Three-Party Service (3PTY)
- Transparency requirements
 - Ability to provision which parameters to transport over GTD
 - Call Forwarding No Answer (CFNA)
 - Call Waiting (CW)
 - Call Transfer (SIP Refer is not supported with SIP subscriber Hold signaling)

- Ability to correlate billing records
- Functionality provided by the Cisco PGW 2200
 - Number Portability (NP)
 - NoA relay
 - Information/Information Request (INF/INR) and Identification Request/Identification Response (IDR/IDS) messaging
 - ITU Method 2 circuit selection
 - NoA modification and routing
 - Calling Party Category (CPC) based routing
 - Ability to modify A-number based on B-number and B-number based on A-number
 - Cause analysis
 - Redirecting A-number screening
 - Virtual VPN behavior
 - Calling Party Number (CGPN) Address Presentation Indicators

Call Progress Signaling for SIP Subscriber on Call Hold

The Cisco BTS 10200 Softswitch can be provisioned to send a call-hold event signal to the other party in the call when a SIP subscriber goes on or off hold. This provisioning is done by means of the SIP-SUB-SEND-CPG-ON-HOLD-SIGNAL token in the CA-CONFIG table. The default value of this token is N. Therefore, you must change this value to Y if you want this signal to be sent for all SIP subscribers.



Note

The message sent to mute the media path is always sent to the other party, regardless of this flag setting.

Limitations

This feature is subject to the following limitations:

- SIP Refer is not supported with SIP Subscriber Send CPG on Hold Signaling.
- Overdecadic digits are not supported.
- The Cisco PGW 2200 does not send INR messages to the Cisco BTS 10200 Softswitch. It responds to INR requests with an INF indication that there is no new information.

GTD Parameters Supported

Table 2-1 shows the GTD parameters supported by this feature and indicates the GTD messages in which each parameter is supported. The values in the GTD Parameter and Name columns of this table are placed in the static Generic Transparency Descriptor Parameter Values (gtd-param-values) table. You select values from the GTD Parameter column to provision the GTD-PARMS token in the Softswitch Trunk Group Profile (softsw-tg-profile) table.

Enabling a parameter causes it to be encoded in an outgoing GTD attachment of a SIP message on the trunk group if the information is available in the call context.

Only the GTD parameter listed for each GTD message type is decoded when a SIP message with a GTD attachment is received by the system from the network.

For example, the GTD ACL parameter in a GTD REL message will be decoded if it is received, whether it is provisioned or not. However, a GTD UUS parameter received in a GTD REL message is ignored, even if it is provisioned, because it is not in the table.

Table 2-1 GTD Parameters and Supported GTD Messages

GTD Parameter	Name	GTD IAM	GTD ACM	GTD CPG	GTD ANM	GTD CON	GTD REL	GTD SUS	GTD RES
ACL	Automatic Congestion Level						Yes		
ATP	Access Transport	Yes	Yes	Yes	Yes	Yes	Yes		
BCI	Backward Call Indicators		Yes	Yes	Yes	Yes			
CAI	Cause Indicators		Yes	Yes			Yes		
CDI	Call Diversion Information		Yes	Yes					
CGN	Calling Party Number	Yes							
CHN	Charge Number	Yes							
CID	Carrier Identification	Yes							
CNN	Connected Number				Yes	Yes			
CPC	Calling Party Category	Yes							
CPN	Called Party Number	Yes							
CSI	Carrier Selection Information	Yes							
DIS	Display Information				Yes	Yes	Yes		
EVI	Event Information Indicators		Yes	Yes			Yes		
FCI	Forward Call Indicators	Yes							
GCI	Global Call Identification	Yes				Yes			
GEA	Generic Address	Yes							
GED	Generic Digits	Yes							
GEN	Generic Name	Yes				Yes			
GNO	Generic Notification	Yes	Yes	Yes					
HOC	Hop Counter	Yes							
JUR	Jurisdiction	Yes							
NOC	Nature of Connection Indicators	Yes							
NSF	Network Specific Facilities		Yes	Yes					
OBI	Optional Backward Call Indicators		Yes	Yes	Yes	Yes			
OCN	Original Called Number	Yes							
OLI	Originating Line Information	Yes							
RCT	Redirect Counter	Yes							
RGN	Redirecting Number	Yes							
RNI	Redirection Information	Yes							

Table 2-1 GTD Parameters and Supported GTD Messages (continued)

GTD Parameter	Name	GTD IAM	GTD ACM	GTD CPG	GTD ANM	GTD CON	GTD REL	GTD SUS	GTD RES
RNN	Redirection Number		Yes	Yes	Yes	Yes	Yes		
RNR	Redirection Number Restriction		Yes	Yes	Yes	Yes			
SCI	Service Code Indicator	Yes							
SRI	Suspend/Resume Indicators							Yes	Yes
TMR	Transmission Medium Required	Yes				Yes			
TNS	Transit Network Selection	Yes							
UID	UID Indicators		Yes	Yes					
UUI	User-To-User Indicators		Yes	Yes	Yes	Yes			
UUS	User-To-User Information	Yes	Yes	Yes	Yes	Yes			

**Note**

Columns in [Table 2-1](#) show the GTD message in which each GTD parameter is supported. This information is for reference only and is not provisionable.

**Note**

UID is only decoded. It is never encoded.

Billing Fields

The BTS 10200 and PGW 2200 produce their own independent billing records. Downstream billing mediation servers use the SIP Call ID to correlate the two records, if required. The SIP Call ID is available in the PGW 2200 CDR record, tag 4203, and in the BTS 10200 CDR record, fields 116 and 144.

Cause Code Selection Precedence

The system performs cause code selection according to the following order of precedence:

1. SIP header reason code
2. GTD body
3. SIP response code

Troubleshooting

There are no troubleshooting tools created specifically for the transparency feature. Use the existing tools to extract traces from log files on the BTS 10200 and the call trace and siptool capabilities on the PGW 2200. Both tools support ASCII attachments such as the GTD attachment.

Provisioning Procedure

See the “[ISUP Transparency on the BTS-PGW ISUP Transparency](#)” section in the *Cisco BTS 10200 Softswitch Provisioning Guide*.

Additional SIGTRAN and SS7 Information

For additional information on provisioning and using SIGTRAN and SS7 protocols on the Cisco BTS 10200 Softswitch, see the [SIGTRAN Solution Guide](#).

ISDN Signaling Support

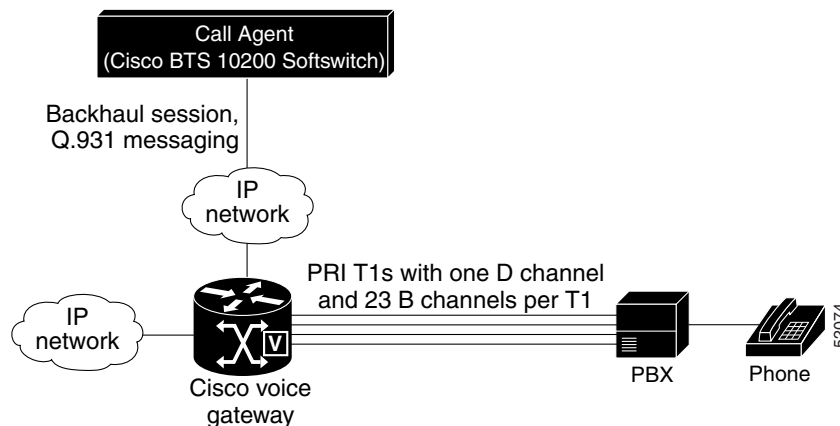
This section describes the Integrated Services Digital Network (ISDN) Primary Rate Interface (PRI) variants and supplementary services supported by the Cisco BTS 10200 Softswitch. ISDN PRI allows the Cisco BTS 10200 Softswitch to interconnect to small and medium businesses using legacy PBX PRI interfaces. The basic ISDN NEs and signaling connections are shown in [Figure 2-3](#).



Note

Standby elements in the figure are omitted for clarity.

Figure 2-3 Example of ISDN NEs



The design provides for transport of PRI information elements (IEs) and messages. Interoperability is supported with the following PRI variants:

- Nortel DMS-100
- AT&T 4ESS
- Lucent 5ESS
- NI2

The Cisco BTS 10200 Softswitch supports the following capabilities:

- ISDN T1 PRI
- Q.921 and Q.931 network side
- ISDN backhaul communication of Q.931 messages from MGWs to the Cisco BTS 10200 Softswitch

- Facility Associated Signaling (FAS)
- Non-Facility Associated Signaling (NFAS)
- Backup D channel

**Note**

For additional details and procedures for the Cisco BTS 10200 Softswitch ISDN implementation, see the *Cisco BTS 10200 Softswitch ISDN Provisioning and Troubleshooting Guide*.

H.323 Signaling Support

The Cisco BTS 10200 Softswitch functions as a logical H.323 gateway to communicate with H.323 gatekeepers (GKs), and with Cisco CallManager and other H.323 gateways. The Cisco BTS 10200 Softswitch also provides signaling for other trunks and lines over MGCP and SIP protocols. In addition, it communicates with signaling gateways (SGs) for SS7 signaling and with trunking gateways (TGWs) that provide the bearer path to the PSTN. This allows H.323 Internet VoIP traffic to be carried seamlessly into the PSTN networks.

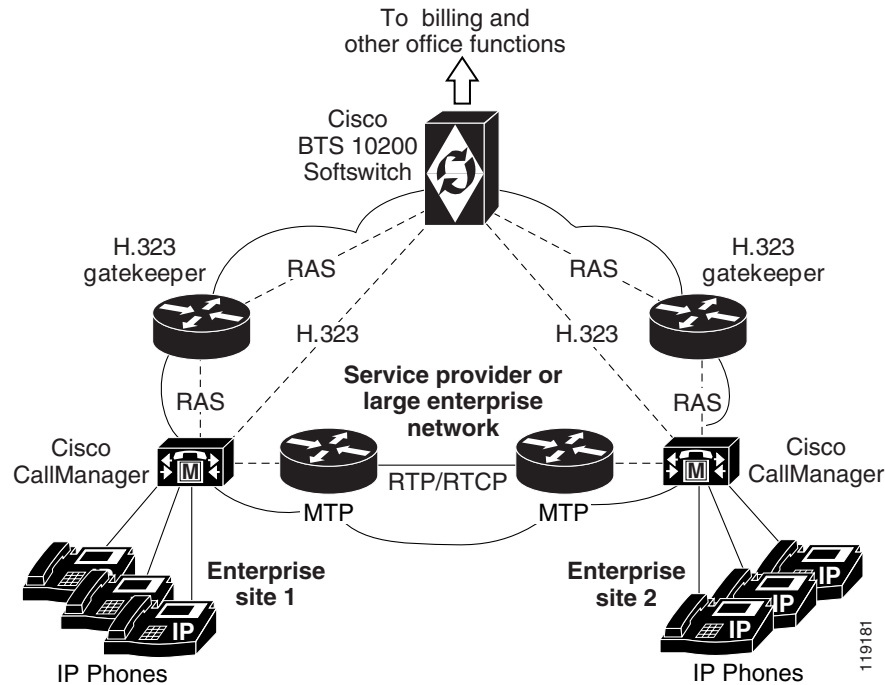
These signaling links are shown in [Figure 2-4](#).

**Note**

You can configure up to four logical H.323 gateways on the Cisco BTS 10200 Softswitch.

The diagram illustrates a VoIP network architecture. It shows a central **Managed packet network** cloud. To the left, a **PSTN network** cloud contains a **PSTN switch** and is connected to the IP network via an **SS7** signaling path and a **Bearer path**. The **IP network** cloud contains several components: an **H.323 gatekeeper** connected to an **ITP** (IP Telephony Proxy) and an **H.323/RAS** (Registration, Admission, and Status) protocol; a **SIGTRAN** interface connecting the **ITP** to a **TGW** (Transit Gateway); an **MGCP** (Media Gateway Control Protocol) interface connecting the **TGW** to an **MGW** (Media Gateway); an **MGW** connected to an **ATA** (Analog Telephone Adapter) which interfaces with **Analog** phones; an **H.323 GW** (Gateway) connected to **H.323** video phones; and a **Cisco BTS 10200 Softswitch** connected to **IP Phones** via **H.323 with RAS Intercluster trunk** and **H.323 without RAS Intercluster trunk** links. The **Managed packet network** also connects to **Cisco CallManager** and **Cisco CallManager** instances, which in turn connect to **IP Phones**. The network is labeled **To PSTN and enterprise networks** with an arrow pointing left.

The interoperability between the Cisco BTS 10200 Softswitch, Cisco CallManager, and Cisco IOS H.323 gateways enhances the delivery of call control features between enterprise networks and service provider networks. These systems interoperate to provide subscriber features such as call forwarding, call waiting, call transfer, and three-way calling. The Cisco BTS 10200 Softswitch can be used to connect calls between two phones that reside on different Cisco CallManager systems (see [Figure 2-5](#)). Signaling of certain information, for example connected name and number information, is transparently passed from the terminating Cisco CallManager to the Cisco BTS 10200 Softswitch and back to the originating Cisco CallManager.

Figure 2-5 Example of Connecting Calls from Phones on Separate Cisco CallManager Systems**Note**

For additional technical discussion, prerequisites, and provisioning steps, see the *Cisco BTS 10200 Softswitch H.323 Provisioning and Troubleshooting Guide*.

SIP and SIP-T Signaling Support

The Cisco BTS 10200 Softswitch uses Session Initiation Protocol (SIP) and SIP for telephones (SIP-T) signaling to communicate with other SIP-based NEs. This implementation is based upon the evolving industry standards for SIP, including IETF document RFC 3261, *SIP: Session Initiation Protocol*.

**Note**

This section provides an overview of SIP implementation on the Cisco BTS 10200 Softswitch. For SIP feature details and applicable procedures, see the *Cisco BTS 10200 Softswitch SIP Protocol Guide* and the *Cisco BTS 10200 Softswitch SIP Protocol Provisioning Guide*.

SIP Functions

The Cisco BTS 10200 Softswitch supports both SIP trunks and SIP-based subscriber lines (SIP phones). It provides the following SIP-related functions:

- Protocol conversion between SIP and several other protocols, including SS7, PRI, H.323, MGCP, and CAS
- Tandem back-to-back user agent (UA) for direct SIP-to-SIP calls (trunk to trunk, phone to phone, and trunk to/from phone), and SIP-to-SIP-T calls

**Note**

There is no provisioning associated with the back-to-back UA functionality. The Cisco BTS 10200 Softswitch automatically acts as a back-to-back UA when there is a SIP-to-SIP call.

- SS7 bridging between softswitches by means of SIP-T methods
- Native support of SIP endpoints such as SIP phones, including authentication and registration management. (For example, the Cisco BTS 10200 Softswitch maintains the current location of SIP subscribers.)
- Verification of the SIP REGISTER message to ensure that it came from a provisioned endpoint

SIP roles performed by the Cisco BTS 10200 Softswitch include:

- User agent server (UAS)
- User agent client (UAC)
- Registrar

Applicable SIP references are listed in the [“SIP and SIP-T References” section on page 2-15](#).

SIP Features

The Cisco BTS 10200 Softswitch supports the following SIP features:

- Reliable provisional response
- 3XX redirect response on SIP trunks
- SIP hairpin
- Third-party call control (3PCC)
- ANI-based routing for SIP calls
- DTMF relay for communications with interactive voice response (IVR) servers
 - SUBSCRIBE/NOTIFY method
 - INFO method
- Message waiting indicator
- Diversion header
- UAC and UAS forking
- SIP session timer
- Type of service (ToS) for SIP signaling
- DNS services (DNS SRV) lookup for initiating SIP calls
- DNS naming authority pointer (NAPTR) lookup for initiating SIP calls
- Mapping the carrier identification code (CIC) in the SIP uniform resource identifier (URI) to a transit network selection (TNS)
- SIP register
- SIP authentication
- SIP refer
- SIP trunk audit

- SIP-trunk route advance with provisionable timer for Invite retransmission

SIP-T Support

The Cisco BTS 10200 Softswitch supports SIP-T functions. SIP-T is used to bridge calls between two SS7 networks. SIP-T encapsulates the SS7 ISUP information elements (based on GR-317 ISUP version) and carries them through the packet network. It provides for encapsulation/decapsulation at the PSTN gateways and helps route the call through the packet network. SIP-T functions are described in IETF RFC 3398, *Integrated Services Digital Network (ISDN) User Part (ISUP) to Session Initiation Protocol (SIP) Mapping*.

FCP Interface

The Cisco BTS 10200 Softswitch uses Feature Control Protocol (FCP) for internal communications between the Call Agent (CA) and Feature Server (FS) components. FCP is a Multipurpose Internet Mail Extension (MIME) application on top of SIP. FCP uses SIP for transport, and carries call state control and status information needed for feature control.

SIP Billing Support

The Cisco BTS 10200 Softswitch provides call data for billing on SIP calls. Specific fields are supported in the call detail records for calls that originate or terminate on a SIP trunk or subscriber line. For detailed information on billing management and data, see the *Cisco BTS 10200 Softswitch Billing Interface Guide*.

SIP and SIP-T References

The BTS 10200 SIP implementation is based on the evolving standards in the Internet Engineering Task Force (IETF) Request for Comments (RFC) publications, and may not be fully compliant in all cases. The BTS 10200 is largely compliant with RFC 3261. For the level of compliance with other RFC publications and drafts, see the specific feature descriptions in the [Cisco BTS 10200 Softswitch SIP Feature and Provisioning Guide](#).

PacketCable-Based Signaling Support

This section summarizes Cisco BTS 10200 Softswitch support for PacketCable-based signaling and includes the following topics:

- [PacketCable-Based Functions](#), page 2-16
- [Event Message Implementation](#), page 2-17
- [Security Implementation](#), page 2-18

PacketCable-Based Functions

In a PacketCable-based network, the Cisco BTS 10200 Softswitch functions as both a call management server (CMS) and a media gateway controller (MGC).

The Cisco BTS 10200 Softswitch provides call control, call routing, and signaling for several types of NEs:

- Multimedia terminal adapters (MTAs) and embedded MTAs (eMTAs)
- Cable modem termination systems (CMTSs)
- Trunking gateways (TGWs)

The Cisco BTS 10200 Softswitch supports cable access for voice application, including communications with the Cisco UBR 7246 and Cisco UBR 924 universal broadband routers. It also provides interfaces to Record Keeping Servers (RKSs) for billing purposes, and IP security functions.

The Cisco BTS 10200 Softswitch provides support for the following PacketCable-based protocols and functions:

- Network-Based Call Signaling (NCS) protocol.
- Trunking Gateway Control Protocol (TGCP).



Note

NCS protocol and TGCP are based on MGCP; they are referred to as profiles of MGCP.

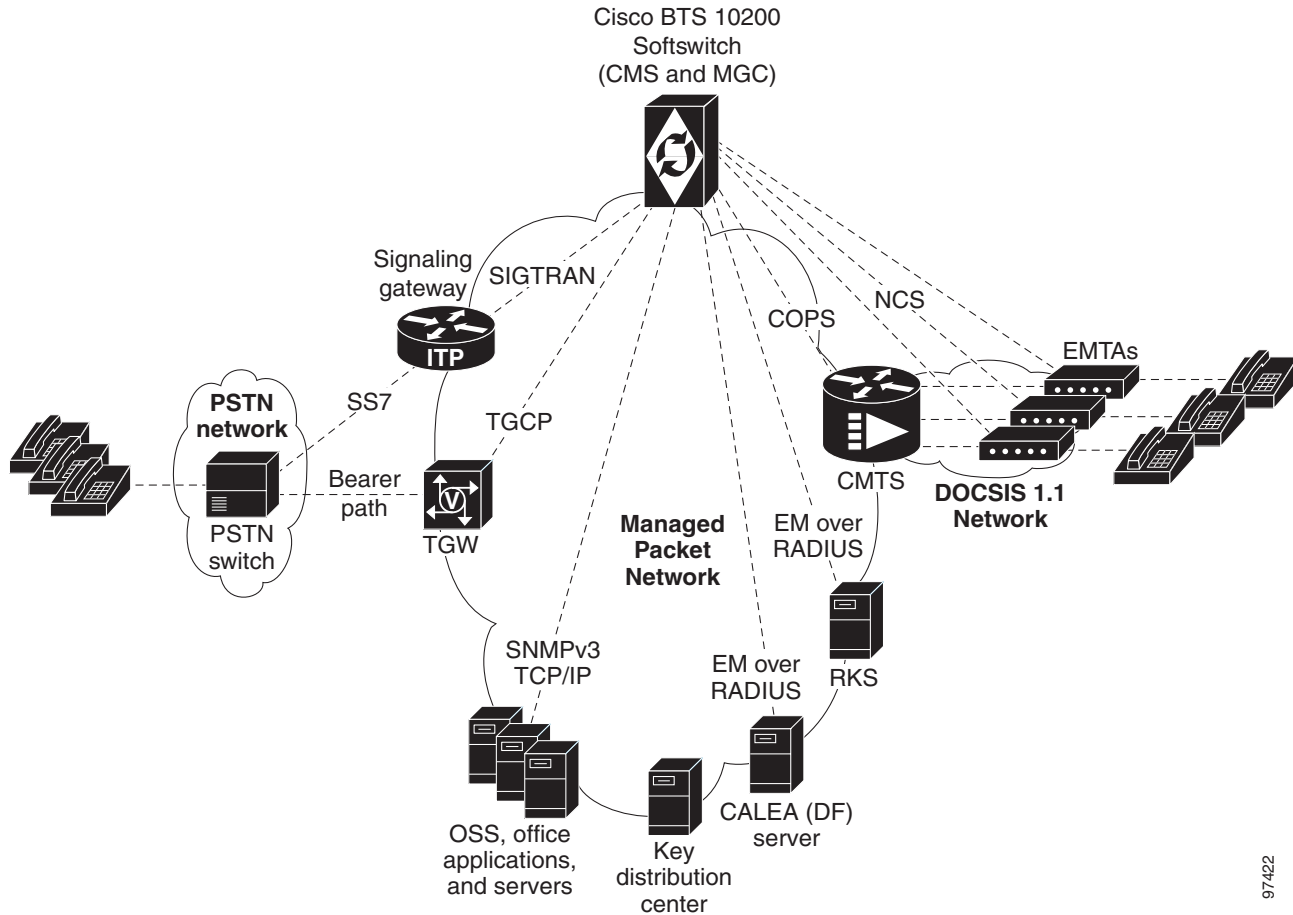
- Dynamic Quality of Service (DQoS)/Common Open Policy Service (COPS) query and response protocol.
- RADIUS authentication protocol (IETF RFC 2865), used for transmission of event messages (EMs) to an external RKS for billing purposes.
- Security features, including implementation of IP security (IPsec) architecture, key management using Internet Key Exchange (IKE) and Kerberos, and encryption of certain IPsec keys.
- Interface for support of lawful intercept and the Communications Assistance for Law Enforcement Act (CALEA). See the “Lawful Intercept Interface” section in the *Cisco BTS 10200 Softswitch Network and Subscriber Feature Descriptions* document for a description of this feature.



Note

For detailed information on compliance with specific paragraphs of the IETF standards (for TGCP, IP Security, NCS, and so forth), contact your Cisco account team.

Figure 2-6 shows a typical network with PacketCable-based NEs and the applicable external interfaces of the Cisco BTS 10200 Softswitch.

Figure 2-6 Example of PacketCable-Based Network Architecture

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Event Message Implementation

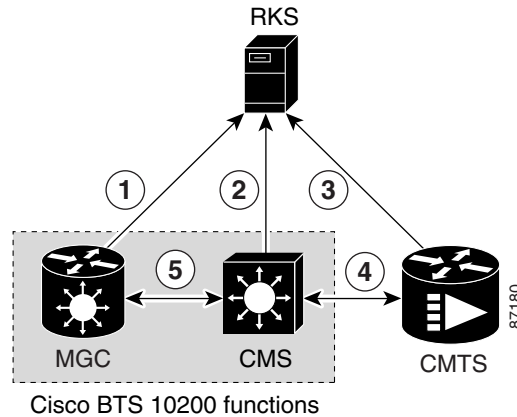
This section describes the implementation of the event message (EM) feature on the Cisco BTS 10200 Softswitch. EMs are real-time call data records containing information about network usage and activities. They are typically used for billing purposes in a PacketCable-based network. The Cisco BTS 10200 Softswitch (which performs the CMS and MGC functions) transfers EMs to an external RKS that assembles call detail records (CDRs) from the EMs.



Note

Event messages are also transmitted over RADIUS from the Cisco BTS 10200 Softswitch to a CALEA interface, with IPsec for encryption and authentication, and IKE for key management.

Figure 2-7 illustrates the PacketCable NEs and interfaces involved in the generation and processing of EMs.

Figure 2-7 Event Message Interfaces**Notes for Figure 2-7**

1. **MGC to RKS**—EMs generated by the MGC function in the Cisco BTS 10200 Softswitch are sent to the RKS.
2. **CMS to RKS**—EMs generated by the CMS function in the Cisco BTS 10200 Softswitch are sent to the RKS.
3. **CMTS to RKS**—EMs generated by the CMTS are sent to the RKS. The Cisco BTS 10200 Softswitch (MGC/CMS) is not involved.
4. **CMS to CMTS**—The CMS function in the Cisco BTS 10200 Softswitch sends the Billing Correlation ID (BCID) to the CMTS using the DQoS GateSet message.
5. **CMS to MGC**—There is an internal exchange of originating/terminating information such as BCID and FEID.

**Note**

For additional technical discussion, prerequisites, and provisioning steps, see the *Cisco BTS 10200 Softswitch PacketCable Protocol Guide*.

Security Implementation

The implementation of PKT-SP-SEC-I07-021127, *PacketCable Security Specification*, November 27, 2002, provides a security scheme for the voice-over-cable network based on a set of security protocols. These protocols, described in the documents listed below, provide authentication (to help prevent theft of bandwidth, denial-of-service attack, replay, and so forth) and enable message integrity, privacy, and confidentiality.

- IETF documents covering IPsec architecture:
 - RFC 2401, *Security Architecture for the Internet Protocol*, November 1998
 - RFC 2406, *IP Encapsulating Security Payload (ESP)*, November 1998
- IETF documents covering key management protocols IKE and Kerberos with extensions:
 - RFC 2409, *The Internet Key Exchange (IKE)*, November 1998
 - RFC 1510, *The Kerberos Network Authentication Service (V5)*, September 1993, with updates presented in PKT-SP-SEC-I06-021018

The Cisco BTS 10200 Softswitch performs the security functions of the CMS and MGC in the PacketCable environment. It supports security in accordance with PKT-SP-SEC-I07-021127 for both signaling and media:

- Signaling security—For signaling from CMS to eMTA, CMS to CMTS, and MGC to TGW
- Media (bearer) security—For signaling between originating eMTA and terminating eMTA, which is facilitated by the CMS during call signaling setup.

A special parameter, IPSEC_ENABLED, must be set in the optcall configuration file (optcall.cfg) at the time of software installation to enable the IPsec feature. The IPSEC_ENABLED value cannot be changed by use of CLI commands.

**Note**

The value of the IPSEC_ENABLED parameter and all other optcall.cfg parameters for your installation are listed in the *Network Information Data Sheet* that Cisco provided with your system.

