



SIP Devices Support Overview

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The purpose of this guide is to detail the Session Initiation Protocol (SIP) device support, and the changed trunk support in Release 4.5.x. Support for SIP trunks existed in previous releases of the Cisco BTS 10200 Softswitch, but support for SIP devices was new in Release 4.1. The user guide covers areas impacted by the SIP feature, and is an addition to the existing Cisco BTS 10200 Softswitch documentation.



Note

The term “SIP devices” refers to Cisco ATA 186/188 adaptors, and to Cisco IP phones.

The SIP support feature provided in Release 4.5.x was built on the existing Cisco BTS 10200 Softswitch software and hardware platform. The user guide describes the features and protocol changes in Cisco BTS 10200 Release 4.5.x for supporting SIP trunks and subscribers.

The Cisco BTS 10200 Softswitch complex includes a Call Agent (CA), Feature Server (FS), Element Management System (EMS), and an optional HTTP feature server (HTTP-FS) which is comprised of a GUI Feature Server (GUI-FS) and Mini-Browser Adapter (MBA). In this book, all references to the term “Cisco BTS 10200” refer to the Call Agent unless otherwise specified.

SIP protocol support for subscriber features is provided by the Call Agent and the POTS Feature Server.

The Cisco BTS 10200 Softswitch uses SIP and SIP for telephones (SIP-T) signaling to communicate with other SIP-based network elements. The implementation is based upon the evolving industry standards for SIP, including IETF document RFC 3261, SIP: Session Initiation Protocol. The Cisco BTS 10200 Softswitch supports both SIP trunks and SIP-based subscriber lines (SIP devices), and provides the following SIP-related functions:

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

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

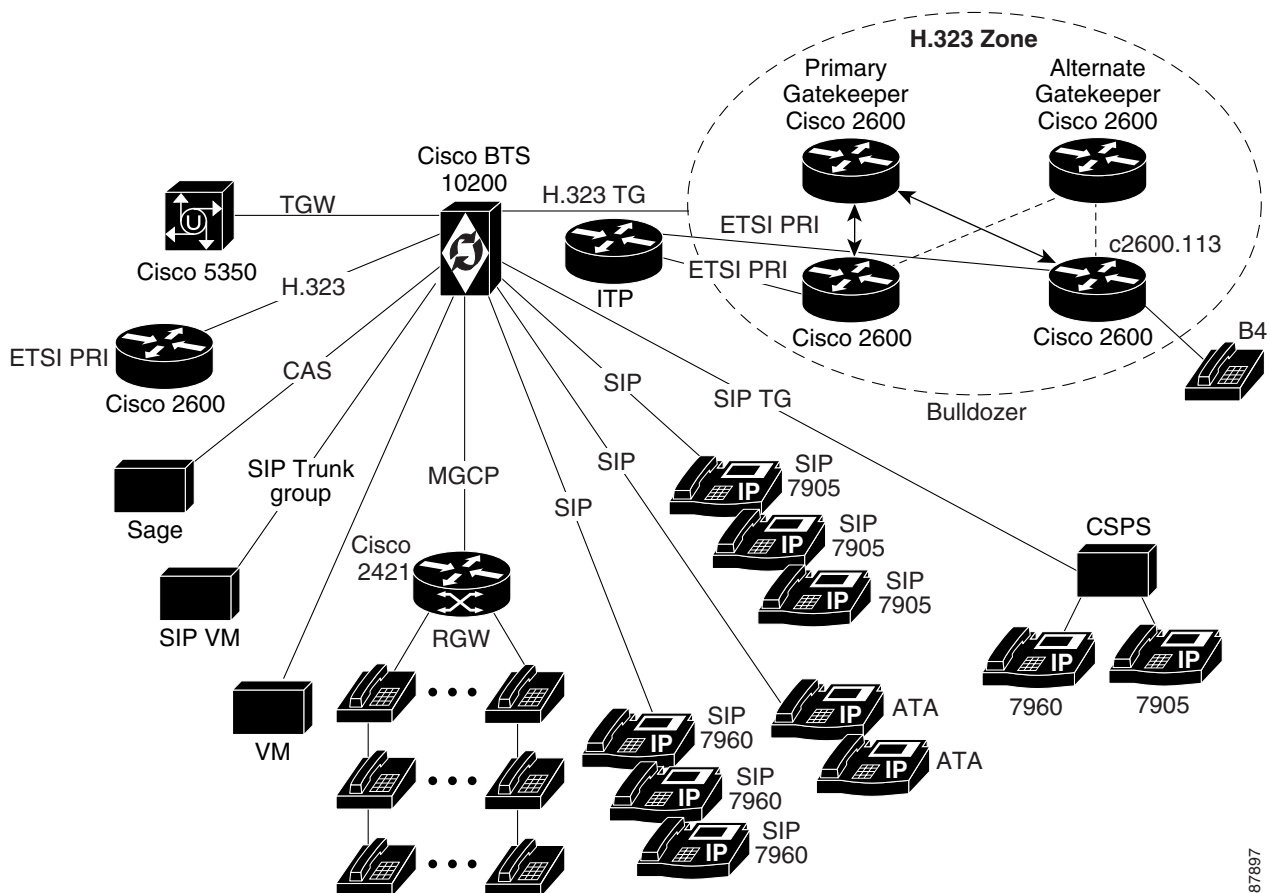
SIP Roles Played by Cisco BTS 10200 Softswitch

Figure 1-1 shows the new Cisco BTS 10200 Softswitch architecture with native support for SIP phone subscribers. In addition to its existing capabilities, in this new architecture, Cisco BTS 10200 provides Registrar (with SIP user authentication) services.

The Cisco BTS 10200 Softswitch supports call processing for SIP trunks and phone users. As a result of native SIP subscriber support, SIP subscribers can use features formerly available only to MGCP subscribers.

For a comparison of the MGCP and SIP feature support, see [Appendix A, “MGCP Features vs. SIP Features.”](#)

Figure 1-1 Cisco BTS 10200 Softswitch Network Architecture



SIP roles performed by the Cisco BTS 10200 Softswitch include:

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

**Note**

The Cisco BTS 10200, as part of the Back-to-Back functionality, plays the role of the UAS and UAC.

Most features provided by the SIP phones comply with LATA Switching Systems Generic Requirements (LSSGR), depending on the phone implementation and capabilities. Due to the nature of the SIP protocol, however, your experience with a feature may differ from what is documented in the LSSGR for that feature.

Limitation On Treatment of Transient Calls

If the active Call Agent experiences a problem and switches over to the standby side, stable calls are preserved. However, calls that are in a transient state (call setup is not complete) might be dropped or improperly set up. During a Call Agent switchover, the Cisco BTS 10200 Softswitch cannot complete call setup for these transient calls.

Transient calls and inactive connected calls originated on the Cisco BTS 10200 Softswitch are cleaned up through a periodic audit mechanism that runs once per hour. The frequency of this audit can be modified. However, changing this requires careful consideration to avoid adverse effects on call processing. Contact Cisco TAC if you have identified a need to change this frequency.

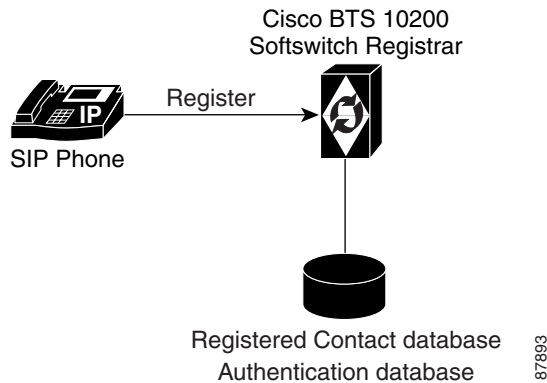
SIP Phone Registrar

SIP Register support enables SIP devices to be served by Cisco BTS 10200 directly without a proxy. The Cisco BTS 10200 acts as a Registrar and authenticates the SIP request.

To initiate a session with a SIP phone, Cisco BTS 10200 must know the current address of the phone. Registration creates bindings in a location service for a particular domain that associate an address-of-record Uniform Resource Identifier (URI) with one or more contact addresses. A user notifies its availability at the address provided in the contact for the specified duration.

The SIP phone registers with Cisco BTS 10200, setting up a binding between the Address of Record (AOR) and its contact address. The registration is valid for a period of time until it expires.

[Figure 1-2](#) demonstrates the SIP phone Registrar function.

Figure 1-2 SIP Phone Registrar Function

Cisco BTS 10200 supported SIP subscribers with SIP phones starting in Release 4.1. SIP users register with Cisco BTS 10200 and originate calls through Cisco BTS 10200. To identify the SIP user, Cisco BTS 10200 uses the challenge-based Digest Access Authentication.

User Agent Client and User Agent Server

The User Agent is a SIP endpoint (such as a phone or a gateway). In this scenario, the User Agent is software running on the endpoint to enable SIP service.

The User Agent can work either as a client or server. When a call is placed, the User Agent Client (UAC) places the request, and the User Agent Server (UAS) services the request and sends a suitable response. The roles change continually, however; for example, with call hold, either user can put the other user on hold.

Figure 1-3 shows the Cisco BTS 10200 working as a UAC, sending out a call request.

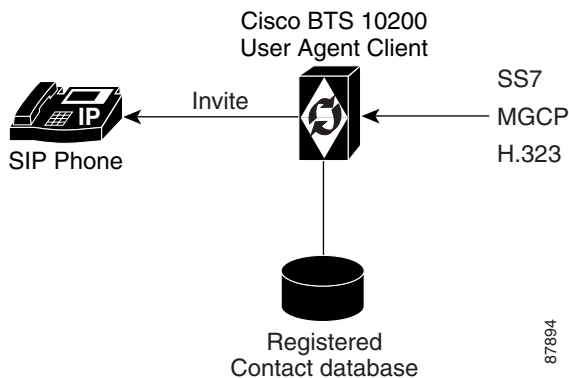
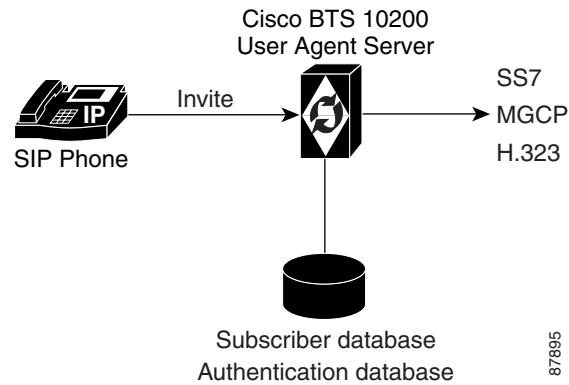
Figure 1-3 User Agent Client

Figure 1-4 shows the Cisco BTS 10200 working as a UAS, accepting a call request.

Figure 1-4 User Agent Server

Back-to-Back User Agent

The back-to-back user agent acts as a UAC and UAS for a single call. It keeps the two call segments separate on the Cisco BTS 10200 Softswitch. Typically, a proxy routes a call, but does not act as a User Agent. The Cisco BTS 10200 acts as a User Agent. In a call between two SIP endpoints (such as SIP phone or SIP trunk), Cisco BTS 10200 terminates the originating half of the call, playing the UAS role, and then sets up the terminating half of the call as a UAC.



Note

There is no provisioning associated with the back-to-back functionality. The Cisco BTS 10200 automatically acts as a back-to-back user agent for a SIP to SIP call.

Figure 1-5 shows the Cisco BTS 10200 working as a back-to-back user agent.

Figure 1-5 Back-to-Back User Agent Server

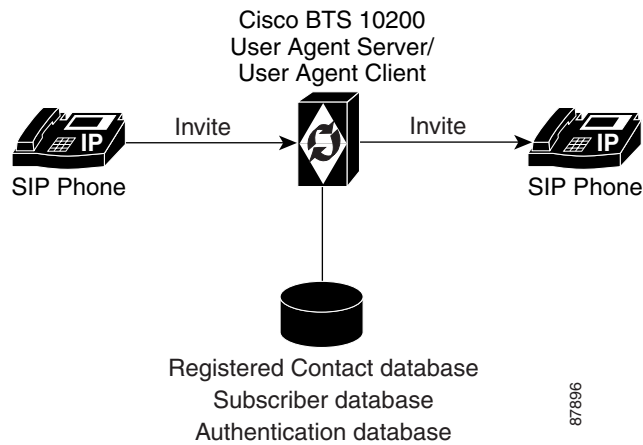


Figure 1-5 shows the call flow for registration.

Figure 1-6 Call Flow for Registration

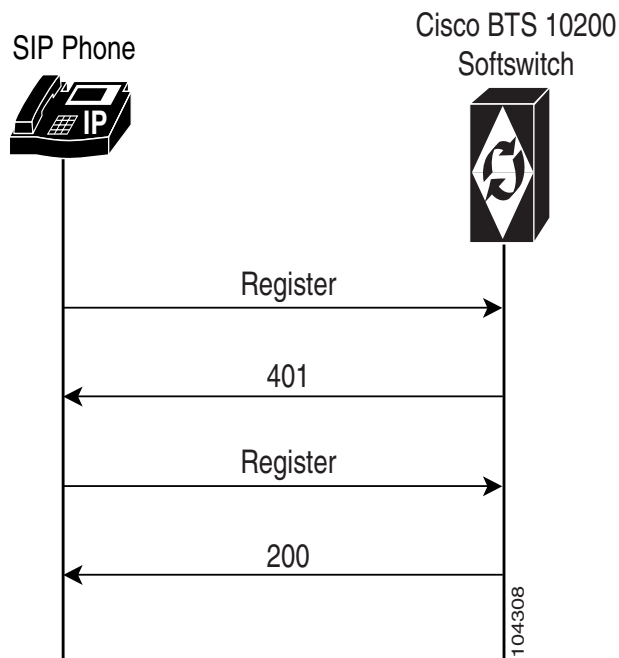
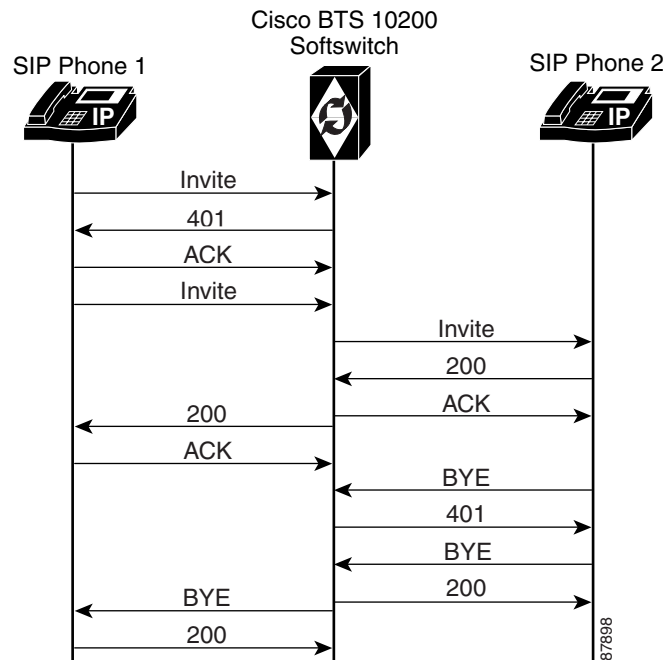


Figure 1-7 shows the call flow for a back-to-back User Agent.

Figure 1-7 Back-to-Back User Agent Call Flow with Authentication



SIP Subscriber Features

SIP Subscriber features are categorized as follows:

- **Cisco BTS 10200 Softswitch-Based Features**—These are provided entirely by the Cisco BTS 10200, and require suitable provisioning of the Cisco BTS 10200.
- **Phone-Based Features**—These are provided exclusively by the phone, which require provisioning on the phone.
- **Jointly-Provided Features**—These are provided jointly. The features are provided partly by the phone's capability, and partly by Cisco BTS 10200 capabilities. To use these features, you must provision both the phone and the Cisco BTS 10200.

New or Enhanced SIP Subscriber Features for Release 4.5.x

Table 1-1 lists the new or enhanced SIP features supported in Release 4.5.x.

Table 1-1 New Cisco BTS 10200 Softswitch-Based SIP Features

SIP Feature	Description	For More Information
T.38 Fax Relay Call Agent Controlled Mode Across SIP Trunk Interface	In previous releases, Cisco BTS 10200 partially supported T.38 fax relay call agent controlled mode in MGCP and H.323 trunk interfaces. Release 4.5.x offers supports for the PacketCable environment, as well as extending T.38 fax support to SIP, NCS, and TGCP trunk interfaces.	Refer to the T.38 Fax Relay feature module.
Secure Provisioning for SIP Endpoints	Enhanced SIP Registration was added to Release 4.5.x to ensure that a SIP REGISTER message to the Cisco BTS 10200 is from a provisioned endpoint. The feature also ensures that the source IP address and contact parameter for all originating calls is from the provisioned SIP endpoint, and that no calls can originate from an unregistered endpoint.	Refer to the Secure Provisioning for SIP Endpoints feature module.
SIP Session Timers	SIP Session Timer values configured prior to Release 4.5.x are reset to the default values after upgrading to Release 4.5.x.	Refer to SIP Timer Values, page 2-23 for an overview description. For provisioning information, refer to Session Timers in the <i>Cisco BTS 10200 SIP Provisioning Guide</i> .
SIP Dynamic Memory Audit	Refer to SIP Dynamic Memory Audit, page 2-18 for a more in-depth description.	Also, refer to the Activity table section in the <i>Cisco BTS 10200 Command Line Interface Reference Guide</i> .

Table 1-1 New Cisco BTS 10200 Softswitch-Based SIP Features

SIP Feature	Description	For More Information
SIP Transport	<p>When a 503 response is received, the entity receiving the response proceeds by submitting the same request as a new transaction (with new branch ID) to the next IP address in the SRV list. Previously, when SIA received the 503 response, it tore down the call.</p> <p>As a result, some calls may not have completed if one of the nodes in an SRV list returned the 503, while other nodes in the list were capable of handling the request successfully.</p> <p>If an SRV server receiving the INVITE does not respond within the retransmission timer period, this enhancement allows for configuring the Cisco BTS 10200 to send the next retransmission of the same request to either the same server (as recommended in RFC3263) or the next server in the SRV list (legacy Cisco BTS 10200 behavior) using a provisionable flag <code>DNS_SRV_ADV_ON_RETRANS_TIMEOUT</code> on the <code>SOFTSW-TG-PROFILE</code> table.</p>	
INVITE Retries Are Configurable	<p>The SIP protocol adapter enables the configuration of the number of INVITE and non-INVITE retries on timer expiry (instead of the fixed 6), and also makes the retry timers configurable. This improves route advancing and completes calls more quickly if one IP address is down.</p> <p>SIP timers are configurable on a per trunk basis, or on a system wide basis for all non-trunk messages. The number of retries and retry interval could be tuned using RFC3261 timers. The guidelines and recommendations on configuring SIP timers will be available in the <i>Cisco BTS 10200 SIP Protocol User Guide</i>.</p> <p>The timers are configurable on a global basis through the Softswitch Trunk Group Profile (SOFTSW Trunk Group Profile) and Call Agent Configuration (CA-CONFIG) tables.</p>	Refer to the SOFTSW Trunk Group Profile and CA-CONFIG sections in the <i>Cisco BTS 10200 Command Line Interface Reference Guide</i> .
SIP Retry	SIP responses received from a Re-Invite or Update sent during an active call that contain a Retry-After header invoke the SIA retry mechanism. Currently, this includes only the SIP 500 class response. The check has been expanded to include other responses.	
SIP Stack Message Size	<p>Previous to Release 4.5.x, the SIP stack could only send and receive messages up to 1500 bytes in size from the network. If a message was bigger than 1500 bytes, only the first 1500 bytes were read and parsed.</p> <p>Release 4.5.x fixes the limitation imposed on the maximum size of the SIP messages which can be sent or received. The SIP stack can now receive/send SIP messages up to 3000 bytes in size.</p>	

Table 1-1 New Cisco BTS 10200 Softswitch-Based SIP Features

SIP Feature	Description	For More Information
SIP Trunk Hop Counter	<p>The SIP Trunk Hop Counter feature was added into Cisco BTS 10200 Release 4.1, but was not provisionable. However, the feature's provisionable components are available in Release 4.5.x, and are provisionable using the SOFTSW Trunk Group Profile table:</p> <ul style="list-style-type: none"> • HOP-COUNTER-MAX • HOP-COUNTER-SUPP • MAX-FORWARDS <p>Note Seven SIP hops are equivalent to one SS7 hop.</p>	For more information, refer to the SOFTSW Trunk Group Profile section in the <i>Cisco BTS 10200 Command Line Interface Reference Guide</i> .
Cisco BTS 10200 Sends SDP for SIP Call When rbk=N for Call Waiting	<p>When a SIP call inbound to the Cisco BTS 10200 is routed to a local subscriber, and this call causes "call waiting" at the subscriber, the originator is provided local ringback indication. However, in this case, the SIP interface was providing remote ringback indication. This may result in the originator not hearing ringback or possibly the discussion of the other dialog.</p> <p>The SIP interface was corrected to provide local ringback.</p>	
SIA Diversion Header	Previously, when one diversion header was received in the initial SIP Invite message, the Original Called Number (OCN) was populated. To be consistent with other adapters, the value is now copied to the Redirect Number (RDN) as well.	
SIP Outbound Numbers Require +CC Depending on NOA	<p>RFC 3398 states that any outbound SIP number with NOA of NATIONAL must be prefixed with "+CCnumber" which is an international format, and any number with NOA=subscriber must be formatted also with international significance. This was not done in previous releases.</p> <p>Sending Full E.164 is enabled by a flag in softsw-tg-profile to enable interworking with downstream devices that require this number format.</p>	
SIP Stack	<p>SIP message loop detection was removed in Release 4.5.x. Previously, the SIP stack did not allow receipt of an un-changed Request URI on hairpinned INVITEs. This required the entity that was hairpinning the SIP call back to Cisco BTS 10200 to modify the user portion of the Request-URI, so that Cisco BTS 10200 does not detect a loop.</p> <p>Since adding the Digman feature, the digit manipulation is performed within Cisco BTS 10200, thereby preventing a requirement for this check. As such, this check is a configurable option that can be toggled through SIA's configuration file for SIP stack.</p>	

Table 1-1 New Cisco BTS 10200 Softswitch-Based SIP Features



SIP Feature	Description	For More Information
DOMAIN-NAME Verified on SIP INVITE Messages	<p>SIP INVITE messages are now validated. This helps detect invalid provisioning or configuration, as well as providing a basic check against intrusion from unauthorized call-agents.</p>  <p>Note This is applicable to the CP environment only.</p> <p>An alarm is generated when an invalid SIP INVITE is detected due to an invalid domain name. The event is corrected with the following steps:</p> <ol style="list-style-type: none"> 1. Verify the provisioned domain name and the one used by SIM to process the command-line parameter. 2. Verify the message source, whether it is on-net request or a potential unspoofed intrusion attempt. <p>A “403 FORBIDDEN MSG” is returned in response to an “INVITE MSG” from an unauthorized/unrecognized CP. The CP processes the call as a basic call.</p> <p>Due to a change in the Logical IP Migration feature, this feature is suppressed by default in Release 4.5.x. To turn on the DOMAIN NAME validation feature, use the -enable_dnv option included in Args= of the platform.cfg for POTS and ASM.</p>	
SIP Stack Attempts Next Address If TCP Connection Fails for INVITE	<p>When provisioning a SIP trunk-grp, the SOFTSW_TSAP_ADDR is usually set to a FQDN that resolves to two or more IP addresses for the destination SIP endpoint(s). Prior to Release 4.5.x, when transmitting a SIP request, if the NON_SRV_TRANSPORT of the softsw-tg-profile is set to TCP and there is a failure to communicate with the first IP address of the FQDN, then no other IP address is tried.</p> <p>In Release 4.5.x, the software has been enhanced so that each of the IP addresses that the FQDN resolves to is tried in succession when there is a failure to communicate with the destination SIP endpoint.</p>  <p>Note This functionality has always worked when UDP is the selected transport.</p>	
SIA Authentication	<p>This is an enhancement for efficient lookup while processing requests targeted for SIP subscribers. It involves internal caching of the Serving Domain Name of the user sending a SIP request and using it for subsequent messages instead of doing a separate table lookup for each request.</p>	

Table 1-1 New Cisco BTS 10200 Softswitch-Based SIP Features

SIP Feature	Description	For More Information
SIP NOTIFY Rejected from Unity	<p>In previous releases, if a request was received from a trunk that had the same domain name as the Cisco BTS 10200 serving domain name in the from header, the call failed, and the MWI Notify from Unity was rejected.</p> <p>This is resolved in Release 4.5.x. In Release 4.5.x, it performs the trunk identification before subscriber identification, and thus identifies the trunk correctly.</p>	
Change Counter Names to SIS and SIP for Release 4.5.x Measurements	<p>The PEG Counters names were changed to match industry terminology. There are two types of SIP counters, those used by multiple stacks related to SIP, and those used by the SIP Adapter.</p> <p>The Cisco BTS 10200 documentation uses the new names for several counters. In Release 4.5.x:</p> <ol style="list-style-type: none"> 1. SIP common counters now begin with SIS_ instead of SIP_ for the SIA, SIM, POTS, and AIN categories. 2. SIA-specific counters now begin with SIA_ instead of SIP_. 3. AUDIT_SIP counters in the SIA category now begin with SIA_AUDIT instead. 	

Table 1-2 lists the SIP subscriber features introduced in previous releases of 4.x.

Table 1-2 Cisco BTS 10200 Softswitch-Based SIP Features

SIP Feature	Acronym
Call forwarding deluxe activation	CWDA
Call forwarding deluxe deactivation	CWDD
Call forwarding deluxe interrogation	CWDI
Call forwarding on busy variable activation	CFBVA
Call forwarding on busy variable deactivation	CFBVD
Call forwarding on busy interrogation	CFBI
Call forwarding on no answer variable activation	CFNAVA
Call forwarding on no answer variable deactivation	CFNAVD
Call forwarding on no answer interrogation	CFNAI
Call forwarding unconditional activation	CFUA
Call forwarding unconditional deactivation	CFUD
Call forwarding unconditional interrogation	CFUI
Do Not Disturb activation	DND_ACT
Do Not Disturb deactivation	DND_DEACT
Anonymous call rejection activation	ACR_ACT
Anonymous call rejection deactivation	ACR_DEACT
Calling identity delivery and suppression (per call)—suppression part	CIDSS
Calling identity delivery and suppression (per call)—delivery part	CIDSD
Calling name delivery blocking	CNAB
Outgoing call bearing activation	OCBA
Outgoing call bearing deactivation	OCBD
Outgoing call barring interrogation	OCBI

Other Common Subscriber Features

Table 1-3 lists the most-commonly used existing subscriber features, some of which have been enhanced or modified for use by the Cisco BTS 10200 with the SIP protocol.



Note

Table 1-3 lists the most commonly used Softswitch-based features; however, it is not an exhaustive list.

Table 1-3 *Most Common Cisco BTS 10200 Softswitch-Based Features*

Feature	Acronym
Activation and Deactivation of Anonymous Call Rejection	ACR
Billing	
Call Forwarding	CF
Calling Name and Number Delivery	CND
Caller ID Suppression	CIDS
Called Party Termination	CPT
Cisco BTS 10200 Supplementary Vertical Service Code Features	VSC
Customer Access Treatment	CAT
Customer-Originated Trace	COT
Direct Inward Dialing	DID
Direct Outward Dialing	DOD
Do Not Disturb	DND
Emergency Call	E911
E.164 and Centrex Dialing Plan (Extension Dialing)	E.164
Incoming and Outgoing Simulated Facility Group	ISFG and OSFG
Interworking	
Multiple Directory Numbers	MDN
Operator Services (0-, 0+, 01+, 00 Calls)	
Outgoing Call Barring	OCB
Remote Activation of Call Forwarding	RACF
SIP Endpoint Caveats	
SIP Subscriber to SIP Calls	
Type of Service	TOS

Phone-Based Features

Table 1-4 lists the phone-based features.

For more information about these features, see the *Cisco BTS 10200 Softswitch Network and Subscriber Feature Descriptions* document.

Table 1-4 Phone-Based Features

Feature	Acronym
Call Hold and Resume	CHD
Call Waiting	CW
Cancel Call Waiting	CCW
Three-Way Calling	TWC
Call Waiting Caller ID	CWCID
CODEC Up-Speeding	CODEC
Do Not Disturb	DND

For features (such as DND) that are available independently on the phones and the Cisco BTS 10200 Softswitch, you can provision either device to enable the feature.



Caution

When provisioning features that are available independently on the switch and the phone, use caution to avoid conflicts between the two.

Jointly-Provided Features

Table 1-5 lists the most commonly used jointly-provided based features.

For more information about these features, see the *Cisco BTS 10200 Softswitch Network and Subscriber Feature Descriptions* document.

Table 1-5 Jointly-Provided Features

Session Timer
Reliable Provisional Responses
Text-GUI Features
Call Transfer (Blind and Attended)
Distinctive Ringing
Distinctive Ringing for Centrex DID Calls

SIP Trunk Features

The SIP Trunk features supported by Release 4.2 are listed in [Table 1-6](#). For more information about the features, see [Chapter 3, “SIP Protocol Trunk Features.”](#)

Table 1-6 **SIP Trunk Features**

Feature	Acronym
Call Redirection	
DNS SRV	
Type of Service	ToS
Reliable Provisional Responses	
Diversion Indication	
Carrier Identification Code over SIP	CIC
Number Portability Information over SIP	NP
Voice-Mail over SIP: Message Waiting Indication	MWI
Voice-Mail over SIP: Cisco BTS 10200 Centrex Subscribers	
SIP Trunk Sub-Groups	
Session Timer	
SIP-T	
DTMF SIP Signaling	
Asserted Identity	
Third-Party Call Control	3PCC
ANI-Based Routing	
TCP/UDP	
Business Group ID	BGID
Trunk Group ID	TGID

Limitations on Number of URLs, Parameters, and Headers (Release 4.5.1, Maintenance Release 2)

The system imposes limits on the decoding of incoming SIP messages. These limits are applicable to both subscriber-related and trunk-related incoming SIP messages. These limitations are intended to protect the system from decoding extremely large messages, which in turn could overload the system and cause performance problems.



Note

These limits are not provisionable. If you need to change any of these limits, contact your Cisco account team.

Table 1-7 lists the limits related to URL and ReqUri.

Table 1-7 Limits on URL and ReqUri

Description	Limit
Maximum number of URLs (SIP+Tel+Unknown) in a SIP message	25
Maximum number of parameters in the ReqUri of a message	10
Maximum number of header parameters (parameters occurring after “?” character) in the Request-URI of a message	5
Maximum number of parameters in a SIP URL	10
Maximum number of header parameters (parameters occurring after “?” character) in a SIP URL	5
Maximum number of parameters in a Tel URL	5

Table 1-8 lists the maximum number of parameters allowed in each type of SIP message header.

Table 1-8 Maximum Number of Parameters Allowed in SIP Message Headers

Header Name	Maximum Number of Parameters Allowed in Header
Contact	10
Via	10
Route	5
Record-Route	5
Diversion	10
Call-Info	5
Alert-Info	5
Error-Info	5
P-Asserted-Identity	5
Accept-Contact	5
To	5
From	5

Table 1-8 Maximum Number of Parameters Allowed in SIP Message Headers (continued)

Header Name	Maximum Number of Parameters Allowed in Header
Referred-By	5
Refer-To	5

Table 1-9 lists the maximum number of unknown Option tags of a specified kind in a SIP message.

Table 1-9 Maximum Number of Unknown Option Tags in SIP Message

Message	Maximum Number of Unknown Option Tags Allowed
Supported	5
Unsupported	5
Require	5

Table 1-10 lists the maximum number of parameters allowed in each type of SIP message header.

Table 1-10 Maximum Number of Parameters Allowed in SIP Message Headers

Header Name	Parameter Type	Maximum Number of Parameters Allowed in Header
Replaces	All parameters	5
Event	All parameters	5
Reason	All parameters	5
Accept	All parameters	5
Session-Expires	All parameters	5
Min-SE	All parameters	5
Warnings	All parameters	5
Accept-Language	Number of languages in the Accept-Language header	5
Accept-Language	Language parameters	5
Accept-Encoding	All parameters	5
Authorization	All parameters	15
Retry-After	All parameters	5

Table 1-11 lists the maximum number of headers allowed in a SIP message.

Table 1-11 Maximum Number of Headers Allowed in a SIP Message

Header Name	Maximum Number of Headers Allowed
Contact	5
Via	5
Route	5
Record-Route	5

Table 1-11 *Maximum Number of Headers Allowed in a SIP Message (continued)*

Header Name	Maximum Number of Headers Allowed
Diversion	5
Call-Info	5
Alert-Info	5
Error-Info	5
P-Asserted-Identity	5
Contact	5
To	1
From	1
Call-ID	1
CSeq	1
Session-Expires	1
Min-SE	1
Referred-By	1
Refer-To	1
Replaces	1
Allow-Events	5
Event	1
Reason	5
Accept	5
Accept-Encoding	5
Authorization	1
Retry-After	1

