CHAPTER 2

SIP Subscribers

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The Cisco BTS 10200 Softswitch supports SIP subscribers on SIP phones that are compliant with RFC 3261 or RFC 2543. This section describes the support for SIP subscribers and how to provision SIP subscriber features.

In this document:

- SIP subscriber means a SIP phone that is registered directly to the BTS 10200 and for which the BTS 10200 maintains subscriber information.
- SIP Automatic Number Identification (ANI)-based subscriber means a SIP phone that communicates with the BTS 10200 over a SIP trunk.

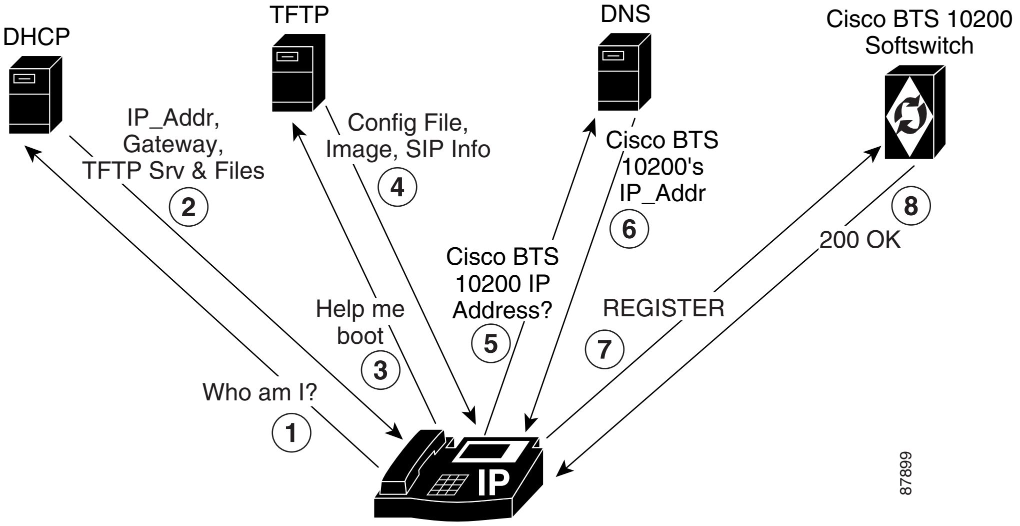
Note For quick-reference tables listing the subscriber features, see the "Comparison of SIP-Based Features and MGCP-Based Features" section on page 2-16.

This section covers the following topics:

SIP Phone Initialization

Figure 2-1 shows an example of SIP phone initialization on bootup, that is, how a typical phone might initialize itself and establish its identity with the BTS 10200. (The image shows actions that occur external to the BTS 10200—it does not show how the BTS 10200 controls SIP initialization.) The circled numbers in the image indicate the numerical order in which the sequence occurs.

Figure 2-1 Example of SIP Phone Initialization


DHCP
TFTP
DNS
Cisco BTS 10200 Softswitch
IP_Addr, Gateway, TFTP Srv & Files
2
Config File, Image, SIP Info
4
Cisco BTS 10200's IP_Addr
6
8
200 OK
Cisco BTS 10200 IP Address?
5
REGISTER
7
Help me boot
3
Who am I?
1
IP
87899

Provisioning a SIP Subscriber

To provision a SIP subscriber, see the "SIP Subscribers" section in the *Provisioning Guide*.

SIP Registration and Security

SIP subscribers use the SIP REGISTER method to record their current locations with the BTS 10200. Registering clients can specify an expiration time for the contacts being registered. However, the BTS 10200 has a minimum and maximum acceptable duration, both of which are configurable.

Note Third-party registration is not supported.

It is possible to register multiple contacts for a single AOR; however, if multiple contacts are registered for a single subscriber, the BTS 10200 uses only the most recently registered contact to deliver the call to that subscriber. For this reason, multiple contacts are not supported.

Note Only one contact should be registered for an AOR.

When a SIP user attempts to register or set up a call, the BTS 10200 challenges the SIP subscriber based on provisioning in the serving-domain-name table. If the serving-domain-name table indicates that authentication is required, the BTS 10200 challenges the SIP request (Register/INVITE) according to the authentication procedures specified in SIP Protocol RFC 3261. If the BTS 10200 receives valid credentials, the authenticated AOR from the user-auth table identifies the subscriber based on the aor2sub table. (For specific provisioning parameters, see the applicable tables in the Cisco BTS 10200 Softswitch CLI Database.)

Registration creates bindings in the BTS 10200 that associate an AOR with one or more contact addresses.

The registration data is replicated on the standby BTS 10200. The BTS 10200 imposes a minimum registration interval as a provisionable value. If the expiration duration of the incoming registration request is lower than the provisioned minimum, a 423 (Interval Too Brief) response is sent to the registering SIP endpoint.

The BTS 10200 generates a warning event when a request from a client fails authentication. This can indicate a provisioning error or an attempt by an unauthorized client to communicate with the BTS 10200.

The contacts registered for an AOR can be looked up using the status command, as demonstrated by the following example.

```
CLI> status sip-reg-contact AOR_ID=4695550184@sia-SYS44CA146.ipclab.cisco.com

AOR ID -> 4695550184@sia-SYS44CA146.ipclab.cisco.com
USER -> 4695550184
HOST -> 10.88.11.237
PORT -> 5060
USER TYPE ->  USER_PHONE_TYPE
EXPIRES -> 3600
EXPIRETIME -> Thu Jan 22 14:33:36 2004

STATUS -> REGISTERED CONTACT

Reply :Success:
```

Enhanced SIP Registration

SIP Registration ensures that a SIP REGISTER message to the BTS 10200 is from a provisioned endpoint, that is, an endpoint with a provisioned secure fully qualified domain name (FQDN) or IP address. The feature also ensures that the source IP address and contact parameter for all originating calls are from the provisioned SIP endpoint, and that no calls can originate from an unregistered endpoint.

Description

Prior to Release 4.5.1, SIP endpoint registration was based on AOR, user ID, and password; there was no verification of the origination of the REGISTER message. Certain service providers may prefer that the source IP address of SIP requests be verified against a provisioned FQDN of the endpoint to address the possibility of theft of VoIP service.

The BTS 10200 can indicate SECURE_FQDN provisioning for specified SIP term-type subscribers. This indication consists of specifying an FQDN with the subscriber AOR. The FQDN is the address/location of the SIP endpoint and is added to the AOR table. The FQDN does not have a service port.

To enable or disable SECURE_FQDN on a successful registered subscriber:

1. Take AOR out of service to remove all registered contacts.
2. Enable or disable SECURE_FQDN for the subscriber.
3. Bring AOR back in service (INS).
4. Reboot the analog terminal adapter (ATA).

A subscriber with the secure FQDN feature enabled has the following characteristics:

- One and only one AOR is associated with the endpoint.
- Does not have any static-contact associated with it.
- User ID and Password Authentication are supported.
- One FQDN (specified without service port).
- The DNS lookup of the FQDN should result in one and only one IP address.
- Cannot place or receive a call unless successfully registered.

Example

This example presents a case in which a VoIP subscriber (Subscriber 1) uses the following options for the user ID, password, and phone number:

- user-id-1
- password-1
- phone-no-1

Without security, another VoIP subscriber, Subscriber 2, could access Subscriber 1's information (perhaps by getting a Cisco ATA configuration file with the encryption key in clear text, and then getting the full configuration file with all the data). Subscriber 2 could then register with the BTS 10200 with Subscriber 1's combination of user-id-1, password-1, and phone-no-1, as well as Subscriber 2's own IP address. Without the secure FQDN feature, the Cisco BTS 10200 would accept this information unless specific measures were taken, and Subscriber 2 could steal service and make calls on behalf of Subscriber 1.

Provisioning Commands

This section shows the CLI commands you need to provision a secure FQDN of a SIP endpoint.

Note Use this procedure to provision subscribers on the BTS 10200. The procedure does not cover the security of configuration files provisioned on the SIP adapter (for example, an ATA), which are the responsibility of the service provider.

The SECURE_FQDN token is present in both the subscriber and aor2sub tables. A non-null value in the field indicates that the SECURE_FQDN validations apply to all SIP messages received from the endpoint associated with that AOR.

- The SECURE_FQDN value can be specified on a subscriber only if the AOR for the subscriber is out of service (OOS). When an AOR is taken administratively OOS, its registered contacts are deleted.
- A static contact cannot be specified for a SECURE_FQDN subscriber. Any existing static contact record for an AOR must be deleted before the subscriber can be made a SECURE_FQDN SIP endpoint.

- The SECURE_FQDN in the aor2sub table is stored both in the Oracle database and the shared memory.

The aor2sub records cannot be added or deleted directly. To add aor2sub records, you must specify specify the AOR ID on a subscriber record.

Provision a New SIP Subscriber

Step 1 To provision a new SIP subscriber with the secure FQDN feature, enter the following command:

Note This command automatically adds a corresponding entry in the aor2sub table.

```
add subscriber id=sub1; sub-profile-id=subpf1; category=individual; dn1=241-555-1018;
term-type=SIP; aor-id=<aor-id of SIP adapter port for sub1>; secure-fqdn=<secure-fqdn of
the SIP adapter>;
```

Step 2 (Optional) To provision an additional subscriber on the same SIP adapter, enter the following command:

```
add subscriber id=sub2; sub-profile-id=subpf1; category=individual; dn1=241-555-1022;
term-type=SIP; aor-id=<aor-id of SIP adapter port for sub2>; secure-fqdn=<secure-fqdn of
the SIP adapter>;
```

Note If there are multiple subscribers on a single SIP adapter (such as an ATA), these subscribers might share the same IP address. Therefore, you can provision all of these subscriber records with a single SECURE_FQDN, and in the DNS, this FQDN can point to the applicable IP address. The id, dn1, and aor-id tokens must have unique values for each subscriber.

Enable or Disable Secure FQDN for an Existing Subscriber

To enable or disable the secure FQDN feature for a successfully registered subscriber, enter the following commands:

Step 1 Take the AOR OOS. This command removes all registered contact.

```
change aor2sub aor-id=241-555-1018@sia-SYS41CA146.ipclab.cisco.com; status=oos;
```

Step 2 To enable the secure FQDN feature for an existing subscriber, enter the following command:

```
change subscriber id=sub1; secure-fqdn=ata-SYS41CA146.ipclab.cisco.com
```

To disable the secure FQDN feature for an existing subscriber, enter:

```
change subscriber id=sub1; secure-fqdn=null
```

Note If SECURE_FQDN is not provisioned for the subscriber, the system does not provide the secure FQDN feature to that subscriber. If SECURE_FQDN has previously been provisioned for the subscriber, setting SECURE_FQDN to null disables the feature.

Step 3 To bring the AOR back INS, enter the following command:

```
change aor2sub aor-id=241-555-1018@sia-SYS41CA146.ipclab.cisco.com; status=ins;
```

Step 4 Reboot the adapter device (such as ATA) for this subscriber.

Operations

The system performs the following checks. If any of the following conditions are not met, the request is rejected, and an alarm is generated.

No Calls to or from an Unregistered Secure-Provision SIP Endpoint

An unregistered secure-provision SIP endpoint cannot originate or receive calls.

Third-Party Registrations for Secure FQDN Endpoint Not Allowed

Third-party registrations for secure FQDN endpoints are not allowed.

BTS 10200 Challenges Registration

On receiving a REGISTER message from a secure-provision SIP endpoint, the BTS 10200 challenges the registration, asking for authentication. Verification of the resend REGISTER message with user ID and Password is as follows, after the user ID and Password are authenticated:

- Ensure that there is only one contact in the contact header.
- Ensure that the source IP address of the REGISTER message is the same IP address of the provisioned FQDN for that endpoint.
- Ensure that the IP address or the FQDN of the contact is the same as the provisioned FQDN for that endpoint.

If any of these conditions are not met, registration is rejected and a security event and alarm is generated, indicating that the source of the registration is illegal.

The contact address can verify all subsequent SIP request source IP address of the request from the endpoint until the registration expired or is deregistered.

Registration Expires

If the registration expires or the end point de-registers, the registration process in the "BTS 10200 Challenges Registration" occurs before any new calls are accepted.

Call Originates From or Terminates to a Secure-Provision SIP Endpoint

When a call originates from or terminates to a secure-provision SIP endpoint:

1. The system authenticates the user ID and password on all messages requiring authentication.
2. If the Contact header is available, the system ensures that only one contact is present, and that it has the same IP address or FDQN of the provisioned endpoint.
3. All messages sent by the endpoint and the source IP address of the message must be the same as the internal cache contact address (for example, the cache contact address is the contact obtained during registration).
4. Response from an endpoint that has a contact header must conform to the second item in this list.

Call Processing

The SIP application in the BTS 10200 implements the secure provisioning feature for all incoming SIP messages (requests and responses) from SIP endpoints.

When a SIP request message is received from a SIP endpoint and Auth_Rqed=Y for the serving domain, the request is challenged. When the request is resubmitted with credentials, the AOR of the authenticated SIP endpoint is used to perform the SECURE_FQDN validation, provided a SECURE_FQDN value is provisioned in the AOR2SUB record. If AUTH_REQD=N, the SECURE_FQDN validation is performed without the request being challenged.

Validation

The validation processing for a SIP request, that comes from a SIP endpoint provisioned with this feature, is as follows:

1. The SECURE_FQDN validation occurs on every request (including CANCEL/ACK).
2. The SECURE_FQDN is verified to have a DNS resolution, if it is a domain name. If there is no DNS resolution, a 500 Internal Server Error response is returned.
3. The DNS resolution for the SECURE_FQDN is verified to yield a single IP address Secure-IP1.

   If the address is incorrect, a 500 Internal Server Error response is returned.
4. The Source IP address of the packet is verified as identical to Secure-IP1.

   If the address is not identical, a 403 Forbidden response is returned.
5. If the Request is a REGISTER, it is verified to have a single Contact header.

   If there is not a single contact header, a 403 Forbidden response is returned.
6. If the SIP request is an initial INVITE (including an INVITE resubmitted with credentials), it is verified that there is an unexpired registered contact for the AOR.

   If there is not an unexpired registered contact, a 403 Forbidden response is returned.
7. When a Contact header is present, the Contact FQDN/IP address of the request is verified to yield a single IP address Secure-IP1.

   If it does not yield the proper address, a 500 Internal Server Error response is returned.
8. The IP address of the Contact host is verified as identical to the IP address Secure-IP1 of the SECURE_FQDN.

   If the addresses are not identical, a 403 Forbidden response is returned.
9. The provisioning of a static contact on a AOR is not disabled, but any provisioned value is ignored because of the SECURE_FQDN validation rules. A static contact is irrelevant for SECURE_FQDN AORs, since the SIP request is denied if no registered contact exists.
10. The To and From header URLs in a REGISTER request are verified to be identical, for SECURE_FQDN subscribers. This is to block third-party registration.

Received SIP Response Message

When a SIP response message is received from a SIP endpoint, the following occurs:

1. The Source IP address of the packet is verified to be identical with the IP address of the Secure-IP1.

   If the addresses are not identical, the response is dropped. This has the same result as the non-receipt of that response, such as would happen with a call failure.
2. When a Contact header is present on a reliable 1xx or 2xx response, the Contact FQDN/IP address of the response is verified to resolve to the Secure-IP1.

   If the address does not resolve properly, the response is dropped. This has the same result as the non-receipt of that response, such as would happen with a call failure.
3. The response for a BYE sent by the BTS 10200 is not validated. This is the least likely point in a call for theft.

Rules for Sending a SIP INVITE Message from the BTS 10200

When a SIP INVITE message is sent to a SIP endpoint, the following occurs:

1. The INVITE is sent to the registered contact of the endpoint. If there is no registered contact or if the registered contact has expired, the INVITE is not sent and the call is declined.
2. Any static contact provisioned for the subscriber is ignored.

Note Provisioning of static contact is not allowed for secure SIP endpoints; therefore, these rules are merely due diligence.

Validation of ACK Request

When a SIP ACK message is received from a SIP endpoint, the following occurs:

1. The ACK for a 200-class response is validated like any other SIP request.
2. The ACK for a failure response (3xx or higher) is not validated.

Measurements

The following measurements are supported for secure FQDN violations:

- A SIA-SECURE_FQDN-VIOLATION-REQ counter is incremented when a SIP request fails the validation for secure SIP endpoints.
- A SIA-SECURE_FQDN-VIOLATION-RESP counter is incremented when a SIP response fails the validation for secure SIP endpoints.

Note For a full list of measurements, see the *Cisco BTS 10200 Softswitch Operations and Maintenance Guide*.

Events and Alarms

A Warning event is raised when a SIP request or response fails the validation for secure SIP endpoints. The alarm has the following attributes:

Type: SECURITY(6)

DESCRIPTION: Secure SIP Endpoint Validation Failure

SEVERITY: WARNING

Note For a full list of events and alarms, see the *Cisco BTS 10200 Softswitch Troubleshooting Guide*.

SIP User Authentication

The BTS 10200 can act as an authentication server. Authentication is enabled on the serving domain through provisioning.

Whenever a SIP request is received from a SIP subscriber, the request is authenticated to ensure it is indeed from an identified user. Authentication also enables request authorization, because users may be authorized to perform only specific requests.

The following examples are the functional scenarios in which authentication is required:

- When a SIP user registers a contact with the BTS 10200 Registrar using a REGISTER request.
- When a SIP user initiates a call using an INVITE request.
- When a SIP user sends any request in an ongoing call. Examples include:
  - Renegotiation of the call parameters using a re-INVITE
  - Terminating the call using a BYE
  - Initiating a call transfer using a REFER
- When a SIP user sends a request outside a dialog. Example: OPTIONS.

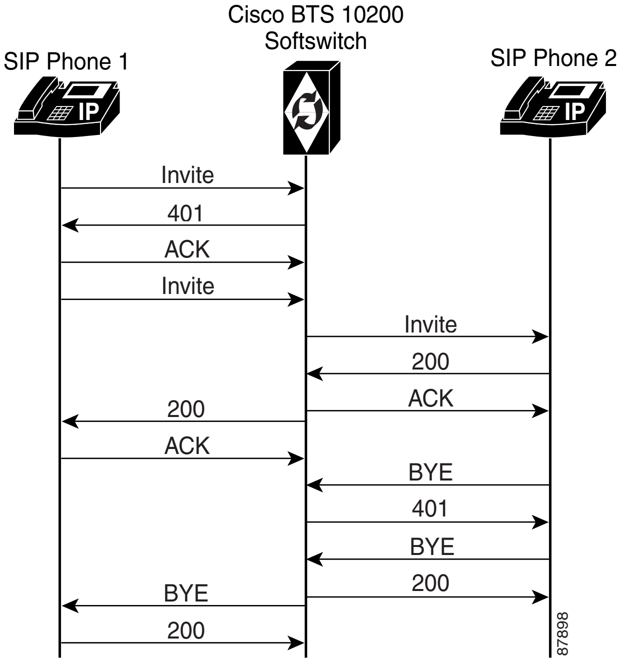
The following tables affect authentication for SIP subscribers:

- aor2sub
- serving-domain-name
- auth-realm
- user-auth

See the Cisco BTS 10200 Softswitch CLI Database for more information about the tables.

Figure 2-2 shows how an incoming request is processed and indicates the role of the Authentication Service in the BTS 10200.

Figure 2-2 Authentication and Processing of an Incoming Request (for Example, INVITE)



The BTS 10200 validates the hostname of the REQURI of every incoming SIP request against the list of names provisioned in the serving-domain-name table. The BTS 10200 hostname used by devices (in the REQURI), when they send requests to the BTS 10200, should be provisioned in the serving-domain-name table of that BTS 10200. If a name is not provisioned (and therefore not found) in the serving-domain-name table, the BTS 10200 rejects the SIP request with a “404 Not Found REQURI Serving Domain” response.

The BTS 10200 authenticates IP phones by using the MD5 digest defined in RFCs 3261 and 2617. The BTS 10200 verifies a user’s credentials on each SIP request from the user. For more information, see the User Authorization table in the Cisco BTS 10200 Softswitch CLI Database.

SIP Subscriber Calls

SIP subscribers must present valid credentials on a SIP INVITE message in order to place calls.

The system allows SIP subscribers to call other SIP subscribers or SIP trunks connected to the BTS 10200. The provisioned dial plan determines whom a subscriber can call. A SIP subscriber can receive a call as long as the subscription’s registration is current, or a static registration has been provisioned.

Provisioning Session Timers for SIP Subscribers

The system uses session timers to periodically refresh SIP sessions during call processing or in-progress calls. You can enable or disable session timers for calls to and from all SIP subscribers on the BTS 10200 through the SUB_SESSION_TIMER_ALLOWED parameter in the ca-config table. The timers are disabled by default.

Use the commands in this section to provision session timers for SIP subscribers. Session timer defaults for subscribers are preset in the system. The timers can be adjusted through the commands shown in this section.

Note For a detailed description of session timers, see "SIP Session Timers" section on page 4-7.

Step 1 Adjust the session timer values in the SIP Timer Profile (sip-timer-profile) table.

Note The session duration field value is in seconds with a range of 100 to 7200.
The minimum session duration field value is in seconds with a range of 100 to 1800.

We recommend a value of at least 1800 for each of these fields.

```
add sip_timer_profile id=<timer_profile_id>; session_expires_delta_secs=7200; min-se=1800;
```

Step 2 Enable session timers for SIP subscribers:

```
add ca-config type=SUB-SESSION-TIMER-ALLOWED; datatype=BOOLEAN; value=Y;
```

Step 3 If not already done, add a default sip-timer-profile-id to the ca-config table:

```
add ca_config type=SIP-TIMER-PROFILE-ID; datatype=STRING; value=<sip_timer_profile_id>;
```

SIP Timer Values for SIP Subscribers

Note This section describes how to provision SIP timer values for SIP subscribers. For a comprehensive listing of SIP timers, see Chapter 4, "SIP System Features."

You can customize SIP timers through the sip-timer-profile table. A record in this table can then be configured to apply to all subscribers switch-wide. The system operates with default SIP protocol timer values, as noted in the SIP specification. These default values are adequate for many installations. If customization is required, a sip-timer-profile table can be provisioned and associated with all calls.

Use the following steps to provision the SIP timer values.

Step 1 Adjust the SIP timer values in the sip-timer-profile table if necessary (example shown).

```
add sip-timer-profile id=<timer_profile_id>; timer-t1-milli=500;
```

Step 2 If not already done, add a default sip-timer-profile-id to the ca-config table:

```
add ca-config type=sip-timer-profile-id; datatype=string; value=<sip_timer_profile_id>;
```

Diversion Indication for SIP Subscribers

Diversion indication provides supplemental redirection information to the SIP entity receiving a call. The SIP entity uses this information to identify from whom the call was diverted, and why the call was diverted. It also provides information for each redirection if multiple redirections occurred. This is provided in the form of a SIP Diversion header.

Forwarding information allows applications such as SIP voice-mail servers to access the mailbox of the original called party for proper outgoing greeting and message deposit when a forwarded call is received. Billing systems also use the information to determine the charged party of the call where it is the last forwarding party that is billed.

The BTS 10200 supports the Diversion Indication feature according to the specifications in the IETF document draft-levy-sip-diversion-02.txt, *Diversion Indication in SIP*. For incoming calls, the BTS 10200 uses the party number information from the top-most and bottom-most diversion headers. The BTS 10200 reads the diversion count across all diversion headers to determine the total diversion count. For outgoing calls, The BTS 10200 sends 0, 1, or 2 diversion headers, depending on the forwarding information of the call.

Diversion header parameter support is limited to the diversion counter and the diversion reason. These two parameters in diversion headers are populated for outgoing calls and interpreted on incoming calls.

For INVITEs sent out by the BTS 10200, the following behavior applies:

- If no diversion information is available, no diversion headers are included.
- If there is an original called party, one diversion header is added to the outgoing INVITE message.
- If there is a last forwarding party, a second diversion header is added on top of the original called party diversion header.
- Each outgoing diversion header is populated with the party number, the diversion reason, and the diversion count.
- Privacy parameters are sent and received in the Diversion header.
- If the original called number (OCN) and/or the redirected DN (RDN) are being sent in Diversion headers towards local SIP subscribers, and the presentation value is not allowed, the system applies anonymous to them as follows:
  - If an OCN exists, it populates the URL as anonymous@anonymous.invalid in the To header.
  - If a Diversion header is added, it populates the user part of the diversion header with anonymous.

SIP Privacy Header

The SIP Privacy Header feature provides privacy services to the caller to withhold personal information (caller details) from the parties involved in a call. This feature enables the user to request privacy functions from the Cisco BTS 10200 operating as a SIP network-based privacy service.

This feature allows the Cisco BTS 10200 SIP interface to apply privacy services using the Privacy header (as defined in RFC 3323). The Privacy header is received by Cisco BTS 10200 from an originator requesting privacy or from a SIP proxy on the originator's behalf. If privacy services are requested, they are applied to an initial INVITE message sent out on a SIP trunk or sent towards a terminating SIP subscriber.

This feature does the following:

- Applies privacy services exclusively to initial INVITE requests sent from a Cisco BTS 10200 SIP interface
- Provides support for privacy by interpreting the set of services requested in the Privacy header
- Allows to set the calling number restriction when the Cisco BTS 10200 SIP interface receives the initial INVITE requests

When privacy is requested and applied, Cisco BTS 10200 adds privacy information to the initial outbound INVITE request. It assigns anonymous entries to the user, the display name, and host field of the From header and the user field of the Contact header. A single VIA header is set with the host name of this local Cisco BTS 10200 (this follows normal back-to-back user agent behavior).

When privacy is applied, Cisco BTS 10200 provides anonymous entries to the Form header and the User field of the Contact header, as shown here:

```
FROM: "Anonymous" sip:Anonymous@Anonymous.invalid;tag=AParty
Via: <single VIA with local BTS host>
Contact: sip:Anonymous@localhost:5060
```

SIP Signaling Details

In SS7-to-SIP calls on Cisco BTS 10200, the user and header privacy services are applied to the outbound SIP INVITE message, if restrictions are applied to the calling name and number fields.

For SIP-to-SIP calls on Cisco BTS 10200, the header level privacy is always applied to initial outbound SIP INVITE messages. If header level privacy is requested, the header privacy token is handled in the following way:

- The privacy services are enabled. It is assumed that the header level privacy is applied. The token is removed from the Privacy header.
- If the privacy services are not enabled, the Cisco BTS 10200 indicates that header level privacy was not applied, even though the back-to-back user agent (UA) operations implicitly provided that level of privacy. The token remains on the privacy header.

Note This feature does not support a session level privacy service because Cisco BTS 10200 does not terminate or manage the media path. In case of SIP-to-SIP trunk calls, Cisco BTS 10200 passes the privacy token and any of the privacy services to the Cisco BTS 10200 SIP interface (which does not render).

The Cisco BTS 10200 SIP interface applies the user level privacy to the outbound SIP INVITE request if any privacy services are requested, even if the user privacy was applied previously by the originator. Privacy services are not applied under the following conditions:

- If the ID privacy service is requested, the P-Asserted-ID (PAID) header (as defined in RFC 3325) is not sent to local SIP subscribers. The Cisco BTS 10200 SIP trunks are assumed to be pointing towards trusted SIP devices. Therefore, the PAID header is always sent out to the SIP trunks (assuming that the SIP trunk is provisioned to allow sending the PAID header).

- If the NONE privacy service is requested, Cisco BTS 10200 does not apply privacy services. In this case, if the call is routed out a SIP trunk, the Privacy header passes the NONE token outbound without modification.
- The Cisco BTS 10200 ignores the Proxy-Require header that requests the privacy service for incoming messages. This header is not passed through in a SIP-to-SIP call through the Cisco BTS 10200.
- For an outbound SIP trunk call, privacy services requested from the originator which are not rendered by the Cisco BTS 10200 SIP interface are represented in the Privacy header as remaining tokens and forwarded to the next SIP device. For an outbound SIP subscriber call, the Privacy header is not sent under any condition.

In general, the Cisco BTS 10200 does not add privacy services to initial outbound INVITE messages. However, if a SIP trunk is provisioned to send the PAID header, and the calling number presentations are restricted, then the Privacy header with ID token is sent in the INVITE request. This was Cisco BTS 10200 behavior maintained prior to this feature.

PRIVACY Token

The PRIVACY token is in the SUBSCRIBER table. NONE is a value that can be set for the PRIVACY token. When the user requests that no privacy services be applied, the PRIVACY token with a NONE value is applied in the originating device, regardless of provisioning or defaults.

Note For SIP-to-SIP calls, the PRIVACY token is passed through if the call is routed out a SIP trunk on the Cisco BTS 10200.

Privacy services are not applied on the terminating SIP side when the PRIVACY token with a NONE value is received, regardless of any provisioning settings or privacy indications.

When the Cisco BTS 10200 receives the PRIVACY token with a NONE value, the following conditions hold true:

- The PRIVACY token does not affect the Cisco BTS 10200 SIP interface configuration that involves the selection of FROM or PAID SIP headers for deriving call information when a SIP call is received.
- If a Cisco BTS 10200 SIP interface is provisioned to interpret the calling party using the PAID header when an initial INVITE is received, the calling name and number presentations are set to Allowed.
- If a Cisco BTS 10200 SIP interface is provisioned to interpret the calling party using the FROM header when an initial INVITE is received, the PRIVACY token does not affect how the calling name and number are mapped by the FROM header.
- When the user terminates the SIP calls from the Cisco BTS 10200 SIP interface, the PRIVACY token does not affect how the calling name and number presentations are applied in the FROM or PAID headers towards that terminated calls.

Note For more information on calling name and number mapping on the Cisco BTS 10200 SIP interface, check with technical support.

Feature Interactions

The SIP Privacy Header feature interacts with the Cisco BTS 10200 and SIP features. The TGID draft (R5) applies the TGID user information parameters in the contact header for outbound SIP trunks. When the privacy services are applied, the personal information of the user is removed because the user part of the contact header is anonymous.

Prerequisites

The user should have knowledge of RFC 3323 and RFC 3325 before using this feature.

Limitations

The SIP Privacy Header feature of the Cisco BTS 10200 Softswitch has the following limitations:

- The Cisco BTS 10200 does not support session level privacy because it does not terminate or manage the media path.
- All outbound SIP trunks are expected to be provisioned within the trusted domain. Therefore, Cisco BTS 10200 SIP trunks do not apply privacy services to the callers who are on different domains (as per RFC 3325).

Feature Considerations

The user has to consider the following points before using the SIP Privacy Header feature:

- In RFC 3323, the NONE token in the Privacy header is sent or received as a single token. It is invalid to send or receive the Privacy CRITICAL token as a single token.
- It is invalid to receive an anonymous or non-existent user information field in the PAID header, as the purpose of the PAID header is to assert an identity.
- The Cisco BTS 10200 SIP interface (in keeping with PacketCable 1.5) supports an anonymous name display field in the PAID header that indicates the restricted name.
- If the Privacy tokens "session” and “critical" are received and the call is routed towards a SIP subscriber, the call fails. This occurs because the Cisco BTS 10200 cannot apply the session privacy service, and there no intermediary switch that can route the call.

Provisioning

Use the following flags to provision the privacy services:

- USE_PAI_HDR_FOR_ANI
- APPLY_USER_PRIVACY
- SIA_SUB_SEND_PAID_HDR

USE_PAI_HDR_FOR_ANI

The USE_PAI_HDR_FOR_ANI flag is in the SOFTSW-TG_PROFILE table. If the customer wants the Privacy header feature to handle the privacy ID token on SIP trunks, the USE_PAI_HDR_FOR_ANI flag on the SIP trunk profile must be enabled. In the USE_PAI_HDR_FOR_ANI flag, the disable privacy service has the following conditions:

- When this flag is disabled, the ID token in the Privacy header and the entire PAID header are ignored if received, and they are not sent under any condition.
- When this flag is disabled, the ID token is not sent, regardless of the Privacy header feature enabled for outbound SIP trunks.

APPLY_USER_PRIVACY

The APPLY_USER_PRIVACY flag is in the SOFTSW-TG_PROFILE table. The APPLY_USER_PRIVACY SIP trunk profile flag can enable or disable privacy services on the terminating SIP trunk. If the originator requests a privacy service, the calling party information in the initial outbound SIP INVITE is set to anonymous, in order to hide the caller identity. Privacy is requested when the calling party name and/or number indicate presentation restrictions. Privacy is also requested when the Cisco BTS 10200 SIP interface receives a SIP call with a Privacy header containing privacy service requests. Regardless of what privacy function is requested, the Cisco BTS 10200 SIP interface provides only User and Header level privacy.

SIA_SUB_SEND_PAID_HDR

A new flag SIA_SUB_SEND_PAID_HDR is added to the CA-CONFIG table. This flag can have a Yes (Y) or No (N) value. The default value is N (default disabled). Use this flag to determine if the PAID header is sent to SIP subscribers. If the flag is enabled, the PAID header is sent if the calling party screening indicator is set to “network provided”, and the Privacy: ID token did not exist in the originating message. This flag applies only to terminating SIP subscribers.

Comparison of SIP-Based Features and MGCP-Based Features

Table 2-1 lists the MGCP features available (in the MGCP-Based Feature column) and then describes how the feature differs when it is used as a SIP feature.

Table 2-1 MGCP Features and SIP Support

MGCP-Based Feature	Abbreviation	Support for SIP Phone Compared to Support for MGCP-Based Phone
8XX Toll-Free	8xx	Same as MGCP.
911 Emergency-Service	911	Only E911 support (without the suspend procedure for 45 minutes). Basic 911 with suspend procedure is not supported. Emergency Call (911) is supported for SIP endpoints with one caveat: If the calling party (SIP subscriber) disconnects the call, the called party control is not available. Otherwise, the call will be released. Expanded emergency service (E911) does not require this, but basic emergency service (911) does. Both 911 and E911 are supported for MGCP endpoints. The Public Safety Answering Point (PSAP) is selected based on default user location. No mobility is supported.

Table 2-1 MGCP Features and SIP Support (continued)

MGCP-Based Feature	Abbreviation	Support for SIP Phone Compared to Support for MGCP-Based Phone
Anonymous Call Rejection	ACR	Same as MGCP, when provided by the BTS 10200. Also provided by the phone.
Anonymous Call Rejection Activation	ACR_ACT	BTS 10200 functionality is same for SIP subscribers as for MGCP. ACR_ACT is also supported on some SIP phones. Depending on the specific phone, the feature on the BTS 10200 might work jointly with the feature on the phone.
Anonymous Call Rejection Deactivation	ACR_DEACT	BTS 10200 functionality is same for SIP subscribers as for MGCP. ACR_DEACT is also supported on some SIP phones. Depending on the specific phone, the feature on the BTS 10200 might work jointly with the feature on the phone.
Automatic Callback	AC	SIP phone users cannot activate the service. MGCP users cannot activate the service toward SIP phone users.
Automatic Callback Activation	AC_ACT	SIP phone users cannot activate the service. MGCP users cannot activate the service toward SIP phone users.
Automatic Callback Deactivation	AC_DEACT	SIP phone users cannot activate the service. MGCP users cannot activate the service toward SIP phone users.
Automatic Recall	AR	SIP phone users cannot activate the service. MGCP users cannot activate the service toward SIP phone users.
Automatic Recall Activation	AR_ACT	SIP phone users cannot activate the service. MGCP users cannot activate the service toward SIP phone users.
Automatic Recall Deactivation	AR_DEACT	SIP phone users cannot activate the service. MGCP users cannot activate the service toward SIP phone users.
Busy Line Verification	BLV	Not supported.
CALEA and LI	—	For information on lawful intercept (LI) and Communications Assistance for Law Enforcement (CALEA), see the "Lawful Intercept and Enhanced CALEA" chapter in the *Cisco BTS 10200 Softswitch Network and Subscriber Feature Descriptions*.
Call Block	CBLK	Same as MGCP.
Call Forward Busy [1]	CFB	Same as MGCP.
Call Forward Busy Variable Activation	CFBVA	Single-stage digit collection.
Call Forward Busy Variable Deactivation	CFBVD	Same as MGCP.
Call Forward Busy Interrogation	CFBI	Single-stage digit collection.
Call Forward Combined [1]	CFC	Same as MGCP.
Call Forward Combined Activation	CFC_ACT	Single-stage digit collection.
Call Forward Combined Deactivation	CFC_DEACT	Same as MGCP.
Call Forward No Answer [1]	CFNA	Same as MGCP.

Table 2-1 MGCP Features and SIP Support (continued)

MGCP-Based Feature	Abbreviation	Support for SIP Phone Compared to Support for MGCP-Based Phone
Call Forward No Answer Variable Deactivation	CFNAVA	Single-stage digit collection.
Call Forward No Answer Variable Deactivation	CFNAVD	Same as MGCP.
Call Forward No Answer Interrogation	CFNAI	Single-stage digit collection.
Call Forward Unconditional [1]	CFU	Same as MGCP.
Call Forward Unconditional Activation	CFUA	Single-stage digit collection.
Call Forward Unconditional Deactivation	CFUD	Same as MGCP.
Call Forward Unconditional Interrogation	CFUI	Single-stage digit collection.
Call Hold	CHD	Functionality provided by the phone. The BTS 10200 supports the interface.
Call Park	CPRK	Not supported.
Call Park and Retrieve	CPRK_RET	Not supported.
Call Transfer	CT	For SIP phones, this feature is provided as part of REFER support on the BTS 10200. See the "Call Transfer (Blind and Attended) with REFER" section on page 2-38 for details.
Call Waiting	CW	Functionality provided by the phone. The BTS 10200 supports the interface.
Call Waiting Deluxe	CWD	Varies with phone functionality.
Call Waiting Deluxe Activation	CWDA	Varies with phone functionality.
Call Waiting Deluxe Deactivation	CWDD	Varies with phone functionality.
Call Waiting Deluxe Interrogation	CWDI	Varies with phone functionality.
Calling Identity Delivery and Suppression (Delivery) [2]	CIDSD	Presentation status from the phone, and single-stage digit collection.
Calling Identity Delivery and Suppression (Suppression) [2]	CIDSS	Presentation status from the phone, and single-stage digit collection.
Calling Identity Delivery on Call Waiting	CIDCW	Functionality provided by the phone. Cisco BTS 10200 supports the interface.
Calling Name Delivery [3]	CNAM	Same as MGCP.
Calling Name Delivery Blocking	CNAB	Presentation status from the phone, and single-stage digit collection.

Table 2-1 MGCP Features and SIP Support (continued)

MGCP-Based Feature	Abbreviation	Support for SIP Phone Compared to Support for MGCP-Based Phone
Calling Number Delivery [3]	CND	The calling party number, if available, is delivered in the From header of the outgoing INVITE from the BTS 10200 to the terminating SIP phone. The number is delivered to the SIP phone even if the CND feature is not provisioned for the subscriber.
Calling Number Delivery Blocking	CNDB	Presentation status from the phone, and single-stage digit collection.
Cancel Call Waiting	CCW	Functionality provided by the phone. Cisco BTS 10200 supports the interface.
Class of Service	COS	CoS screening is supported for SIP subscribers. For account code and authorization code features on SIP endpoints, the BTS 10200 uses only the Interactive Voice Response (IVR)-based method of prompting, not the tone-based method. For account codes and authorization codes, the system applies IVR-based prompts for SIP endpoints, regardless of the values you provision for the PROMPT_METHOD parameter in the cos-restrict table. To provision these features, see the "Class of Service Screening" provisioning procedure in the *Provisioning Guide*.
Collection of Account and Authorization Codes	—	For a description of the account code and authorization code features, see the Class of Service (CoS) section in the *Cisco BTS 10200 Softswitch Network and Subscriber Feature Descriptions* document. For SIP endpoints, the system uses only the *Interactive Voice Response (IVR)-based* method of prompting, not the tone-based method. For account codes and authorization codes, the system applies IVR-based prompts for SIP endpoints, regardless of the values you provision for the PROMPT_METHOD parameter in the cos-restrict table. To provision these features, see the "Class of Service Screening" provisioning procedure in the *Cisco BTS 10200 Softswitch Provisioning Guide*. In addition, review the examples given in "Account Code Provisioning Example" section on page 2-23 and "Authorization Code Provisioning Example" section on page 2-24.
Custom-Dial-Plan	CDP	Same as MGCP.
Customer Originated Trace	COT	Same as MGCP.
Directed Call Pickup without Barge-in	DPN	Not supported.
Directed Call Pickup with Barge-in	DPU	Not supported.

Table 2-1 MGCP Features and SIP Support (continued)

MGCP-Based Feature	Abbreviation	Support for SIP Phone Compared to Support for MGCP-Based Phone
Distinctive Alerting Call Waiting Indication	DACWI	This feature is provided to Centrex users only. Provisioning for SIP does not differ from provisioning for MGCP. However, the delivery method for DACWI is different. The Centrex administrator provisions a list of DNs that are to receive DACWI tones. In MGCP, the phone plays the tone specified by the BTS 10200 in the protocol message. In SIP, the tone provisioned for the DN is specified by the BTS 10200 in the Alert-Info header of the INVITE as a file URL. A SIP phone, if capable, interprets this header and plays the specified distinctive ringing or call-waiting tone.
Distinctive Ringing Call Waiting	DRCW	Provisioning for SIP does not differ from provisioning for MGCP. However, the delivery method for DRCW is different. The subscriber provisions a list of DNs to receive DRCW tones. In MGCP, the phone plays the tone specified by the Cisco BTS 10200 in the protocol message. In SIP, the tone provisioned for the DN is specified by the Cisco BTS 10200 in the Alert-Info header of the INVITE as a file URL. A SIP phone, if capable, can interpret this header and play the specified distinctive ringing or call-waiting tone.
Distinctive Ringing Call Waiting	DRCW_ACT	Same as MGCP.
Do Not Disturb	DND	Same as MGCP, except that the reminder ring cannot be used with SIP devices. For additional information on DND, see the "Do Not Disturb" section on page 2-31.
Do Not Disturb Activation	DND_ACT	Same as MGCP.
Do Not Disturb Deactivation	DND_DEACT	Same as MGCP.
Group Speed Call—1 Digit	GSC1D	**Note** You can use the same command, **add sc1d**, to provision the 1-digit speed call DN for either individual or group speed call; similarly, use **add sc2d** to provision the 2-digit speed call DN for either individual or group. For a general description of the speed call and group speed call features, see Chapter 3, "Subscriber Features," in the *Cisco BTS 10200 Softswitch Network and Subscriber Feature Descriptions* document. For the general provisioning procedure see Chapter 7, "Features," in the *Cisco BTS 10200 Softswitch Provisioning Guide*.

Table 2-1 MGCP Features and SIP Support (continued)

MGCP-Based Feature	Abbreviation	Support for SIP Phone Compared to Support for MGCP-Based Phone
Group Speed Call—2 Digit	GSC2D	**Note** You can use the same command, **add sc1d**, to provision the 1-digit speed call DN for either individual or group speed call; similarly, use **add sc2d** to provision the 2-digit speed call DN for either individual or group. For a general description of the speed call and group speed call features, see Chapter 3, "Subscriber Features," in the *Cisco BTS 10200 Softswitch Network and Subscriber Feature Descriptions* document. For the general provisioning procedure see Chapter 7, "Features," in the *Cisco BTS 10200 Softswitch Provisioning Guide*.
Hotline	HOTLINE	Not supported.
Hotline Variable	HOTV	Not supported.
Hotline Variable Activation	HOTVA	Not supported.
Hotline Variable Deactivation	HOTVD	Not supported.
Hotline Variable Interrogation	HOTVI	Not supported.
Incoming Simulated Facility Group	ISFG	Same as MGCP.
Local Number Portability	LNP	Same as MGCP.
Multiline Hunt Group	MLHG	MLHG is supported for SIP subscribers. SIP subscriber provisioning is slightly different than MGCP and Network-based Call Signaling (NCS) subscriber provisioning. For the differences between the MGCP-based procedures and the SIP-based procedures see the "Multiline Hunt Group (MLHG)" section in the *Network and Subscriber Feature Descriptions* document. Some MLHG features are applied differently for SIP subscribers than for MGCP/NCS subscribers. • If a SIP subscriber is not registered, the system does not attempt to deliver the call to that subscriber. Instead, it searches for the next idle line. • If a SIP phone is capable of receiving multiple calls, and no other line in the MLHG is idle, the system can attempt to deliver the call to the busy SIP phone, depending on the provisioning in the subscriber table. For additional feature details, see the "Multiline Hunt Group (MLHG)" section in the *Network and Subscriber Feature Descriptions*.

Table 2-1 MGCP Features and SIP Support (continued)

MGCP-Based Feature	Abbreviation	Support for SIP Phone Compared to Support for MGCP-Based Phone
Multiple Directory Number	MDN	Provisioning for SIP does not differ from provisioning for MGCP. However, the delivery methods for distinctive-ringing (a distinctive ring tone for each line of the MDN subscriber), and the distinctive tone on call waiting are different. You provision distinctive ringing and call waiting tones for each DN of the MDN subscriber in the same manner for MGCP and SIP. In MGCP, the phone plays the tone specified by the Cisco BTS 10200 in the protocol message. In SIP, the tone provisioned for the DN is specified by the Cisco BTS 10200 in the Alert-Info header of the INVITE as a file URL. A SIP phone, if capable, can interpret this header and play the specified distinctive ringing or call waiting tone.
Outgoing Call Barring	OCB	Same as MGCP.
Outgoing Call Barring Activation	OCBA	Single stage digit collection.
Outgoing Call Barring Deactivation	OCBD	Single stage digit collection.
Outgoing Call Barring Interrogation	OCBI	Single stage digit collection.
Outgoing Simulated Facility Group	OSFG	Same as MGCP.
Remote Activation of Call Forwarding	RACF	Same as MGCP.
Remote Activation of Call Forwarding PIN	RACF_PIN	Same as MGCP.
Refer	REFER	This is not for MGCP users. Cisco BTS 10200 supports the SIP REFER interface to enable services such as Call-Transfer (attended, unattended) provided by the phone.
Selective Call Acceptance	SCA	Same as MGCP.
Selective Call Acceptance Activation	SCA_ACT	Same as MGCP.
Selective Call Forwarding	SCF	Same as MGCP.
Selective Call Forwarding Activation	SCF_ACT	Same as MGCP.
Selective Call Rejection	SCR	Same as MGCP.
Selective Call Rejection Activation	SCR_ACT	Same as MGCP.
Speed Call—1 Digit Speed Call—2 Digit	SC1D SC2D	Same as MGCP.

Table 2-1 MGCP Features and SIP Support (continued)

MGCP-Based Feature	Abbreviation	Support for SIP Phone Compared to Support for MGCP-Based Phone
Speed Call Activation—1 Digit Speed Call Activation—2 Digit	SC1D_ACT SC2D_ACT	Speed call activation is supported for SIP subscribers. For MGCP and NCS endpoints, subscriber data provisioning (speed call digits and DN) is performed by the end user on the handset. However, for SIP endpoints, handset provisioning is not supported; therefore the service provider should perform this provisioning through the **add sc1d** and **add sc2d** CLI commands. For the complete provisioning procedures, see the ”Features” chapter of the *Provisioning Guide*. The examples below show you how to do this. **`add sc1d sub-id=406-555-1805; dn2=654-555-1222;`** (Repeat as needed for DN3–DN9.) ⚠ **Caution** For Centrex groups, use only DN2–DN7. Otherwise there could be a conflict with other features that begin with dialing 8 or 9. However, if you have provisioned the multiline variety package (MVP) feature for the Centrex group, and you are *not* using digit 8 for extension dialing and *not* using digit 9 for PSTN access, you can use DN8 and DN9 for speed call. **`add sc2d sub-id=406-555-1806 dn20=654-555-1333;`** (Repeat as needed for DN21–DN49.)
Three-Way Calling	TWC	Functionality provided by the phone. The BTS 10200 supports the interface.
Three-Way Call Deluxe	TWCD	Varies with phone functionality.
Usage-Sensitive Three-Way Calling	USTWC	Functionality provided by the phone. The BTS 10200 supports the interface.
Warmline	WARMLINE	Not supported.

1. See additional information on call forwarding features in the “Call Forwarding” section on page 2-26.
2. See additional information on the delivery and suppression feature in the “Caller ID Delivery Suppression” section on page 2-29.
3. See additional information on calling name and calling number in the “Calling Name and Number Delivery” section on page 2-29.

Account Code Provisioning Example

In this example, note that ACCT_CODE_ALLOW is set to Y. This causes the system to prompt the caller for an account code.

```
add cos_restrict ID=Acct_Code; CASUAL_RESTRICT_TYPE=ALL_CICS_ALLOWED;
NATIONAL_RESTRICT_TYPE=ALL_NANP_CALLS; NATIONAL_WB_LIST=NONE;
INTL_RESTRICT_TYPE=ALL_CC_ALLOWED; II_RESTRICT=NONE BLOCK;_900=N; BLOCK_976=N; BLOCK_DA=N;
BLOCK_NANP_OPER_ASSIST=N;
BLOCK_INTL_OPER_ASSIST=N;
ACCT_CODE_ALLOW=Y;
ACCT_CODE_LENGTH=4;
AUTH_CODE_ALLOW=N;
BLOCK_INFO=N;
BLOCK_TW=N;
BLOCK_INTL=N;
NOD_WB_LIST=NONE;
```

```
PROMPT_METHOD=TONE; <<< For SIP endpoints, the system ignores this value.
ALLOW_CALLS_ON_IVR_FAILURE=N;

Change subscriber id=212-555-1206; cos_restrict_id=Acct_Code;
```

Authorization Code Provisioning Example

In this example, note that AUTH_CODE_ALLOW is set to Y. This causes the system to prompt the caller for an authorization code.

```
add cos_restrict ID=Auth_Code;
CASUAL_RESTRICT_TYPE=ALL_CICS_ALLOWED;
NATIONAL_RESTRICT_TYPE=LOCAL_ONLY;
NATIONAL_WB_LIST=NONE;
INTL_RESTRICT_TYPE=ALL_CC_ALLOWED;
II_RESTRICT=NONE;
BLOCK_900=N;
BLOCK_976=N;
BLOCK_DA=N;
BLOCK_NANP_OPER_ASSIST=N;
BLOCK_INTL_OPER_ASSIST=N;
ACCT_CODE_ALLOW=N;
AUTH_CODE_ALLOW=Y;
AUTH_CODE_LENGTH=23;
AUTH_CODE_GRP_ID=ivr23d;
BLOCK_INFO=N;
BLOCK_TW=N;
BLOCK_INTL=N;
NOD_WB_LIST=NONE;
PROMPT_METHOD=TONE; <<< For SIP endpoints, the system ignores this value.
ALLOW_CALLS_ON_IVR_FAILURE=N;
Change subscriber id=212-555-1207; cos_restrict_id=Auth_Code;
```

Cisco BTS 10200 Softswitch-Based Features

Softswitch-based features are directly provided by the BTS 10200. SIP phones can provide some features on their own; for information on the features provided by the different SIP phones, see the SIP phone administration guides.

This section describes Softswitch-based features entirely provided by the BTS 10200.

Note BTS 10200 announcements are customizable on a business group basis. If an announcement is not provisioned or cannot be played, a reorder tone is played.

Summary

Table 2-2 lists the most commonly used features; however, it is not an exhaustive list.

Note The sections that follow the table provide additional details on selected softswitch-based features.

s

Table 2-2 BTS 10200-Based SIP Features

SIP Feature	Acronym
Activation and Deactivation of Anonymous Call Rejection	ACR
Anonymous Call Rejection Activation	ACR_ACT
Anonymous Call Rejection Deactivation	ACR_DEACT
Call Forwarding	CF
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
Call Waiting Deluxe Activation	CWDA
Call Waiting Deluxe Deactivation	CWDD
Call Waiting Deluxe Interrogation	CWDI
Called Party Termination	CPT
Caller ID Suppression	CIDS
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
Calling Name and Number Delivery	CND
Customer Access Treatment	CAT
Customer-Originated Trace	COT
Differentiated Services Code Point	DSCP
Direct Inward Dialing	DID
Direct Outward Dialing	DOD
Do Not Disturb	DND
Do Not Disturb Activation	DND_ACT
Do Not Disturb Deactivation	DND_DEACT
Emergency Call	E911
E.164 and Centrex Dialing Plan (Extension Dialing)	E.164
Incoming and Outgoing Simulated Facility Group	ISFG and OSFG
Multiple Directory Numbers	MDN
Operator Services (0-, 0+, 01+, 00 calls)	—
Outgoing Call Barring	OCB

Table 2-2 BTS 10200-Based SIP Features (continued)

SIP Feature	Acronym
Outgoing Call Barring Activation	OCBA
Outgoing Call Barring Deactivation	OCBD
Outgoing Call Barring Interrogation	OCBI
Remote Activation of Call Forwarding	RACF
Vertical Service Codes	VSC

Call Forwarding

The differences between the feature for SIP and the feature for MGCP are as follows:

- There is no tone provided for SIP users to prompt for forwarding digits. The SIP users enter the forwarding digits immediately after the VSC. This is called single-stage dialing.
- There is no dial tone played after the SIP user successfully activates or deactivates the Forwarding features. The SIP user always hears an announcement (if announcements are provisioned) or a re-order tone.

Call Forwarding Activation and Deactivation

Activation and deactivation of call forwarding features use the vertical service code (VSC), also known as a star code.

With SIP support, the call forwarded to number can be a Centrex extension number (only applicable for business users) or an E.164 number.

Note Forwarding to a URL AOR is not supported.

SIP subscribers do not hear a final dial tone upon completing activation or deactivation. Instead, an announcement plays for the subscriber, indicating that the status of the forwarding feature is being activated or deactivated. This is irrespective of the Final Stage Dial Tone (FDT) flag (Y/N) provisioned for these features.

Call Forwarding to an E.164 Number or an Extension Number

Activation and deactivation are accomplished using single-stage dialing.

Detailed Provisioning Procedure and Feature Description

Additional information on this feature is provided at the following links:

- Call forwarding sections in the *Cisco BTS 10200 Softswitch Provisioning Guide*

- "Call Forwarding Features" section in the *Cisco BTS 10200 Softswitch Network and Subscriber Feature Descriptions*

Call Park and Directed Call Pickup Features

The feature behavior of SIP endpoints is similar to the behavior of MGCP and NCS-based endpoints, but there can be a difference. The difference in behavior is due to the SIP Media Terminal Adapter (MTA) or phone device used, but it does not affect the performance of Cisco BTS 10200 Softswitch.

Prerequisites for SIP Endpoints

In order to deliver the CPRK, CPRK-RET, DPN, and DPU features on the Cisco BTS 10200 and SIP endpoints, the following must be met:

- If a subscriber presses the # key and the SIP-based MTA does not send the extension number in a SIP message to the Cisco BTS 10200, the Cisco BTS 10200 does not park the call on the subscriber's extension. If the MTA sends the subscriber's extension number, the Cisco BTS 10200 parks the call on the parking extension (subscriber's extension) number.
- If the parking party tries to park a call on his or her own extension by hanging up, the end point puts the parking party's extension in the SIP message to Cisco BTS 10200.
- If the Cisco BTS 10200 sends a SIP error response to a SIP-based MTA on a request for call park, the MTW is responsible of playing a reorder tone.
- For all of the features listed in this section (CPRK, CPRK-RET, DPN, and DPU)
  - The SIP endpoint must support en-bloc signaling. En-bloc means that the endpoint delivers both the VSC and the dialed directory number (DN) in a single string.

Note This is different from the case of an MGCP/NCS endpoint. In that case the end user dials the VSC, the system returns a tone, and then the end user dials the DN.

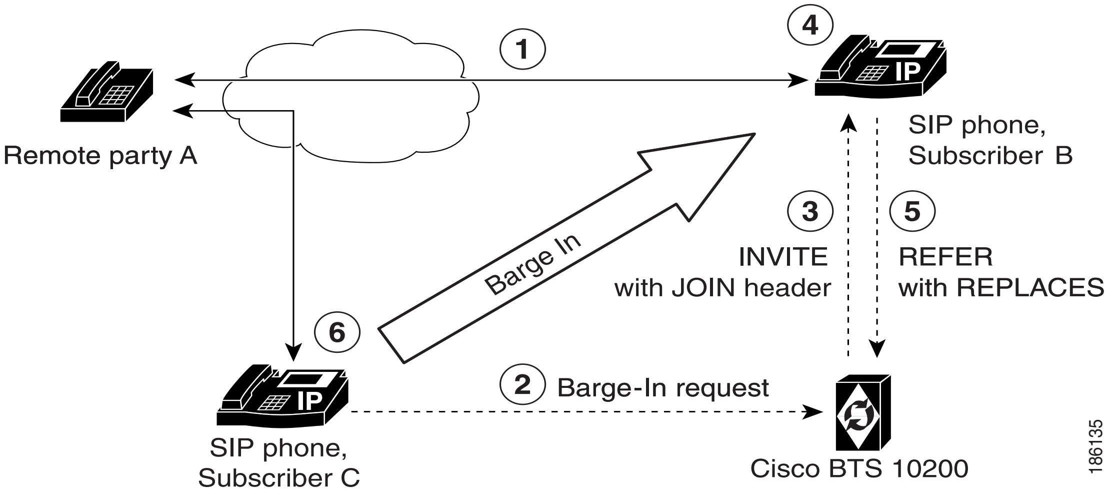
  - For these features to work for all of the subscribers in the group, you must provision the REFER feature for those subscribers.

If the end user performs an invalid action, for example, the end user dials an invalid extension after the VSC or dials a VSC for a feature that is not assigned the line, the Cisco BTS 10200 sends an error message to the SIP endpoint. It is the responsibility of the SIP endpoint to play the reorder tone to the end user.

- For CPRK
  - A REFER request with VSC must be supported on the SIP endpoint.
  - The SIP endpoint must be capable of playing a confirmation tone.
- For DPU, the following prerequisites apply to the barged-in-upon SIP endpoint (shown as Subscriber B in Figure 2-3):
  - For the barged-in-upon SIP endpoint to be able to accept the barge-in request, it must support the JOIN header (based on RFC 3911).
  - The SIP endpoint must be capable of playing a barge-in tone to the users to indicate the call has been barged in to. This tone must be played when the SIP endpoint receives an INVITE message with the JOIN header, and before the SIP endpoint sends the success response back to the Cisco BTS 10200.

- SIP MTA must be capable of handling REFER failure or REFER reject sent from the Cisco BTS 10200 and must be capable of sending a RE-INVITE to the Cisco BTS 10200 to connect the called and calling parties, prior to the call park attempt is made. For example, if a call is already parked on an extension and an attempt to park another call is made, the REFER request fails and the call made prior to the call park initiation is restored.
- The SIP endpoint must support the REFER message with REPLACES header. This header is required for establishing a two-way call between the other two parties when the endpoint hangs up during a three-way call. This process is illustrated in Figure 2-3.

Figure 2-3 Barge-In Process



Notes for Figure 2-3

1. Remote party A and Subscriber B are in a stable call.
2. Subscriber C wants to barge in to the call on the Subscriber B side and sends a barge-in request to the Cisco BTS 10200.
3. The Cisco BTS 10200 sends an INVITE message with a JOIN header to Subscriber B.
4. Subscriber B plays a barge-in tone to its own headset and then sets up a three-way call. (The ability to play the tone and the ability to set up the three-way call are both in the SIP phone.)
5. If Subscriber B hangs up, a REFER message with a REPLACES header to the Cisco BTS 10200 is sent.
6. The Cisco BTS 10200 sets up a two-way call between Subscriber A and Subscriber C, just as it would if this were an attended call transfer.

Note If Subscriber B does *not* hang up, but Subscriber A or Subscriber C hangs up, Subscriber B continues in a two-way call with the remaining party.

Limitation on JOIN Header

As shown in Figure 2-3, the Cisco BTS 10200 can send the JOIN header in an outbound Invite message. However, the Cisco BTS 10200 does not support the JOIN header on inbound messages.

Provisioning

You provision the CPRK, CPRK-RET, DPU, and DPN features for SIP endpoints as you would if you were provisioning for MGCP/NCS endpoints, as described in the "Feature Provisioning" chapter of the Cisco BTS 10200 Softswitch Provisioning Guide. However, you must also provision the REFER feature for all members of the group.

Calling Name and Number Delivery

Calling number delivery (CND) provides the SIP subscriber endpoint with the calling number of an incoming call. Calling name delivery (CNAM) provides the endpoint with the name of the calling party.

CND

The calling party number, if available, is delivered in the From header of the outgoing INVITE from the BTS 10200 to the terminating SIP phone. The number is delivered to the SIP phone even if the CND feature is not provisioned for the subscriber. The delivered information is as follows:

- If the calling number is available and the presentation indication is *not restricted*, the number is inserted into the user information portion of the From header.
- If the calling number is available and the presentation indication is *restricted*, the user information portion of the From header is set as "Anonymous."
- If the calling number is not available, the user information portion of the From header is left empty.

CNAM

The calling party name is delivered in the outgoing INVITE from the BTS 10200 to the terminating SIP phone only if the CNAM feature is provisioned for the SIP subscriber. The delivered information is as follows:

- If the calling number and name are available and the presentation indication of both the calling number and calling name are *not restricted*, the calling name is inserted into the display name field of the From header.
- If the calling number and name are available and the presentation indication of either calling number or calling name is *restricted*, the display name field of the From header is set as "Anonymous."
- If the calling name is not available, the display name field of the From header is left empty.

Additional information on this feature is provided at the following links:

- CND, CNAM, CNDB, and CNAB sections in the *Cisco BTS 10200 Softswitch Provisioning Guide*
- "Calling Identity Features" section in the *Cisco BTS 10200 Softswitch Network and Subscriber Feature Descriptions*

Caller ID Delivery Suppression

The treatment for caller's identity is based on the presence of "anonymous" in the Display-Name field of the From header in the INVITE message. If the caller's identity is restricted in the incoming SIP INVITE message, the presentation is suppressed.

Caller Identity presentation (allowed/restricted) information for SIP subscribers is not maintained in the the BTS 10200 database. This information is maintained on the individual phones and can be provisioned through the phone softkeys. Permanent restriction on the phone can be overridden if the caller dials a feature (*) code on a per-call basis. This is a single-stage dialing for SIP subscribers.

Additional information on this feature is provided at the following links:

- "CND, CNAM, CNDB, and CNAB" sections in the *Cisco BTS 10200 Softswitch Provisioning Guide*
- "Calling Identity Delivery and Suppression (CIDSD and CIDSS)" section in the *Cisco BTS 10200 Softswitch Network and Subscriber Feature Descriptions*

Customer Access Treatment

Provisioning this feature for SIP is the same as provisioning it for MGCP. The provisioning commands for this feature are shown in the "Centrex Group" section in the *Cisco BTS 10200 Softswitch Provisioning Guide*.

Direct Inward Dialing

Provisioning Direct Inward Dialing (DID) for SIP is the same as provisioning it for MGCP.

Assign the DID number to the subscriber as DN1 in the Subscriber table.

For information about the operation of this feature, see the "Direct Inward Dialing" section in the *Cisco BTS 10200 Softswitch Network and Subscriber Feature Descriptions*.

Direct Outward Dialing

With the Direct Outward Dialing (DOD) service, a station user can place external calls to the exchange network without attendant assistance by:

1. Dialing the DOD (public) access code (usually the digit 9)
2. Receiving a second dial tone
3. Dialing the external number (a number outside the customer group)

Access to the DOD feature is subject to station restrictions.

Note For IP phones, the second dial tone is provided by the phone itself. However, the prefix code is presented to the BTS 10200 along with the DDD number in the INVITE message. Secondary dial-tone capability is dependent on the SIP device used.

For information about the operation of this feature, see the "DOD for PBX" section in the *Cisco BTS 10200 Softswitch Network and Subscriber Feature Descriptions*.

Do Not Disturb

The Do Not Disturb (DND) feature enables a user to block incoming calls to the station on which the feature is activated. If no call forwarding features are activated, calls to the station are routed to busy treatment. This feature should be provisioned and activated on the BTS 10200 because of feature interaction with advanced features like executive override.

This is a single-stage dialing activation feature. The Alert-Info header plays the result of activation/deactivation—Success is a confirmation tone and failure is a failure message.

The reminder ring option (which is available with the DND feature on MGCP-based lines) cannot be used with SIP devices.

For features (such as DND) that can be fully provisioned on the BTS 10200 or on the phone, you can provision either one of the devices to enable the feature.

⚠ **Caution** Prior to provisioning your system, determine how you want to apply and configure features in your network to avoid conflicts between features provided by the BTS 10200 and features provided by the phones.

Additional information on this feature is provided at the following links:

- "Do Not Disturb (DND)" section in the *Cisco BTS 10200 Softswitch Provisioning Guide*
- "Do Not Disturb (DND)" section in the *Cisco BTS 10200 Softswitch Network and Subscriber Feature Descriptions* document

E.164 and Centrex Dialing Plan (Extension Dialing)

The system supports E.164 and Centrex Dialing Plan (extension dialing) addressing from SIP subscribers served by the local BTS 10200.

The SIP phone's dial plan must be configured so that it factors in the number of digits in the Centrex group. Centrex dialing can be provisioned within a range of 1 through 7 digits. Each Centrex group should have its own separate dial plan.

Note The CDP feature should be assigned to every Centrex category user.

A SIP URL with E.164 Addressing would look similar to the following example:

```
sip:4695550123@rcdn.cisco.com;user=phoneA sip:50603@rcdn.cisco.com;user=phone
```

Additional information on this feature is provided at the following links:

- "Provisioning a Centrex Group" section in the *Cisco BTS 10200 Softswitch Provisioning Guide*
- *Cisco BTS 10200 Softswitch Routing and Dial Plan Guide*
- "Numbering Plans and Dialing Procedures" section and "Features for Centrex Subscribers Only" sections in the *Cisco BTS 10200 Softswitch Network and Subscriber Feature Descriptions*

Operator Services (0-, 0+, 01+, and 00 Calls)

There is no Cisco BTS 10200 Softswitch subscriber-specific provisioning involved for Operator Services.

Additional information on this feature is provided in the "Operator Services" section in the *Cisco BTS 10200 Softswitch Network and Subscriber Feature Descriptions*.

User-Level Privacy

User-level privacy is provisioned in the Subscriber table.

Setting the privacy parameter to **user** directs the system to apply the user-provided privacy information. This setting (**privacy=user**) applies only to SIP endpoints that are capable of including privacy information.

Vertical Service Code Features

This section explains how to plan VSCs in a network with SIP subscribers, and lists the VSC-enabled features.

Planning VSCs In Networks with SIP Subscribers

Some features require SIP subscriber to enter a series of numbers and characters on the SIP client or handset. Typically, the subscriber dials VSC digits followed by additional dialing keys representing the parameters for the feature call. For MGCP subscribers, the BTS 10200 sends a response tone or announcement between the VSC code and the additional digits. However, for SIP endpoints, all the digits are dialed at a stretch without waiting for an intervening response tone from the BTS 10200. The following paragraph explains how certain combinations of VSC can cause mismatches between the feature the subscriber is attempting to manage versus the response of the BTS 10200, and how to plan VSCs to avoid these mismatches.

You should not deploy certain combinations of VSCs on networks with SIP endpoints. If you deploy a VSC longer than 2 digits, make sure that the longer VSC does not begin with the same sequence of characters as one of the shorter VSCs. In some cases, the system might match the shorter string even if the subscriber dialed the longer string. Consider the following example, for which the subscriber is expected to dial a VSC followed by a DN.

A SIP subscriber is provisioned with *93 for Feature1 and *938 for Feature2, and dials *938+2135551801 to invoke Feature2. The BTS 10200 receives *9382135551801 in the INVITE message. By default, it takes the first six characters, in this case *93821, and uses this string to look up the feature in the VSC table. There is no match for *93821, therefore the BTS 10200 proceeds as follows. First, it uses *9 to look for a match in the VSC table and it cannot be found. Then it uses *93, finds a match, and delivers Feature1. This is incorrect. The user's intention was to invoke Feature2 and not Feature1. The solution is for the service provider to change one of the two VSCs (either *93 or *938) in the VSC table.

Supported VSC-Enabled Features for SIP Endpoints

The following BTS 10200 Vertical Service Code (VSC) features are supported on SIP endpoints:

- CIDSS

- CIDSD
- CNAB
- OCBA, OCBD, and OCBI
- CFUA, CFUD, and CFUI

  Reminder ringback cannot be enabled for SIP subscribers. If you are turning on the CFU feature for a SIP subscriber, make sure that reminder ring capability is turned off. This should be done at a subscriber level.

  Here is the command format at the feature level:

  ```
  add feature fname=CFU; tdp1=TERMINATION_ATTEMPT_AUTHORIZED;
  tid1=TERMINATION_ATTEMPT_AUTHORIZED; feature_server_id=FSPTC235; ttype1=R;
  fname1=CFUA; fname2=CFUD; type1=MCF; value1=Y; type2=RR; value2=N;
  description=CFU MCF multiple call forwarding allowed, RR ring reminder not
  allowed;
  ```

  Here is the command format at the subscriber feature level:

  ```
  add subscriber-feature-data sub_id=sip_sub2; FNAME=CFU; type2=RR; VALUE2=N;
  ```

- CFNAVA, CFNAVD, and CFNAI
- CFBVA, CFBVD, and CFBI
- RACF_PIN

Voice Mail

The voice-mail (VM) feature on the BTS 10200 allows subscribers to retrieve waiting voice messages from a VM server. The BTS 10200 receives a message-waiting indication (MWI) from the VM server and forwards the MWI to the subscriber's handset. The subscriber can then retrieve messages from the server. The VM feature is available to individual subscribers and Centrex subscribers.

SIP trunks interconnecting the BTS 10200 to an external VM server must be provisioned as SIP VM trunks. To do that, you set the VM flag (voice-mail-trunk-grp) for these trunks in the Softswitch Trunk Group Profile (softsw-tg-profile) table. (See the "SIP Trunk to Voice-Mail Server" section on page 3-48.)

Note For a description of the basic VM feature, see the "Voice Mail and Voice Mail Always" section in the *Cisco BTS 10200 Softswitch Network and Subscriber Feature Descriptions*. For general VM provisioning details, see the "Voice Mail" section in the *Cisco BTS 10200 Softswitch Provisioning Guide*.

VM Actions

The following voice mail-related actions are supported in the BTS 10200:

- VM Deposit
- MWI Notification
- Retrieving VM
- Calling Back a Message Depositor

VM Deposit

There are two methods for depositing voice mail. In the first, the subscriber dials the pilot number for the VM server, and the call terminates on the voice-mail trunk. The VM system then collects the message for a target mailbox, using IVR prompts to guide the subscriber.

This method of depositing voice mail does not use any special BTS 10200 capabilities; it just requires that the VM SIP trunk is provisioned and the pilot number is added to the dial plan of the subscriber calling the VM system.

In the second (more common) method, the subscriber activates a call forwarding feature on the BTS 10200, such as CFNA, CFU, or CFB, and specifies the forwarding number as the pilot number of the VM server.

MWI Notification

When a SIP phone registers with the BTS 10200, the BTS 10200 sends an unsolicited SIP NOTIFY message to convey the MWI status to the phone. This occurs on every registration, including refreshes.

Whenever a change in VM status occurs for a subscriber (for example, when a VM message is deposited for the subscriber, or when all such messages have been retrieved), the VM server sends an update to the BTS 10200. If the subscriber is on a SIP phone, the BTS 10200 sends an unsolicited SIP Notify message to convey the MWI status to the phone. The number in the NOTIFY message Request URL (which is the assigned subscriber number) identifies the subscriber.

When the BTS 10200 is congested by a flood of registrations (which might occur, for example, when power is restored to a region after an outage), it can automatically suppress the MWI indication to the registering phones, so that registration throughput is not adversely affected.

The BTS 10200 implements draft-ietf-sipping-MWI-01.txt, *A Message Summary and Message Waiting Indication Event Package for the Session Initiation Protocol (SIP)*, with the following caveat: It supports receiving unsolicited NOTIFYs from a VM system; however, it does not support subscribing to these notifications. Further, the BTS 10200 does not support subscriptions for MWI. It sends unsolicited NOTIFYs for MWI to SIP subscribers. No subscription is expected from the SIP phones for the purpose of receiving this notification.

The notification of MWI by the BTS 10200 is enabled by default (VMWI=Y in the Subscriber table). You can disable it by setting VMWI=N.

Tip For MGCP subscribers, the BTS 10200 sends the MGCP RQNT message to turn on MWI on the analog phone. This activates the MWI indicator on the subscriber phone. The indicator can be visual (a lamp, an envelope, or another icon on a display) or it can be auditory, such as a stutter dial tone that is provided when the user next goes off-hook.

For information on setting the MWI and VMWI parameters in the Subscriber table, see the "Message Waiting Indicator (MWI)—Audible and Visual" section in the *Cisco BTS 10200 Softswitch Network and Feature Descriptions*.

Retrieving VM

To retrieve a VM message, subscribers dial the pilot number for the VM server. The BTS 10200 routes the call to the SIP trunk for VM, based on the provisioned dial plan for the subscriber and the route, destination, and trunk-group entries.

Once the VM message is retrieved, the VM server sends a NOTIFY message to the BTS 10200 to turn off the MWI indicator.

Calling Back a Message Depositor

When subscribers call into a VM server, this feature allows for calling back the person who left the voice-mail message. The feature requires that a Softswitch trunk for the VM server be provisioned in the Cisco BTS 10200 Softswitch with the relevant routes, destination, and dial plans in order to admit VM-originated calls into the BTS 10200.

VM Implementation for Centrex Subscribers

For calls received on SIP VM trunks from the VM server, a subscriber is provisioned and associated as the main sub-ID for each trunk. The subscriber information represents properties of a specific Centrex group and does not represent any particular subscriber. No AOR is provisioned for this subscriber. This information is used for call processing.

Note For general VM provisioning details, see "Voice Mail" in the *Cisco BTS 10200 Softswitch Provisioning Guide*.

VM Within a Single Centrex Group

The following examples show commands for provisioning Centrex VM. Before you perform the following steps, you must already have a Centrex group provisioned on your system. See the procedure in the "Centrex Group" section of the *Cisco BTS 10200 Softswitch Provisioning Guide*.

Step 1 Add the destination ID for the voice-mail main subscriber.

```
add destination dest-id=tb16-local; call-type=LOCAL; route-type=SUB;
```

Step 2 Add a dial plan profile and dial plan for a SIP trunk to the VM server.

```
add dial-plan-profile id=tb16;

add dial-plan id=tb16; digit-string=469-555; dest-id=tb16-local; min-digits=10;
max-digits=10
```

Step 3 Add the softswitch trunk group profile for voice mail.

```
add softsw-tg-profile id=VM_Profile; protocol-type=SIP; voice_mail_trunk_grp=Y;
```

Note As an option, you can provision the diversion-header-supp token in the softsw-tg-profile table to Y. This instructs the VM server to select the target inbox based on the original called number in the Diversion header of the SIP message.

Step 4 Add the SIP trunk group.

Note This SIP trunk group serves several purposes. It is used (1) by the subscriber to access the VM server, (2) by the BTS 10200 to forward incoming calls to the VM server, and (3) by the VM server to notify the BTS 10200 that a message is waiting for the subscriber.

```
add trunk-grp id=80032; softsw-tsap-addr=vm.domainname.com:5060; call-agent-id=CA146;
tg-type=softsw; tg-profile-id=VM_Profile; dial-plan-id=tb16
```

Step 5 Add a subscriber to the Centrex group to serve as the VM main subscriber.

```
add subscriber id=vmctxg1; CATEGORY=ctxg; BILLING-DN=469-555-0144; DN1=469-555-0144;
SUB-PROFILE-ID=Centrex_sp2; TERM-TYPE=TG; ctxg_id=ctxgsip1; tgn_id=80032;
```

Step 6 Link the VM main subscriber with the trunk group.

```
change trunk-grp; id=80032; main_sub_id=vmctxg1;
```

Step 7 Map the voice-mail Centrex extension to the VM main subscriber.

```
add ext2subscriber CTXG-ID=ctxgsip1; EXT=540144; CAT-CODE=1; SUB-ID=vmctxg1;
```

Step 8 If your VM server does not support FQDN hostnames, you must provision a serving-domain-name record in the BTS 10200 using the IP addresses resolved from the sia-xxxCAnnn.domain address. Otherwise, the VMWI status from SIP voice-mail platforms fails authentication with the BTS 10200. The details for this step are provided in Step 6 of the "SIP Trunk to Voice-Mail Server" section on page 3-48.

Provisioning Voice Mail Across Multiple Centrex Groups

A VM application server can provide VM service for Centrex subscribers from multiple Centrex groups on the BTS 10200. For the VM server to identify the subscriber and provide service configured for a Centrex group, the BTS 10200 must indicate the Centrex group with which the subscriber is associated.

When the BTS 10200 forwards a call from a Centrex extension to VM, the VM server identifies the Centrex group of the extension to deposit the message in the correct mailbox. Further, when the VM server sends a SIP NOTIFY message to indicate that messages are waiting for a Centrex subscriber on the BTS 10200, it must identify the Centrex group in the request URI of the NOTIFY message sent to the BTS 10200.

For any INVITE sent out a SIP trunk by the BTS 10200 to the VM server, a BTS 10200 proprietary SIP URL parameter bgid is added to the From, To, Diversion, and Request URIs, if the user part of those URLs contains a Centrex extension number format in the user information field. The bgid value is provisioned as the trunk-subgroup-type on the SIP trunk, and identifies the Centrex group.

An example of this parameter syntax follows:

```
INVITE sip:50001@vm.cisco.com:5060;user=phone;bgid=grpA SIP/2.0
From: <sip:50603@bts.cisco.com;user=phone;bgid=grpA>;tag=1_1146_f40077_3jwv
To: <sip:50586@bts.cisco.com;user=phone;bgid=grpA>
Diversion: <sip:50586@bts.cisco.com;bgid=grpA>;reason=unconditional;counter=1
```

When the VM server notifies the BTS 10200 of a MWI for a Centrex subscriber, the VM server sends a Notify SIP request to the BTS 10200 with a Centrex number format in the Request URL, and an associated bgid parameter identifying the Centrex group associated with the subscriber. When the VM server initiates a call to a BTS 10200 Centrex subscriber for VM callback functionality, bgid is added to the request URL of the initial INVITE originating from the VM server. This identifies the Centrex group associated with the subscriber.

The BGID parameter in the REQURI of an INVITE originated from the VM server identifies the called subscriber in the targeted Centrex group. For example, the BGID parameter in the REQURI of a NOTIFY message from the VM server to the BTS 10200 identifies the subscriber in the targeted Centrex group whose MWI lamp is turned on or off.

The BTS 10200 does not support extension-dialed calls from one Centrex group to another. Therefore, the bgid parameter has an identical value if it is present in any of the URLs in the From, To, Diversion, and Request URL headers for a given INVITE message. The trunk group configuration includes a trunk subgroup field for specifying the bgid parameter value. One trunk group is provisioned for each Centrex group; the bgid parameter in the trunk group table is unique to the specific Centrex group. Routing tables are configured so that each trunk handles SIP calls to and from the VM server for a specific Centrex group. To qualify a specific trunk for bgid and VM, provision as follows:

- In the Trunk Group (trunk-grp) table, provision the bgid value in the trunk-sub-grp field.
- In the softsw-tg-profile table:
  - Provision the trunk-sub-grp-type field as BGID.
  - Provision the voice-mail-trunk-grp field as Y.

The following provisioning steps illustrate how to provide VM service for BTS 10200 Centrex subscribers across multiple Centrex groups.

Step 1 Add a SIP trunk profile for voice-mail trunks. Qualify voice-mail trunks by setting the voice-mail flag, and set the trunk sub-group type to indicate use of business group identifier:

```
add softsw_tg_profile ID=<profile_id>; PROTOCOL_TYPE=SIP; VOICE_MAIL_TRUNK_GRP=Y;
TRUNK_SUB_GRP_TYPE=BGID;
```

Step 2 Add a SIP trunk for each business group identifier. Each trunk points to the address of the voice-mail sever.

In the following command, be sure to enter a unique business group identifier for each Centrex group, for example, **bg1**, **bg2**, and **bg3**, for the three Centrex groups in this example.

Also, be sure to specify the FQDN and port that the VM server uses for SIP message exchange, for example, **vmserver:5060**.

```
add trunk_grp ID=<trk_grp_id1>; TG_TYPE=SOFTSW; TG_PROFILE_ID=<profile_id>;
SOFTSW_TSAP_ADDR=vmserver:5060; DIAL_PLAN_ID=dp; TRUNK_SUB_GRP=bg1;

add trunk_grp ID=<trk_grp_id2>; TG_TYPE=SOFTSW; TG_PROFILE_ID=<profile_id>;
SOFTSW_TSAP_ADDR=vmserver:5060; DIAL_PLAN_ID=dp; TRUNK_SUB_GRP=bg2;

add trunk_grp ID=<trk_grp_id3>; TG_TYPE=SOFTSW; TG_PROFILE_ID=<profile_id>;
SOFTSW_TSAP_ADDR=vmserver:5060; DIAL_PLAN_ID=dp; TRUNK_SUB_GRP=bg3;
```

Step 3 Create a dial plan for calls received on the SIP trunks, so that they can be routed based on the called party number. For example, the identifier for the dial plan in this example is **dp**. (The dial plan provisioning details are not shown here.) The Centrex group routing and dial plan tables should be provisioned so that calls originating from a specific Centrex group subscriber are sent out the SIP trunk with the business group identifier representing that Centrex group.

Jointly Provided Features

Some features are provided jointly by the phone and by the BTS 10200. Here are some examples:

- Call Transfer (Blind and Attended) with REFER
- Distinctive Ringing
- Distinctive Ringing for Centrex DID Calls

The sections that follow provide information about these features.

Call Transfer (Blind and Attended) with REFER

The SIP Call Transfer (CT) feature is supported for SIP subscribers. For SIP phones, this feature is provided as part of REFER support on the BTS 10200.

The CT feature requires phone support for sending the SIP REFER message. See the phone documentation for details on the user interface and procedures for effecting a call transfer. Both blind and attended transfers are supported. Attended transfer to a transfer-target is supported only after the target answers; that is, consultative attended transfer is supported. Attended transfer is not possible while the transfer-target is being alerted (ringing state).

The difference between provisioning the feature for SIP and provisioning it for MGCP is as follows:

- Call transfer on both the Cisco IP Phone 7905/7912 and the Cisco IP Phone 7940/7960 is done using softkeys. On the Cisco ATA 186/188, call transfer is done using the Flash key (or by pressing the on-hook button briefly) on the analog phone attached to the Cisco ATA 186/188.
- Call-transfer functionality for SIP-based systems is performed using the REFER feature, *not* the traditional CT feature. To enable CT for SIP subscribers, you must provision the REFER feature as an office trigger in the BTS 10200. See the "SIP Call Transfer with REFER and SIP INVITE with Replaces" section on page 3-43 for additional details and provisioning procedures.

Distinctive Ringing

Distinctive ringing uses a special ringing pattern to alert the called user of incoming calls from preselected telephone numbers. This is a CLASS feature and is offered to both business and residential users. There is no difference between provisioning the feature for SIP and provisioning it for MGCP.

You can edit the list of selected numbers though the Screening List Editing (SLE) feature, which requires the configuring of an IVR with the BTS 10200. Distinctive ringing can be assigned to a station and to the group, and it can be applied to users based on the call type/calling number. When assigned to a group, distinctive ringing is applied to users in the group based on the call type. When assigned to the line, distinctive ringing is applied to the user based on the calling number. The BTS 10200 sends an Alert-Info header in the outgoing INVITE message, instructing the SIP phone to play a specific ring tone.

Distinctive ringing depends on the SIP phone's capability to support processing of the information received in an Alert-Info header.

Distinctive Ringing for Centrex DID Calls

The BTS 10200 sends an Alert-Info header in the outgoing INVITE message, instructing the SIP phone to play a specific ring tone. Distinctive ringing depends on the SIP phone's capability to process the information received in the Alert-Info header. There are no differences between provisioning the feature for SIP and provisioning it for MGCP.

Phone-Based Features

The phone provides some features standalone, without BTS 10200 support. If the SIP phone requires provisioning to provide this function, refer to the SIP phone documentation for instructions.

Table 2-3 lists the phone-based features.

Table 2-3 SIP Phone-Based Features

Feature	Acronym
Call Hold and Resume	CHD
Call Waiting	CW
Call Waiting Caller ID	CWCID
Cancel Call Waiting	CCW
CODEC Up-Speeding	CODEC[1]
Do Not Disturb	DND
Three-Way Calling	TWC

[1] For feature calls between MGCP and SIP subscribers, the BTS 10200 supports the CODEC up-speeding capability. The SIP phone would also need to support this capability for the up-speeding capability to be fully supported in the call.

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

⚠ **Caution** Prior to provisioning your system, determine how you want to apply and configure features in your network to avoid conflicts between features provided by the BTS 10200 and features provided by the phones.