CHAPTER 3

Telephony Features

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This chapter explains how the features delivered to BTS 10200 SIP subscribers compare to those delivered to BTS 10200 MGCP subscribers. The chapter covers the following topics:

SIP-Based Features vs. MGCP-Based Features

Table 3-1 lists the MGCP features available (in the Feature column) and then describes how the feature differs when it is used as a SIP feature.

Table 3-1 MGCP Features and SIP Support

MGCP-Based Feature	Abbreviation	Support for SIP Phone Endpoint Compared to Support for MGCP-Based Endpoint
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.
Anonymous Call Rejection	ACR	Same as MGCP, when provided by Cisco 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 through the Services key on some SIP phones. Depending on the specific phone, the feature on the BTS 10200 might work jointly with the feature on the phone.

Table 3-1 MGCP Features and SIP Support (continued)

MGCP-Based Feature	Abbreviation	Support for SIP Phone Endpoint Compared to Support for MGCP-Based Endpoint
Anonymous Call Rejection Deactivation	ACR_DEACT	BTS 10200 functionality is same for SIP subscribers as for MGCP. ACR_DEACT is also supported through the Services key 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.
Call Block	CBLK	Same as MGCP.
Call Forward Busy	CFB	Same as MGCP.
Call Forward Busy Variable Activation	CFBVA	Single-stage digit collection, and also through the phone’s Services key.
Call Forward Busy Variable Deactivation	CFBVD	Same as MGCP, and also available through the phone’s Services key.
Call Forward Busy Interrogation	CFBI	Single-stage digit collection.
Call Forward Combined	CFC	Same as MGCP.
Call Forward Combined Activation	CFC_ACT	Single-stage digit collection, and also through the phone’s Services key.
Call Forward Combined Deactivation	CFC_DEACT	Same as MGCP, and also available through the phone’s Services key.
Call Forward No Answer	CFNA	Same as MGCP.
Call Forward No Answer Variable Deactivation	CFNAVA	Single-stage digit collection, and also through the phone’s Services key.

Table 3-1 MGCP Features and SIP Support (continued)

MGCP-Based Feature	Abbreviation	Support for SIP Phone Endpoint Compared to Support for MGCP-Based Endpoint
Call Forward No Answer Variable Deactivation	CFNAVD	Same as MGCP, and also available through the phone's Services key.
Call Forward No Answer Interrogation	CFNAI	Single-stage digit collection.
Call Forward Unconditional	CFU	Same as MGCP.
Call Forward Unconditional Activation	CFUA	Single-stage digit collection, and also through the phone's Services key.
Call Forward Unconditional Deactivation	CFUD	Same as MGCP, and also available through the phone's Services key.
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 REFER feature for more 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)	CIDSD	Presentation status from the phone, and single-stage digit collection.
Calling Identity Delivery and Suppression (Suppression)	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	CNAM	Same as MGCP.
Calling Name Delivery Blocking	CNAB	Presentation status from the phone, and single-stage digit collection.
Calling Number Delivery	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.

Table 3-1 MGCP Features and SIP Support (continued)

MGCP-Based Feature	Abbreviation	Support for SIP Phone Endpoint Compared to Support for MGCP-Based Endpoint
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 supported, without Auth/Account code collection.
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.
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.
Do Not Disturb Activation	DND_ACT	Same as MGCP, and also available through the phone’s Services key.

Table 3-1 MGCP Features and SIP Support (continued)

MGCP-Based Feature	Abbreviation	Support for SIP Phone Endpoint Compared to Support for MGCP-Based Endpoint
Do Not Disturb Deactivation	DND_DEACT	Same as MGCP, and also available through the phone's Services key.
Group Speed Call—1 digit	GSC1D	Not supported.
Group Speed Call—2 digit	GSC2D	Not supported.
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 not supported for SIP subscribers.
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, and also through the phone's Services key.
Outgoing Call Barring Deactivation	OCBD	Single stage digit collection, and also through the phone's Services key.
Outgoing Call Barring Interrogation	OCBI	Single stage digit collection, and also through the phone's Services key.
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.

Table 3-1 MGCP Features and SIP Support (continued)

MGCP-Based Feature	Abbreviation	Support for SIP Phone Endpoint Compared to Support for MGCP-Based Endpoint
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	SC1D	Not supported.
Speed Call Activation—1 digit	SC1D_ACT	Not supported.
Speed Call—2 digit	SC2D	Not supported.
Speed Call Activation—2 digit	SC2D_ACT	Not supported.
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.

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 Cisco BTS 10200 Softswitch.

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 3-2 lists the most commonly used Softswitch-based features; however, it is not an exhaustive list.

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Table 3-2 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 3-2 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

Activation and Deactivation of Anonymous Call Rejection

The provisioning steps and behavior of this feature are the same as provided for MGCP subscribers. Additional information on this feature is provided at the following links.

Provisioning Procedure	Feature Behavior (Feature Description and Handset Provisioning)
"Anonymous Call Rejection" section in the *Cisco BTS 10200 Softswitch Provisioning Guide*	"Anonymous Call Rejection" section in the *Cisco BTS 10200 Softswitch Network and Subscriber Feature Descriptions* document

Billing

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

CALEA

For information on CALEA, see Chapter 2, "Lawful Intercept and CALEA," in the *Cisco BTS 10200 Softswitch Network and Subscriber Feature Descriptions* document.

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. Alternatively, you can activate or deactivate the feature by using the Services key on certain phones.

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 address of record (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.

Provisioning Procedure	Feature Behavior (Feature Description and Handset Provisioning)
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* document

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.

Provisioning Procedure	Feature Behavior (Feature Description and Handset Provisioning)
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* document

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 Cisco BTS 10200 Softswitch 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.

Provisioning Procedure	Feature Behavior (Feature Description and Handset Provisioning)
"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* document

Called Party Termination

Provisioning or using this feature for SIP is the same as provisioning or using it for MGCP. This is true when the called-party termination is not available, not reachable, or not registered.

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 Chapter 8, "Centrex Provisioning," in the *Cisco BTS 10200 Softswitch Provisioning Guide*.

Customer-Originated Trace

Provisioning this feature for SIP is the same as provisioning it for MGCP.

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

Provisioning Procedure	Feature Behavior (Feature Description and Handset Provisioning)
"Customer Originated Trace" section in the *Cisco BTS 10200 Softswitch Provisioning Guide*	"Customer Originated Trace (COT)" section in the *Cisco BTS 10200 Softswitch Network and Subscriber Feature Descriptions* document

Direct Inward Dialing

Provisioning this feature 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* guide.

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* guide.

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 Do not attempt to provision the feature on both the switch and the phone, because this can cause conflicts.

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

Provisioning Procedure	Feature Behavior (Feature Description and Handset Provisioning)
"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

Emergency Call

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.

Note Only E911 (without the suspend procedure for 45 minutes) support. Basic 911 with suspend procedure is not supported.

Note The Public Safety Answering Point (PSAP) is selected based on default user location. No mobility is supported.

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

Provisioning Procedure	Feature Behavior (Feature Description and Handset Provisioning)
"911 Emergency" section in the *Cisco BTS 10200 Softswitch Provisioning Guide*	"Emergency Services (911)" 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 considers 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.

Example 3-1 A SIP URL with E.164 Addressing

```
sip:4695551234@rcdn.cisco.com;user=phoneA sip:50603@rcdn.cisco.com;user=phone
```

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

Provisioning Procedure	Feature Behavior (Feature Description and Handset Provisioning)
"Provisioning a Centrex Group" section in the *Cisco BTS 10200 Softswitch Provisioning Guide* *Cisco BTS 10200 Softswitch Dial Plan document*	"Numbering Plans and Dialing Procedures" section in the *Cisco BTS 10200 Softswitch Network and Subscriber Feature Descriptions* document "Features for Centrex Subscribers Only" section in the *Cisco BTS 10200 Softswitch Network and Subscriber Feature Descriptions* document

Incoming and Outgoing Simulated Facility Group

Provisioning this feature for SIP is the same as provisioning it for MGCP.

For information on provisioning simulated facility group, see the "Incoming Simulated Facility Group" and "Outgoing Simulated Facility Group" sections in the *Cisco BTS 10200 Softswitch Provisioning Guide*.

Multiple Directory Numbers

Provisioning this feature for SIP is the same as provisioning it 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 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.

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

Provisioning Procedure	Feature Behavior (Feature Description and Handset Provisioning)
"Multiple Directory Number (MDN)" section in the *Cisco BTS 10200 Softswitch Provisioning Guide*	"Multiple Directory Number (MDN)" section in the *Cisco BTS 10200 Softswitch Network and Subscriber Feature Descriptions* document

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

There is no Cisco BTS 10200 Softswitch subscriber-specific provisioning involved for Operator Services. There are provisioning procedures for busy line verification (BLV). Provisioning this feature for SIP is the same as provisioning it for MGCP.

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

Provisioning Procedure	Feature Behavior (Feature Description and Handset Provisioning)
"Busy Line Verification (BLV) " section in the *Cisco BTS 10200 Softswitch Provisioning Guide*	"Operator Services" section in the *Cisco BTS 10200 Softswitch Network and Subscriber Feature Descriptions* document

Outgoing Call Barring

Provisioning this feature for SIP is the same as provisioning it for MGCP.

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

Provisioning Procedure	Feature Behavior (Feature Description and Handset Provisioning)
"Outgoing Call Barring (OCB)" section in the *Cisco BTS 10200 Softswitch Provisioning Guide*	"Outgoing Call Barring (OCB)" section in the *Cisco BTS 10200 Softswitch Network and Subscriber Feature Descriptions* document

Remote Activation of Call Forwarding

Provisioning this feature for SIP is the same as provisioning it for MGCP.

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

Provisioning Procedure	Feature Behavior (Feature Description and Handset Provisioning)
"Remote Activation of Call Forwarding (RACF) and PIN Change" section in the *Cisco BTS 10200 Softswitch Provisioning Guide*	"Remote Activation of Call Forwarding (RACF)" section in the *Cisco BTS 10200 Softswitch Network and Subscriber Feature Descriptions* document

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 vertical service codes (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:

- Calling identity delivery and suppression, suppression part (CIDSS)
- Calling identity delivery and suppression, delivery part (CIDSD)
- Calling name delivery blocking (CNAB)
- Outgoing call bearing activation (OCBA), outgoing call bearing deactivation (OCBD), outgoing call bearing interrogation (OCBI)
- Call forwarding unconditional activation (CFUA), call forwarding unconditional deactivation (CFUD), call forwarding unconditional interrogation (CFUI)

  Reminder ringback cannot be enabled for SIP subscribers. If you are turning on the Call Forward Unconditional (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;
  ```

  And at the subscriber feature level:

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

- 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 on busy variable activation (CFBVA), call forwarding on busy variable deactivation (CFBVD), call forwarding on busy variable interrogation (CFBI)
- RACF Pin Change

Voice-Mail Support

Voice-mail support is configured through SIP trunks in the Cisco BTS 10200 Softswitch. The system also notifies SIP subscribers if a voice mail is waiting. For voice-mail provisioning and operation, see Chapter 5, "Voice-Mail Features."

Jointly Provided Features

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

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

The sections that follow provide information about these features.

Text-GUI Features

The BTS 10200 supports SIP client/handset text-based user interface (UI) provisioning for a select set of features. This is in addition to numerous supplementary features supported natively by the SIP client/handset itself. Some of the features require updating the status on the network database. Cisco BTS 10200 provides complementary support to SIP clients/handsets to update end user feature access status on the switch network database.

Provisioning in this context refers to feature activation or deactivation, and setting any applicable directory numbers (DNs) associated with the feature. If a SIP handset is used, the phone's LCD panel is used as a menu display area to guide the user toward feature provisioning. If a SIP software client is used, the UI display region in the client software is used to guide the user through feature provisioning.

There might be multiple lines on the SIP phone, but currently services configured by softkeys on the phone are available to only one of those lines. The subscriber for that line is provisioned in the BTS 10200 with the MAC address of the phone (see the MAC Address to Subscriber table in the Cisco BTS 10200 Softswitch CLI Database).

Supported Handsets

Cisco BTS 10200 supports any SIP client/handset that supports CallManager XML 3.0.

Supported Features

The following features have SIP client/handset-based provisioning support:

- Call Forwarding Unconditional Activation (CFUA), Call Forwarding Unconditional Deactivation (CFUD)
- Call Forwarding on Busy Variable Activation (CFBVA), Call Forwarding on Busy Variable Deactivation (CFBVD)
- Call Forwarding on No Answer variable Activation (CFNAVA), Call Forwarding on No Answer Variable Deactivation (CFNAVD)
- Do Not Disturb Activation (DND-ACT), Do Not Disturb Deactivation (DND-DEACT)
- Anonymous Call Rejection Activation (ACR-ACT), Anonymous Call Rejection Deactivation (ACR-DEACT)

Accessing Features

The following sections describe how to access the features.

SIP Handset

The SIP handset provides a button labeled "Services" or an icon indicating "Services." Initial access to feature provisioning is through the Services button. After initial access, the UI display area provides a menu-driven interface and a feature-specific menu.

To navigate the menu, the end user presses the Up and Down arrow buttons or menu numbers. At any level of navigation, the end user presses the Exit softkey to go back one step in the menu hierarchy. The user selects menu items using the Select softkey button and uses the numeric dial to enter DN information.

Menu Hierarchy

Feature Options
- Call Forwarding
  - Call-Fwd Busy
  - Activate/Deactivate Feature
  - Set/Change Forwarding Number
  - Number:
  - Call-Fwd Unconditional
  - Activate/Deactivate Feature
  - Set/Change Forwarding Number
  - Number:
  - Call-Fwd No Answer
  - Activate/Deactivate Feature
  - Set/Change Forwarding Number
  - Number:
- Anonymous Call Rejection
  - Activate/Deactivate Feature
- Do Not Disturb
  - Activate/Deactivate Feature

SIP Software Clients

The user interface for applicable software clients is similar to a SIP handset user interface.

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 call transfer (CT) feature*. To enable CT for SIP subscribers, you must provision the REFER feature as an office trigger in the Cisco BTS 10200 Softswitch. See the "SIP Call Transfer with REFER and SIP INVITE with Replaces" section on page 4-42 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 3-3 lists the phone-based features.

Table 3-3 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.