# Cisco BTS 10200 Softswitch SIP Protocol Provisioning Guide

Release 4.5.x
Revised October 21. 2008

**Americas Headquarters**
Cisco Systems, Inc.
170 West Tasman Drive
San Jose, CA 95134-1706
USA
http://www.cisco.com
Tel: 408 526-4000
800 553-NETS (6387)
Fax: 408 527-0883

Text Part Number: OL-5351-10

# CONTENTS

# Preface

**Revised: October 21, 2008, OL-5351-10**

The *Cisco BTS 10200 Softswitch SIP Protocol Provisioning Guide* details support for Session Initiation Protocol (SIP) devices and trunks in Release 4.5.x. This guide serves as a basic guidance for provisioning SIP devices (Cisco ATA 186 and 188, and Cisco IP Phones 7940, 7960, 7905, 7912) and SIP trunks for use with the Cisco BTS 10200. The book is an addition to the existing Cisco BTS 10200 Softswitch documentation.

## Revision History for Release 4.5.x

This document includes all of the information that was contained in the previous issues (the Release 4.4 SIP Protocol Provisioning Guide), and has been updated as follows for Release 4.5.x.

| Release | Changes |
|---|---|
| Release 4.5.x | **Version OL-5351-10:**<br>Updated the SIP Timer Values procedure in Chapter 1<br><br>**Version OL-5351-09:**<br>The procedure for installing the Mini-Browser Adapter (MBA) was clarified ("Configuring the HTTP-FS, MBA, GUI-FS, and SIP Phone Services" section on page 1-9).<br><br>**Version OL-5351-08:**<br>Clarification was added that provisioning changes to DSCP parameters in the ca-config table do not take effect until the CA platform restarts or switches over (Chapter 1, "Provisioning SIP Devices and SIP Subscribers" and Chapter 2, "Provisioning SIP Trunks").<br><br>**Version OL-5351-07:**<br>• (Release 4.5.0 and 4.5.1) Added information about the privacy parameter in the **add subscriber** command in Chapter 1, "Provisioning SIP Devices and SIP Subscribers."<br>• (Release 4.5.0 and 4.5.1) Added information about 3XX call redirection in Chapter 2, "Provisioning SIP Trunks."<br>• (Release 4.5.1, Maintenance Release 1) Added a section about source IP validation in Chapter 2, "Provisioning SIP Trunks."<br>• (Release 4.5.0 and 4.5.1) Added a step for provisioning authentication for access to the VM server (using server-domain-name parameter) in Chapter 3, "Provisioning Voice Mail." |

| Release | Changes |
|---|---|
| Release 4.5.x | **Version OL-5351-06 and earlier:**<br>Added the following new features:<br>• "Provisioning Secure FQDN of a SIP Endpoint" section on page 1-18<br>• "Session Timers" section on page 1-22<br>• "T.38 Fax Relay Call Agent Controlled Mode Across SIP Trunk Interface" section on page 2-14<br>Provided additional information on the following features:<br>• In Chapter 1, "Provisioning SIP Devices and SIP Subscribers," added LINKSYS PAP2.<br>• "Installing the MBA" section on page 1-10 (clarified installation requirements)<br>• "Provisioning a SIP Subscriber" section on page 1-12 (corrected the command for changing aor2sub status)<br>• "Provisioning Subscriber Features" section on page 1-13—Do Not Disturb, Provisioning Secure FQDN of a SIP Endpoint, SIM Memory Audit and SIP Dynamic Memory Audit, and Type of Service.<br>• "Jointly-Provided Features" section on page 1-22 (revised information on timers)<br>• "Provisioning SIP Trunk Features" section on page 2-2—Locating SIP Servers Using DNS Queries, Trunk Group Audit for the SIP Adapter, Diversion Indication, Carrier Identification Code over SIP, SIP Trunk Sub-Groups, SIP Timer Values, SIP-T, ISUP Version, ISUP-Transparency, and GTD, DTMF SIP Signaling, Asserted Identity and User-Level Privacy, Trunk Group Audit for the SIP Adapter<br>• Chapter 3, "Provisioning Voice Mail" |

# Organization

This document is organized as follows:

- Chapter 1, "Provisioning SIP Devices and SIP Subscribers"
- Chapter 2, "Provisioning SIP Trunks"
- Chapter 3, "Provisioning Voice Mail"
- Appendix A, "Sample Configuration Files for SIP Phones"
- Glossary

# Conventions

This document includes the following conventions:

**Note** Means *reader take note*. Notes contain helpful suggestions or references to material not covered in the manual.

# Related Documents

This Release 4.5.x SIP Protocol Provisioning Guide should be used in conjunction with the previous Cisco BTS 10200 Softswitch documents:

- *Cisco BTS 10200 Softswitch Release Notes for Release 4.5.x*
- *Cisco BTS 10200 Softswitch SIP Protocol User Guide*
- *Cisco BTS 10200 Softswitch Physical and Network Site Surveys and Data Sheets*
- *Cisco BTS 10200 Softswitch Cabling Procedures*
- *Cisco BTS 10200 Softswitch Network and Subscriber Feature Descriptions*
- *Cisco BTS 10200 Softswitch System Description*
- *Cisco BTS 10200 Softswitch Application Installation Procedures*
- *Cisco BTS 10200 Softswitch Operations Manual*
- *Cisco BTS 10200 Softswitch Event Messages Guide*
- *Cisco BTS 10200 Softswitch Billing Interface Guide*
- *Cisco BTS 10200 Softswitch Command Line Interface Reference Guide*
- *Cisco BTS 10200 Softswitch CORBA Installation and Programmer's Guides*

# Obtaining Documentation, Obtaining Support, and Security Guidelines

For information on obtaining documentation, obtaining support, providing documentation feedback, security guidelines, and also recommended aliases and general Cisco documents, see the monthly *What's New in Cisco Product Documentation*, which also lists all new and revised Cisco technical documentation, at:

http://www.cisco.com/en/US/docs/general/whatsnew/whatsnew.html

CHAPTER 1

# Provisioning SIP Devices and SIP Subscribers

**Revised: October 21, 2008, OL-5351-10**

The purpose of this chapter is to serve as a basic guidance for configuring Cisco SIP devices, including:

- Cisco ATA 186/188
- Cisco IP Phone 7905
- Cisco IP Phone 7912
- Cisco IP Phone 7940
- Cisco IP Phone 7960
- Cisco LINKSYS Phone Adapter PAP2

The chapter also demonstrates how to provision SIP subscribers on Cisco devices in to the Cisco BTS 10200 Softswitch, and provides guidance on provisioning and enabling features for SIP subscribers in Cisco BTS 10200.

You can find the detailed step-by-step administration guide for the Cisco ATA 186/188 adaptors at:

http://www.cisco.com/univercd/cc/td/doc/product/voice/ata/ataadmn/index.htm

You can find the detailed step-by-step administration guide for the Cisco 7905/7912 phones at:

http://www.cisco.com/univercd/cc/td/doc/product/voice/c_ipphon/english/ipp7905g/addprot/index.htm

You can find the detailed step-by-step administration guide for the Cisco 7940/7960 phones at:

http://www.cisco.com/univercd/cc/td/doc/product/voice/c_ipphon/sip7960/sadmin31/index.htm

For multiple line SIP phones, each line must be provisioned with a DN/Subscriber entry in the Cisco BTS 10200 Softswitch.

For information on the supported SIP protocol features, refer to the *Cisco BTS 10200 Softswitch Release 4.5.x SIP Protocol User Guide*.

# Provisioning SIP Devices

Cisco IP phones are full-featured telephones that can be plugged directly into an IP network and can be used very much like a standard private branch exchange (PBX) telephone. The Cisco SIP IP phone is an IP telephony instrument that can be used in VoIP networks.

The Cisco IP phone model terminals can attach to the existing data network infrastructure, via 10BASE-T/100BASE-T interfaces on an Ethernet switch. When used with a voice-capable Ethernet switch (one that understands type of service [ToS] bits and can prioritize VoIP traffic), the phones eliminate the need for a traditional proprietary telephone set and key system and PBX.

## Configuring a Cisco ATA 186/188 Device

For further details, refer to the Cisco ATA 186/188 Adaptor Administration Guide.

**Step 1** Configure a DHCP server to set up the network configuration for the adaptor.

**Note** If your Cisco IP phone network contains a DHCP server, the Cisco ATA adaptor automatically learns its IP address, subnet mask, and network gateway from the DHCP server when the adaptor starts up.

If the DHCP Server is not available, manually assign each network parameters.

**Step 2** Configure the TFTP server which will store the configuration files and firmware image.

**Note** Use the steps from the “Configuring SIP Parameters via a TFTP Server” section of the Cisco IP Phone 7905 documentation.

**Step 3** Download the required files for SIP phone to the root directory of TFTP server. The files required are:

- Cisco IP Phone 7905 SIP image LD0xxxSIPxxxxxxx.zup .ld1234abcd3456
- SEP<MACADDR>.cnf.xml e.g (SEP0008a3d31e4a.cnf.xml .. specific for a phone)

  or
- XMLDefault.cnf.xml Default config file downloaded to all adaptors that provide the image.

**Step 4** Set up the adaptor configuration, using Configuring Cisco ATA 186/188 Adaptor-Specific Required Information, page 1-3. The adaptor-specific configuration files are added to the root directory.

**Step 5** Use the Web page to edit the configuration, or modify the configuration side, and press the button after.

**Note** To modify the file for the Cisco ATA 186/188 devices, you also can lift the handset and press the ATA function button to get to the Configuration menu. The Configuration menu allows for inputting key sequences to accomplish minimal configuration changes.

**Step 6** Set up the TFTP IP address on the phone. If the adaptor has booted and the network parameters (IPaddr, etc.) are configured, set up the TFTP server IP address if it is not set.

a. Select **NetworkConfig ->AlternatetftpServer**.

b. Set it to **Yes**.

c. Select **TFTP Server** and set the IP address of the TFTP server.

# Configuring Cisco ATA 186/188 Adaptor-Specific Required Information

**Step 1** Create a **ld<lowercase macaddr>.txt** file.

**Step 2** Convert **ld<macaddr>.txt** to bin using cfgfmt.exe. Make sure the ptag.dat file is in the same directory as cfgfmt.exe. Run a Windows Command Window at the command prompt >.

```
cfgfmt ld<macaddr>.txt ld<macaddr>.
```

The following steps elaborate the contents of ld<lowercase macaddr>.txt file.

a. Set the tftp server_ip, image ID, and image file name in the adaptor specific configuration file using the following command:

```
upgradecode:3,0x501,0x0400,0x0100,tftp_server_ip,69,image_id,image_file_name
```

***Example 1-1 Sample tftp server_ip and image file name***

```
upgradecode:3,0x501,0x0400,0x0100,4.5.6.7,69,0x030218A,LD0101SIP030218A.zup
```

b. Enter the UI Password GUI interface password.

```
UIPassword:password
```

c. Enable or disable the DHCP Server.

```
dhcp:1
```

d. Enter the proxy server information (add the Cisco BTS 10200 Registrar or Proxy FQDN).

```
Proxy:domainname.com
```

e. Enter the UID User Phone number.

```
UID:4695557907
```

f. Enter the Password Login Authentication information.

```
PWD:user
LoginID:user
```

g. Enter the UserLoginId To enable login ID.

```
UseLoginID:1
```

h. Enter the SIPRegOnEnable/Disable registration.

```
SIPRegOn:1
```

i. Enter the Codec Set up.

```
RxCodec:2
TxCodec:2
```

j. Specify the Timezone.

```
Timezone:20
```

k. Enter the DNS1IP.

```
DNS1IP:1.2.3.4
```

l. Enter the UseTftpEnable/Disable TFTP server.

```
UseTftp:1
```

## Configuring a Cisco IP Phone 7905

For further details, refer to the Cisco IP Phone 7905 Series Administration Guide.

**Step 1** Configure a DHCP server to set up the network configuration for phone.

> **Note** If your Cisco IP phone network contains a DHCP server, the Cisco IP phone automatically learns its IP address, subnet mask, and network gateway from the DHCP server when the phone starts up.

If the DHCP Server is not available, manually assign each network parameters.

**Step 2** Configure the TFTP server which will store the configuration files and firmware image.

> **Note** Use the steps from the "Configuring SIP Parameters via a TFTP Server" section of the Cisco 7905 documentation.

**Step 3** Download the required files for SIP phone to the root directory of TFTP server. The files required are:

- Cisco 7905 SIP image LD0xxxSIPxxxxxxx.zup .ld1234abcd3456
- SEP<MACADDR>.cnf.xml e.g (SEP0008a3d31e4a.cnf.xml .. specific for a phone)

  or
- XMLDefault.cnf.xml Default config file downloaded to all phones that provides the image.

**Step 4** Set up the phone configuration, using Configuring Cisco ATA 186/188 Adaptor-Specific Required Information, page 1-3. The phone specific configuration files are added to the root directory.

**Step 5** Use the Web page to edit the configuration, or unlock the phone to edit configuration.

To edit using the phone:

a. Use ****#** to unlock.

b. Select **Highlight** to edit the parameter.

c. Make the changes, and press the **SAVE** softkey.

**Step 6** Set up the TFTP IP address on the phone. If the phone has booted and the network parameters (IPaddr, etc.) are configured, set up the TFTP server IP address if it is not set.

a. Select **NetworkConfig ->AlternatetftpServer**.

b. Set it to **Yes**.

c. Select **TFTP Server** and set the IP address of the TFTP server.

# Configuring Cisco IP 7905 Phone-Specific Required Information

**Step 1** Create a **ld<lowercase macaddr>.txt** file.

**Step 2** Convert **ld<macaddr>.txt** to bin using cfgfmt.exe. Make sure the ptag.dat file is in the same directory as cfgfmt.exe. Run a Windows Command Window at the command prompt >.

```
cfgfmt  ld<macaddr>.txt  ld<macaddr>
```

The following steps elaborate the contents of ld<lowercase macaddr>.txt file.

a. Set the tftp server_ip, image ID, and image file name in the phone specific configuration file using the following command:

```
upgradecode:3,0x501,0x0400,0x0100,tftp_server_ip,69,image_id,image_file_name
```

***Example 1-2 Sample tftp server_ip and image file name***

```
upgradecode:3,0x501,0x0400,0x0100,6.7.8.9,0x030218A,LD0101SIP030218A.zup
```

b. Enter the UI Password GUI interface password.

```
UIPassword:paswword
```

c. Enable or disable the DHCP Server.

```
dhcp:1
```

d. Enter the proxy server information (add the Cisco BTS 10200 Registrar or Proxy FQDN).

```
Proxy:domainname.com
```

e. Enter the UID User Phone number.

```
UID:4695557907
```

f. Enter the Password Login Authentication information.

```
PWD:user
LoginID:user
```

g. Enter the UserLoginId To enable login ID.

```
UseLoginID:1
```

h. Enter the SIPRegOnEnable/Disable registration.

```
SIPRegOn:1
```

i. Enter the CODEC Set up.

```
RxCodec:2
TxCodec:2
```

j. Specify the Timezone.

```
Timezone:20
```

k. Enter the DNS1IP.

```
DNS1IP:1.2.3.4
```

l. Enter the UseTftpEnable/Disable TFTP server.

```
UseTftp:1
```

# Provisioning the Cisco IP Phone 7960 for Initial Setup

The following steps are for the initial setup of a Cisco 7960 SIP phone. For further details refer to the Cisco IP Phone 7940/7960 Series Administration Guide.

**Step 1** Configure a DHCP server to set up the network configuration for phone.

**Note** Use the steps from the "Configuring Network Parameters via a DHCP Server" section of the Cisco 7960 documentation.

If the DHCP Server is not available, manually assign each network parameters.

**Step 2** Configure the TFTP server, which will store the configuration files and firmware image.

**Note** Use the "Configuring SIP Parameters via a TFTP Server" section of the Cisco 7960 documentation.

**Step 3** Download the required files for SIP phone to the root directory of TFTP server. When finished downloading, the following files should appear:

- OS79XX.TXT (contains an image name)
- the image file, such as P0S3-04-4-00 or P0S3-04-4-00.bin

**Note** The second character in the file above is a zero, not the letter O. For more information about the image name and file, refer to the Cisco 7940/7960 phone configuration guide.

- SIPDefault.cnf (Phone Global Parameters)
- SIP<MAC>.cnf (for example, SIP003094C25D40.cnf) (SIP<MAC> is the mac-id)

For more information on the files, refer to the Cisco 7960 SIP phone guide.

**Step 4** Set up the phone configuration, using "Creating a Cisco 7960 Phone-Specific Configuration" section on page 1-7.

You can add the phone-specific configuration files to a subdirectory (such as sip_phone). Set tftp_dir: /sip_phone in the SIPDefault.cnf file to allow the phone to get the phone-specific configuration file from that subdirectory (such as the sip_phone file).

**Step 5** Unlock to edit configuration.

a. Select **settings-> option 9**. If Option 9 (unlock config) is present, select it and enter the password **cisco**.

b. Select the parameter to edit, then select **EDIT**. Make the changes and then choose **SAVE**.

**Step 6** Set up the TFTP IP address on the phone.

**Step 7** (Optional) If the phone has booted, but the TFTP server IP address is not automatically obtained by the phone, then set it up as follows:

a. Select **NetworkConfig -> Enable AlternatetftpServer**.

b. Set it to **Yes**.

c. Select **TFTP Server** and set the IP address of the TFTP server.

For more information about the TFTP server, refer to the Cisco 7960 SIP phone guide.

## Creating a Cisco 7960 Phone-Specific Configuration

The following task allows you to create a File SIP<upper case MacAddr>.cnf for each phone.

You must prepare the SIP<uppercase MacAddress>.cnf configuration file for the phone, then change the following parameter for line1 to set up a single line on the phone. To set up multiple lines on the phone, add the information to multiple lines.

**Step 1** Change line1 Extension\User ID

**line1_name: "9025551232"; Line 1**
For Extension number line1_name: "51232" ; Line 1

**Step 2** Enter the line1 display name.

**line1_displayname: "SIP8"**

**Step 3** Enter the line1_authname used for authenticating all requests from the phone.

**line1_authname: "cisco" ; Line 1**

**Step 4** Enter the authentication.

**line1_password: "cisco" ; Line 1**

**Step 5** Add the Proxy Address, which is the IP address of the CA if it's a SIP subscriber; otherwise, add the Proxy IP address.

**proxy1_address: 4.5.6.7**

**Step 6** Enter the Proxy Port; add the CA Port if it's a SIP subscriber. Otherwise, add the Proxy Port.

**proxy1_port: 5060**

**Step 7** Add the XML file, dialplan.xml, that specifies the dialplan desired to /tftpboot/sip_phone.

**dial_template: "dialplan"**

## Connect Cisco IP Phone 7960 to Cisco BTS 10200

For SIP subscribers, AOR must be provisioned. The user portion of the AOR must be the phone number specified in the linex specification in the phone configuration file. The host portion of the AOR must be the proxy address specified in the linex specification in the phone configuration file (and provisioned in the Serving Domain Name table). For more information, see the Address of Record to Subscriber section in the *Cisco BTS 10200 Softswitch SIP Protocol User Guide*.

## Cisco BTS 10200 Softswitch Phone Mapping

Table 1-1 shows only the correspondence between the fields in the Cisco BTS 10200 CLI provisioning and the SIP phone configuration file (SIP<macaddr>.cnf). The table does not include all of the provisioning details for either the Cisco BTS 10200 or for the phones.

*Table 1-1 Cisco BTS 10200 Softswitch Phone Mapping*

| Cisco BTS 10200 Provisioning | Cisco ATA 186/188 | SIP 7905/7912 Phone | SIP 7940/7960 Phone (SIP<macaddr.cnf> |
|---|---|---|---|
| Auth-realm<br>add AUTH_REALM_ID=ciscolab; | NA | NA | NA |
| Serving Domain Name<br>add SERVING_DOMAIN_NAME DOMAIN_NAME=sia-SYS21CA146.ipclab.cisco.com; AUTH_REALM_ID=ciscolab; AUTH_REQD=Y; DESCRIPTION=Cisco Internal; | Proxy: sia-domainname.com:5060 | Proxy: sia-domainname.com:5060 | proxy1_address:<br>sia-domainname.com<br>proxy1_port : 5060 |
| Subscriber – DN<br>add subscriber id=sip_sub4; CATEGORY=INDIVIDUAL; NAME=sipsub4; DN1=4167940001; SUB-PROFILE-ID=sub_profile; TERM-TYPE=SIP; AOR_ID= 4167940001@sia-SYS21CA146.ipclab.cisco.com; | UID:“4167940001” | UID:“4167940001” | line1_name:“4167940001” |
| User Auth<br>add USER_AUTH AUTH_USER=SIP_7940_ONE; AUTH_REALM_ID=ciscolab; PASSWORD=cisco; AOR_ID=4167940001@sia-SYS21CA146.ipclab.cisco.com; | LoginID: SIP_7940_ONE (same as 7960)<br>UseLoginID: 1<br>(To use login ID for authentication. LoginID is used if authenticate user information is different from UID.)<br>PWD:cisco | LoginID: SIP_7940_ONE (same as 7960)<br>UseLoginID: 1<br>(To use login ID for authentication. LoginID is used if authenticate user information is different from UID.)<br>PWD:cisco | line1_authname:“SIP_7940_ONE”<br>line1_password:“cisco” |

<table>
<tr><th>Cisco BTS 10200 Provisioning</th><th>Cisco ATA 186/188</th><th>SIP 7905/7912 Phone</th><th>SIP 7940/7960 Phone (SIP<macaddr.cnf></th></tr>
<tr><td>In opticall.cfg, the EMS DNS name is used as the TSAP address for the HTTP server.<br>DNS_FOR_EMS_SIDE_A_CRIT_COM=crit-aSYS21EMS.ipclab.cisco.comadd http-feature-server id=mba; TSAP_ADDR_SIDEA=crit-aSYS21EMS.ipclab.cisco.com; TYPE=HTTP</td><td>NA</td><td>NA</td><td>services_url: “http://crit-aSYS21EMS.ipclab.cisco.com:5252”</td></tr>
<tr><td>The Pilot number to the Voice Mail server.<br>For POTS subscribers, the dial-plan for the trunk is sufficient.<br>Trunk group type Subscriber is required for voice mail support for Centrex subscribers only.<br>add subscriber id=UM;category=PBX;dn1=469-555-2001;tgn-id=21;sub-profile-id=sp1;term-type=TG;</td><td>To access Voice Mail using messages button on SIP phone.<br>VoiceMailNumber: “4695555555”<br>The Pilot number can also be specified as a Centrex group extension, if the voice mail system is provisioned as a Centrex SIP trunk.</td><td>To access Voice Mail using messages button on SIP phone.<br>VoiceMailNumber: “4695555555”<br>The Pilot number can also be specified as a Centrex group extension, if the voice mail system is provisioned as a Centrex SIP trunk.</td><td>To access Voice Mail using messages button on SIP Phone.<br>messages_uri: “4695555555”<br>The Pilot number can also be specified as a Centrex group extension, if the voice mail system is provisioned as a Centrex SIP trunk.</td></tr>
<tr><td>For regional settings, both phones need to support separate Dial templates to use feature activation/deactivation keys.</td><td>DialPlan:*St4-|#St4-|911|1>#t8.r9t2-|0>#t811.rat4-|^1t4>#.-<br>Add the dial_template defined for regional settings as necessary.</td><td>DialPlan:*St4-|#St4-|911|1>#t8.r9t2-|0>#t811.rat4-|^1t4>#.-<br>Add the dial_template defined for regional settings as necessary.</td><td>dial_template: “star_region_dialplan”<br>Add the dial_template defined for regional settings as necessary. The dial-plan template star_region_dialplan.xml must be defined.</td></tr>
</table>

# Configuring the HTTP-FS, MBA, GUI-FS, and SIP Phone Services

The BTS 10200 supports the use of the “services” key on the SIP phone through the HTTP feature server (HTTP-FS). The HTTP-FS is an optional component of the BTS 10200.

This feature can only be used with phones that support HTM services using softkey (for example, the Cisco 7960).

## Understanding the HTTP Feature Server Component

The HTTP-FS is comprised of two subcomponents: the GUI feature server (GUI-FS) and the Mini-Browser Adapter (MBA). To use the HTTP-FS, you must install both of these subcomponents:

- Install the software package for the GUI-FS, which runs in the Feature Server for POTS/Tandem/Centrex (FSPTC).

- Purchase the Sun Fire V240 hardware, connect the cables, load Solaris 8, and install the MBA software package. Details are as follows:
  - Obtain the appropriate hardware for the MBA. For information on the specific hardware and operating system, see the "Optional Component (Hardware and Software)" section in the *Cisco BTS 10200 Softswitch Release Notes*.
  - Cable the MBA (one signaling connection and one management connection) according to "Appendix A: Cable List" in the *Cabling, VLAN, and IRDP Setup Procedure*.
  - Install the MBA software according to the instructions below ("Installing the MBA" section on page 1-10).

Once installed, these two subcomponents form the HTTP-FS.

# HTTP Feature Server in the Network

For additional information about the functions of the MBA, GUI-FS, and HTTP-FS in the network, see the "HTTP-FS Functions" section in the *Cisco BTS 10200 Softswitch System Description*.

# Installing the MBA

The MBA software must be loaded on a separate host machine (Sun Fire V240), not on any BTS 10200 EMS or CA/FS node.

To load the operating system (Solaris 8) and MBA software, follow the steps in the "Installing the Mini-Browser Adaptor (MBA) Application on Another Machine" section of the *Cisco BTS 10200 Softswitch Application Installation Release 4.5.13, 4.5.0V14 and above, 4.5.1* document.

**Caution** Do not perform any steps in the *Cisco BTS 10200 Softswitch Application Installation* document except those listed in the "Installing the Mini-Browser Adapter (MBA) Application on Another Machine" section. This MBA software is intended to be installed on a separate host. It must *not* be loaded on any other BTS 10200 host machine (EMS or CA/FS).

# Provisioning the GUI Feature Server

This section identifies GUI Feature Server (GFS) provisioning. Cisco BTS 10200 supports SIP client/handset text-based user interface (UI) provisioning for a select set of features, a contrast to many supplementary features supported natively by the SIP client/handset itself. Some features require updating; Cisco BTS 10200 supports SIP clients/handsets to update end user feature access status on the switch network database.

Provisioning refers to activating or deactivating a feature, and setting any applicable Directory Numbers (DNs) associated with the feature. If a SIP handset is used, use the phone's LCD panel as a menu display for feature provisioning. If using a SIP software client, provision the features in the UI display region of the client software.

## Configuration

**Step 1** Use the –start_gfs command-line parameter for POTS feature in platform configuration file to turn on the GUI Feature Server. This is ON by default.

**Step 2** If GUI FS is activated, the –gfsDn parameter to POTS should specify the configured domain name for the GFS that allows communication between EMS and the GFS host.

## Office Provisioning

**Step 1** Add the HTTP server.

```
add http-feature-server id=mba;TSAP_ADDR_SIDEA=prica30.ipclab.cisco.com:11227;TYPE=HTTP;
```

**Step 2** Add SCTP association profile.

```
add sctp-assoc-profile id=sctp_prof_http;bundle-timeout=500; max-assoc-retrans=5;
max-path-retrans=5; retrieve-flag=N; max-rto=6000; min-rto=301; sack-timeout=101;
hb-timeout=1000
```

**Step 3** Add SCTP association.

```
add sctp-assoc id=assoc_http; sctp-assoc-profile-id=sctp_prof_http;remote-port=5253;
remote-tsap-addr1=priems45; platform-id=FSPTC235; DSCP=AF11; ip-tos-precedence=ROUTINE;
local-rcvwin=18000; max-init-retrans=3; max-init-rto=500; ULP=HTTP;
http-feature-server-id=mba;
```

**Step 4** Put the association IN service.

```
control sctp-assoc id=assoc_http; target-state=INS; mode=FORCED;
```

**Step 5** Verify SCTP association.

```
status sctp-assoc id=assoc_http;
```

## SIP Subscriber Services

Individual SIP subscriber provisioning is necessary for delivering GFS features to SIP subscribers, but is outside the scope of the GUI Server provisioning. See the "Provisioning a SIP Subscriber" section on page 1-12 section for individual GUI feature subscriber provisioning.

## MAC to Subscriber

The MAC to Subscriber (MAC2SUB) table links the MAC address of a device to a subscriber ID. The MAC2SUB table is required to use the GUI interface for feature provisioning on a SIP phone. The table is system generated when the token is used in the Subscriber table, or it can be manually added.

***Example 1-3 MAC to Subscriber example***

```
add mac2sub mac-id=SIP0002B9A74E4C; sub-id=sub1;
```

Where:

MAC-ID= MAC ID (Mac Address) of the IP phone or device.

SUB-ID= Subscriber ID.

When provisioning SIP subscribers, you also can specify the MAC ID.

## Setting Up Services

**Step 1** Modify the SIP phone-specific .cnf file on the TFTP server by setting the “services_url” equal to the HTTP address of MBA (along with the port number, such as “services_url=http://1.2.3.4:5252”).

**Step 2** Re-boot the IP phone(s).

# Provisioning a SIP Subscriber

The following steps are required to add a SIP subscriber.

Only the CLI commands for new fields or new tables specific to SIP subscribers are provided in this section. The CLI commands for existing tables such a sub_service_profile, dial_plan, etc. required for the subscriber are not described in this section.

**Note** You can use a combination of CLI commands in Step 9 to add the subscriber and all that subscriber’s related child tables.

**Step 1** Add to AUTH_REALM.

```
add auth_realm
id = ciscolab; description=Cisco Internal;
```

**Step 2** Add the SERVING_DOMAIN_NAME.

The domain name or the IP address in the DomainName field is added. If authentication is required on the phones, set AUTH_REQD=‘y’.

```
add serving-domain-name
domain_name=domainname.com; auth_realm_id=ciscolab; auth_reqd=n;
description=Cisco Internal;
```

**Step 3** Add a SIP subscriber.

```
add subscriber
ID=sip_sub1; CATEGORY=INDIVIDUAL; NAME=SipSub1; STATUS=ACTIVE; LANGUAGE=english;
BILLING-DN=469-555-1111; DN1=469-555-1111; RING-TYPE-DN1=1; SUB-PROFILE-ID=sub_profile;
TERM-TYPE=SIP; AOR-ID=4695551111@cisco.com; privacy=user;
```

**Note** 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.

To use the CLI command combination, see Step 9.

**Step 4** Add the USER_AUTH entry.

This is used only if Auth-Reqd in the serving_domain_name is set to “Y”.

```
add user_auth
```

```
auth_user=sipsub1; auth_realm_id=ciscolab; aor_id=4695551111@domainname.com;
password=cisco_sipsub1;
```

**Step 5** In a lab environment, if the device is not capable of registering itself, a static contact may be used.

```
add static_contact
static_contact_host=172.16.33.77; static_contact_port=5060;
aor_id=4695551111@domainname.com; user_type=phone;
```

**Step 6** Add MAC2SUB.

Required to use the GUI interface for feature provisioning on SIP phone.

```
add mac2sub
mac_id=SIP0008A3D31E4A; sub_id=sip_sub1;
```

To use the CLI command combination, see Step 9.

**Step 7** Provision CA-CONFIG to provide min, max and default value for register expires. If not provisioned the default values for each parameter will be used.

For details, refer to the CA-CONFIG SIP Adapter Configuration Parameters section of the *Cisco BTS 10200 Softswitch SIP Protocol User Guide*.

```
add ca-config
type=SIA_REG_MIN_EXPIRES_SECS; datatype=INTEGER; value=1800;

add ca-config
type=SIA_DEFAULT_REG_EXPIRES; datatype=INTEGER; value=3600;

add ca-config
type=SIA_REG_MAX_EXPIRES_SECS; datatype=INTEGER; value=36000;
```

**Step 8** Put AOR in Service.

```
change aor2sub
aor_id=4695551111@domainname.com; status=INS;
```

**Step 9** Step 3 (Subscriber) and Step 6 (MAC2SUB) can be combined by a single CLI command.

```
add subscriber
id=sip_sub1; CATEGORY=INDIVIDUAL; NAME=SipSub1; STATUS=ACTIVE; LANGUAGE=english;
BILLING-DN=469-555-1111; DN1=469-555-1111; RING-TYPE-DN1=1; SUB-PROFILE-ID=sub_profile;
TERM-TYPE=SIP; aor_id=4695551111@domainname.com; mac_id=SIP0008A3D31E4A; privacy=user;
```

**Note** 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.

# Provisioning Subscriber Features

This section describes how to provision Subscriber features. Existing features introduced prior to Release 4.5.x are hyperlinked to the *Cisco BTS 10200 Softswitch Release 4.5.x Provisioning Guide*, and the differences for provisioning those features when using SIP are listed in the following feature descriptions.

# Activation and Deactivation of Anonymous Call Rejection

This feature was introduced in a previous Cisco BTS 10200 Softswitch release. Provisioning the feature is the same as MGCP when provided by Cisco BTS 10200. ACR is also provided by phone.

For information on provisioning ACR, refer to the Anonymous Call Rejection section in the *Cisco BTS 10200 Softswitch Provisioning Guide*.

# Billing

For detailed information on billing management and data, refer to the *Cisco BTS 10200 Softswitch Billing Interface Guide*.

# CALEA Call Content

CALEA is not available for SIP subscribers.

# Call Forwarding

For information about the feature and all of its options, refer to Call Forwarding Features section in the *Cisco BTS 10200 Softswitch Release 4.5.x Provisioning Guide*.

The Call Forwarding feature was introduced in a previous Cisco BTS 10200 Softswitch release. The difference between the feature for SIP versus MGCP is 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 will always be played an announcement (if announcements are provisioned) or a re-order tone.

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

In Release 4.5.x, activation is accomplished using single-stage dialing. This applies to all activation and deactivation.

# Calling Name and Number Delivery

These features were introduced in a previous Cisco BTS 10200 Softswitch release. There are no differences when provisioning the features for SIP.

For information on provisioning Calling Name Delivery (CNAM), refer to the Calling Name Delivery section in the *Cisco BTS 10200 Softswitch Provisioning Guide*.

For information on provisioning Calling Number Delivery (CND), refer to the Calling Number Delivery section in the *Cisco BTS 10200 Softswitch Provisioning Guide*.

# Caller ID Delivery Suppression

This feature was introduced in a previous Cisco BTS 10200 Softswitch release. The difference when provisioning the feature for SIP versus MGCP is as follows:

- Presentation status from the phone, and single stage digit collection.

For information on provisioning Caller ID Delivery Suppression, refer to the Calling Number Delivery Suppression—Delivery (CIDSD) section and the Calling Number Delivery Suppression—Suppression (CIDSS) section in the *Cisco BTS 10200 Softswitch Provisioning Guide*.

# Called Party Termination

This feature was introduced in a previous Cisco BTS 10200 Softswitch release. There are no differences when provisioning the features for SIP.

# Cisco BTS 10200 Supplementary Vertical Service Code Features

For information on provisioning Vertical Service Codes (VSC), refer to the Vertical Service Code Provisioning section in the Cisco BTS 10200 Softswitch Provisioning Guide.

# Customer Access Treatment

This feature was introduced in a previous Cisco BTS 10200 Softswitch release. There are no differences when provisioning the features for SIP.

For information on provisioning Customer Access Treatment (CAT), refer to the section in the *Cisco BTS 10200 Softswitch Provisioning Guide*.

# Customer-Originated Trace

Use the following CLI for Centrex Subscriber provisioning with the Customer-Oriented Trace (COT) feature:

```
add cdp id=cdp1;DIGIT_STRING=*57;NOD=VSC;FNAME=COT
```

For information on provisioning Customer-Originated Trace (COT), refer to the Customer-Originated Trace section in the *Cisco BTS 10200 Softswitch Provisioning Guide*.

# Direct Inward Dialing

There are no special instructions to provision Direct Inward Dialing (DID), other than assigning the DID number to that subscriber as DN1 in the subscriber table.

# Direct Outward Dialing

The following subsections identify necessary steps for the Custom Dial Plan (CDP) feature to be offered.

### Office Provisioning

**Step 1** Provision the feature table.

```
add/change feature FNAME=CDP; TDP1= COLLECTED_INFORMATION; TID1= CUSTOMIZE_DIALING_PLAN;
TTYPE1=R; FEATURE_SERVER_ID=FSPTC325; DESCRIPTION=Custom Dial Plan Feature;
```

**Step 2** Provision the service table.

```
add service id=2, FNAME1=CDP;
```

## Do Not Disturb

This feature was introduced in a previous Cisco BTS 10200 Softswitch release. The difference when provisioning the feature for SIP versus MGCP is as follows:

- Provisioning is the same as MGCP; the difference is activation. Do Not Disturb (DND) can be activated or deactivated from Cisco BTS 10200. Alternatively, activation and deactivation may also be provided through a key on the phone.

For features (such as DND) that can be fully provisioned on the Cisco BTS 10200 Softswitch or on the phone, 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.

For information on provisioning DND, refer to the Do Not Disturb section in the *Cisco BTS 10200 Softswitch Provisioning Guide*.

## Emergency Call

This feature was introduced in a previous Cisco BTS 10200 Softswitch release. The difference when provisioning the feature for SIP versus MGCP is as follows:

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

For information on provisioning Emergency Call (E911), refer to the 911 Emergency Call section in the *Cisco BTS 10200 Softswitch Provisioning Guide*.

## E.164 and Centrex Dialing Plan (Extension Dialing)

Provision the subscriber-service-profile:

```
add subscriber-service-profile sub_id=sub_1;service-id=2;
```

**Note** CDP feature should be assigned to every CENTREX category users.

## Incoming and Outgoing Simulated Facility Group

This feature was introduced in a previous Cisco BTS 10200 Softswitch release. There are no differences when provisioning the feature for SIP.

For information on provisioning Incoming Simulated Facility Group (ISFG), refer to the Incoming Simulated Facility Group section in the *Cisco BTS 10200 Softswitch Provisioning Guide*.

For information on provisioning Outgoing Simulated Facility Group (OSFG), refer to the Outgoing Simulated Facility Group section in the *Cisco BTS 10200 Softswitch Provisioning Guide*.

## Multiple Directory Numbers

This feature was introduced in a previous Cisco BTS 10200 Softswitch release. The difference when provisioning the feature for SIP versus MGCP is as follows:

- Ringing part supported by Cisco BTS 10200. Cisco BTS 10200 sends a distinctive alerting request for Call-Waiting scenario; some SIP-Phones interpret it and play distinctive call-waiting tone, while others do not.

For information on provisioning Multiple Directory Numbers (MDN), refer to the Multiple Directory Numbers section in the *Cisco BTS 10200 Softswitch Provisioning Guide*.

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

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

## Outgoing Call Barring

This feature was introduced in a previous Cisco BTS 10200 Softswitch release. There are no differences when provisioning the feature for SIP.

For information on provisioning Outgoing Call Barring (OCB), refer to the Outgoing Call Barring section in the *Cisco BTS 10200 Softswitch Provisioning Guide*.

## Remote Activation of Call Forwarding

This feature was introduced in a previous Cisco BTS 10200 Softswitch release. There are no differences when provisioning the feature for SIP.

For information on provisioning Remote Activation of Call Forwarding (RACF), refer to the Remote Activation of Call Forwarding section in the *Cisco BTS 10200 Softswitch Provisioning Guide*.

## 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.

# Provisioning Secure FQDN of a SIP Endpoint

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

Enhanced SIP Registration was added to Release 4.5.x to ensure that a SIP REGISTER message to the Cisco BTS 10200 is from a provisioned endpoint. This feature also provides the option for the system to verify that the source IP address and contact parameter for all originating calls is from the provisioned SIP endpoint, and that no calls can originate from an unregistered endpoint.

**Note** This procedure explains how to provision subscribers on the Cisco BTS 10200 Softswitch. It does not discuss the security of configuration files provisioned on the SIP adapter (for example, an ATA), which are the responsibility of the service provider.

## 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, Release 4.5.1 and later) 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 on a successfully registered subscriber, enter the following commands.

**Step 1** Take the AOR out of service (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 in service (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.

For additional information about the secure FQDN feature, see the "Enhanced SIP Registration" section in the *Cisco BTS 10200 Softswitch SIP Protocol User Guide*.

# SIM Memory Audit and SIP Dynamic Memory Audit

The following are examples of provisioning the activity table and changes for the SIM memory audit.

```
add activity id=SIM-MEMORY-PERIODIC-AUDIT; freq=1H; enabled=y;

add activity id=SIM-MEMORY-SCHEDULED-AUDIT; freq=DAILY; start_time=2:00;
enabled=y;
```

The following are examples of provisioning the activity table and changes for the SIP dynamic memory audit.

```
add activity id=SIA-MEMORY-PERIODIC-AUDIT; freq=1H; enabled=y;

add activity id=SIA-MEMORY-SCHEDULED-AUDIT; freq=DAILY; start_time=2:00;
enabled=y;
```

For additional details on these tables, see Chapter 5, "Database Table" in the *Cisco BTS 10200 Softswitch SIP Protocol User Guide*.

For additional description of these audit processes, see Chapter 2, "SIP Protocol Subscriber Features" in the *Cisco BTS 10200 Softswitch SIP Protocol User Guide*.

# Type of Service

The ToS value for messages sent to SIP subscribers can be set on a system-wide basis—this applies to all subscribers. The policy is selected in the CA-CONFIG table. If the ToS entries are not provisioned in CA-CONFIG table, the following defaults apply:

- Precedence = FLASH (3)
- Delay = low (Y)
- Throughput = normal (N)
- Reliability = normal (N)

**Note** These are the recommended values; these values should be changed only after careful consideration, or if there is a specific need.

## Provisioning Network-Level ToS

The ToS value for messages sent to SIP subscribers can be set on a system-wide basis—this applies to all subscribers. The policy is selected in the CA-CONFIG table. The Cisco BTS 10200 reads the values from this table when it starts up. Therefore, changes to the ToS policy for SIP subscribers become effective at the next restart of the Cisco BTS 10200. If the ToS entries are not provisioned in CA-CONFIG table, the following defaults apply:

- Precedence = immediate (010)
- Delay = low (1)
- Throughput = normal (0)
- Reliability = normal (0)

These are the recommended values; these values should be changed only after careful consideration, or if there is a specific need.

**Caution** If you change any parameters in the ca-config table, these changes do not take effect until the CA platform switches over or restarts.

## Provisioning Type of Service Default Settings for SIP Subscribers

**Note** Note that the 'SIA-TRUNK-GRP-LEVEL-SIG-TOS' flag in call agent configuration is used to select between using TOS settings for all SIP trunks or TOS settings for specific SIP trunks.

**Step 1** Add the SIA-SIG-TOS-LOWDELAY value.

```
add ca-config type=SIA-SIG-TOS-LOWDELAY; datatype=BOOLEAN; value=Y;
```

**Step 2** Add the SIA-SIG-TOS-PRECEDENCE.

```
add ca-config type=SIA-SIG-TOS-PRECEDENCE; datatype=INTEGER; value=2;
```

**Step 3** Add the SIA-SIG-TOS-RELIABILITY value.

```
add ca-config type=SIA-SIG-TOS-RELIABILITY; datatype=BOOLEAN; value=N;
```

**Step 4** Add the SIA-SIG-TOS-THROUGHPUT value.

```
add ca-config type=SIA-SIG-TOS-THROUGHPUT; datatype=BOOLEAN; value=N;
```

# Phone-Based Features

Phone-based features are provided by the SIP phone, which require provisioning on the phone.

There are some features that the phone provides standalone, without Cisco BTS 10200 support.

The Cisco BTS 10200 Softswitch supports interface requirements (such as Re-INVITE support) that are necessary to operate features from the SIP phones, including but not limited to:

- Call Hold and Resume
- Call Waiting
- Three-Way Calling
- Cancel Call Waiting
- Call Waiting Caller ID
- CODEC Up-speeding
- Do Not Disturb (DND)

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

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

For information on provisioning these features, refer to the SIP phone documentation.

# Jointly-Provided Features

In addition to the Softswitch-based and phone-based features, the system offers features provided jointly by the Cisco BTS 10200 and the phone. To use these features, you must provision both the Cisco BTS 10200 and the phone.

## Session Timers

Use the commands in this section to provision session timers on the Cisco BTS 10200 Softswitch.

**Note** To configure SIP protocol and session timers in Release 4.5.x, you must use the new sip-timer-profile table. For customers upgrading to Release 4.5.x: SIP session timer values configured in the ca-config table prior to Release 4.5.x are reset to the default values after upgrading to Release 4.5.x. In Release 4.5.x and later, the session timer values are provisioned through the MIN-SE and SESSION-EXPIRES-DELTA-SECS tokens in the sip-timer-profile table. The id of the sip-timer-profile table record is then specified as the Value for the ca-config record of Type=sip_timer_profile_id.

**Step 1** Adjust the session timer values in the 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** 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

Release 4.5.x enhances support for customizing SIP timers. Values for the timers listed in Table 1-2 can be provisioned in the new sip-timer-profile table. A record in this table can then be configured to apply to one or more SIP trunks or to apply switch-wide. A sip-timer-profile record can be associated with a specific softsw-tg-profile record and/or to a ca-config record. On a fresh software installation, and after a upgrade, the system operates with default SIP protocol timer values, as specified by the SIP specification. These default values are adequate for many installations. If customization is required, then a sip-timer-profile table can be provisioned and associated with all calls, or with calls on specific trunks.

**Note** The session timer parameters, MIN-SE and SESSION-EXPIRES-DELTA-SECS, have been consolidated into this new table and, unlike prior releases, are no longer configurable directly on the ca-config table.

**Note** To configure SIP protocol and session timers in Release 4.5.x, you must use the new sip-timer-profile table. For customers upgrading to Release 4.5.x: SIP timer values configured in the ca-config table prior to Release 4.5.x are reset to the default values after upgrading to Release 4.5.x. In Release 4.5.x and later, the timer values are provisioned in the sip-timer-profile table. The id of the sip-timer-profile table record is then specified as the Value for the ca-config record of Type=sip_timer_profile_id.

**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=Case_test;
```

Table 1-2 lists the timers configurable in the sip-timer-profile table.

**Note** For more detailed descriptions of these timers, see the "SIP Timers" section in Chapter 2 of the *Cisco BTS 10200 Softswitch SIP Protocol User Guide.*

***Table 1-2 SIP Timers Configurable in the SIP-TIMER-PROFILE Table***

| Timer | Values | Definition |
|---|---|---|
| TIMER-T1-MILLI | RANGE (100 – 5000) ms<br>Default = 500 | T1 timer (in milliseconds) for RTT estimate. |
| TIMER-T2-SECS | RANGE(1 – 10) seconds<br>Default = 4 | T2 timer (in seconds) specifies the maximum retransmit interval for SIP non-INVITE requests and INVITE responses. |
| TIMER-T4-SECS | RANGE(1 - 10) seconds<br>DEFAULT = 5 | T4 timer (in seconds) specifies the maximum duration a SIP message will remain in the network. |
| TIMER-A-MILLI | RANGE(100 – 5000) ms<br>DEFAULT = 0[1] | Timer A specifies the INVITE request retransmit interval (in milliseconds) for UDP only. |
| TIMER-B-SECS | RANGE(1 - 3600) seconds<br>DEFAULT = 0* | Timer B specifies INVITE transaction timeout in seconds. |
| TIMER-D-SECS<br>(0 for TCP) | RANGE(33 – 65) seconds<br>DEFAULT = 33 | Timer D (in seconds) specifies wait time for response retransmits. |
| TIMER-E-MILLI | RANGE(100 – 5000) ms<br>DEFAULT = 0* | Timer E (in milliseconds) specifies non-INVITE request retransmit interval, UDP only. |
| TIMER-F-SECS | RANGE(1 - 3600) seconds<br>DEFAULT = 0* | Timer F (in seconds) specifies non-INVITE transaction timeout. |

*Table 1-2 SIP Timers Configurable in the SIP-TIMER-PROFILE Table (continued)*

| Timer | Values | Definition |
|---|---|---|
| TIMER-G-MILLI | RANGE(100 – 5000) ms<br>DEFAULT = 0* | Timer G (in milliseconds) specifies INVITE response retransmit interval. |
| TIMER-H-SECS | RANGE(1 - 3600) seconds<br>DEFAULT = 0* | Timer H (in seconds) specifies Wait time for ACK receipt. |
| TIMER-I-SECS<br>(0 for TCP) | RANGE(1 - 10) seconds<br>DEFAULT = 0* | Timer I (in seconds) specifies Wait time for ACK retransmits. |
| TIMER-J-SECS<br>(0 for TCP) | RANGE(1 – 3600) seconds<br>DEFAULT = 0* | Timer J in seconds specifies Wait time for non-INVITE request. |
| INVITE-INCOMPLETE-TIMER-SECS | RANGE(15 – 600) seconds<br>DEFAULT = 40 | This parameter (in seconds) specifies the INVITE timeout duration once a provisional response less than 180 has been received. This timer is canceled when a response greater than or equal to 180 is received. |
| MIN-SE | RANGE(100-1800) seconds<br>DEFAULT = 900 | This is minimum acceptable value of session-expires in seconds.<br>**Note** This parameter is a session timer. |
| SESSION-EXPIRES-DELTA-SECS | RANGE(100 – 7200) seconds<br>DEFAULT = 1800 | This value (in seconds) is sent in the session-expires header.<br>**Note** This parameter is a session timer. |

1. For default 0, the timer value is computed automatically from TimerT1 and TimerT4.

## Rules for Configuring the SIP Timers

Use the following rules to configure the SIP timers in the Cisco BTS 10200 Softswitch. The rules are necessary due to mutual dependency between the timers. If any rules fail, the Cisco BTS 10200 Softswitch computes the values of the timers.

```
TIMER-T2-SECS * 1000 > TIMER-T1-MILLI
TIMER-T2-SECS * 1000 > TIMER-G-MILLI
TIMER-B-SECS * 1000 > TIMER-A-MILLI
TIMER-F-SECS * 1000 > TIMER-E-MILLI
TIMER-D-SECS > 32
```

In addition to these rules, the timer values must be in the range of values specified in Table 1-2.

## Auto Computation of Timer Values from Timer T1, T4

If the following timer values are not explicitly configured (the default=0), then they are computed based on the values of T1 and T4.

```
TIMER-A-MILLI = TIMER-T1-MILLI
TIMER-B-SECS = (64 * TIMER-T1-MILLI) / 1000
TIMER-E-MILLI = TIMER-T1-MILLI
TIMER-F-SECS = (64 * TIMER-T1-MILLI) / 1000
```

```
TIMER-G-MILLI = TIMER-T1-MILLI
TIMER-H-SECS = (64 * TIMER-T1-MILLI) / 1000
TIMER-I-SECS = TIMER-T4-SECS
TIMER-J-SECS = (64 * TIMER-T1-MILLI) / 1000
```

# Call Transfer (Blind and Attended) via Refer Feature

This feature was introduced in a previous Cisco BTS 10200 Softswitch release. The difference when provisioning the feature for SIP versus MGCP is as follows:

- Call transfer on both the Cisco IP Phone 7905/7912 and the Cisco IP Phone 7940/7960 is done using soft keys. 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. For information on provisioning the Refer feature, see the Refer section of the *Cisco BTS 10200 Softswitch Provisioning Guide*.

# Distinctive Ringing

This feature was introduced in a previous Cisco BTS 10200 Softswitch release. There are no differences when provisioning the feature for SIP.

# Distinctive Ringing for Centrex DID Calls

This feature was introduced in a previous Cisco BTS 10200 Softswitch release. There are no differences when provisioning the feature for SIP.

CHAPTER 2

# Provisioning SIP Trunks

**Revised: October 21, 2008, OL-5351-10**

This chapter provides instructions for provisioning SIP trunks. The purpose of SIP trunks is to service SIP calls between Cisco BTS 10200 and external SIP entities other than local SIP subscribers, such as a voice-mail server, remote call agent, or SIP proxy.

## Provisioning Example

The following example models a local Cisco BTS 10200 subscriber making a call out from a SIP trunk to a SIP proxy serving a NPA-NXX domain.

A trunk must be created and associated with the IP address of the proxy. The dial digits associated with the trunk must be provisioned within the originators' dial plan.

**Note** Before provisioning, identify the following:
* The first 6 dial digits of the SIP proxy NPA-NXX domain: in this example, 469-255.
* Provisioned dial plan ID of the originator in Cisco BTS 10200: in this example, 'dp1'.
* IP address of the SIP proxy: in this example, 1.2.3.4.

### Provisioning Example - CLI

```
add softsw_tg_profile id=<profile_id>; protocol_type=SIP;
add pop id=<pop_id>; state=tx; country=usa; timezone=CST;
add trunk_grp id=<trunk_id>; tg_type=SOFTSW; softsw_tsap_addr=1.2.3.4; dial_plan_id=dp1;
   tg_profile_id=<profile_id>; call_agent_id=<ca_id>; pop_id=<pop_id>;
add route id=<route_id>; tgn1-id=<trunk_id>;
add route-guide id=<route_guide_id>; policy_type=ROUTE; policy_id=<route_id>;
add destination dest-id=<dest_proxy_id>; call-type=LOCAL; route-type= ROUTE;
   route_guide_id=<route_guide_id>;
add dial-plan id=dp1; digit-string=469-255; dest-id=<dest_proxy_id>;
```

# Provisioning SIP Trunk Features

The following sections describe how to provision SIP trunk features.



## General Requirements for Provisioning SIP Trunk Groups

The TSAP_address in the outbound SIP trunk group can be provisioned with a static IP address, but the inbound SIP trunk group must be provisioned with a domain name. This is because it needs to match the domain name in the incoming INVITE message Via header. If you do not provision the TSAP_address this way, the call is rejected with 403 Forbidden message.

To avoid a DNS lookup, to use the static IP address, we suggest using at least three SIP trunk groups: two for outbound with the IP addresses of two remote softswitches, and one for inbound with the domain name of one remote softswitch.

# Validation of Source IP Address for Incoming SIP Messages (Release 4.5.1, Maintenance Release 1)

The system is capable of performing source IP address validation of incoming messages received on SIP trunks. This validation is controlled through a switch-wide parameter (applies to all SIP trunk groups on the switch). You can enable this capability using the following command.

```
add ca-config type=SIA-TG-VALIDATE-SOURCE-IP; datatype=BOOLEAN; value=Y;
```

**Note** By default, SIA-TG-VALIDATE-SOURCE-IP is set to N, and IP address validation is disabled.

# Call Redirection

The following commands control call redirection on all trunks associated to the SIP trunk profile <profile_id>.

**Step 1** Disable call redirection.

```
change softsw_tg_profile id=<profile_id>; REDIRECT_SUPPORTED=NONE;
```

**Step 2** Enable call redirection.

The trunk accepts redirection contacts only with host names of the Cisco BTS 10200 SIP contact, or the TSAP address of any provisioned SIP trunks.

The default is:

```
change softsw_tg_profile id=<profile_id>; REDIRECT_SUPPORTED=VALID_DOMAINS_ONLY;
```

**Step 3** Enable call redirection.

The trunk accepts redirection contacts with any host name. A contact URI is used as the request URL for the redirected call. The redirected call uses the properties of the SIP trunk in the previous call attempt.

```
change softsw_tg_profile id=<profile_id>; REDIRECT_SUPPORTED=ALL_DOMAINS;
```

The following parameters are provisioned through the Call Agent Configuration (ca-config) table, and affect all SIP trunks on the switch. Additional details for the ca-config table are provided in the *Cisco BTS 10200 Softswitch Command Line Interface Guide*, Chapter 1, "Call Agent Provisioning," and Appendix A, "Configurable Parameters and Values."

**Step 1** If necessary, change the upper limit on the number of 300 class redirection responses the Cisco BTS 10200 accepts while performing redirection on any given call attempt; the default is 1.

```
add ca-config type=MAX-3XX-COUNT; datatype=INTEGER; value=2;
```

**Step 2** If necessary, change the limit on the maximum number of call redirection attempts the Cisco BTS 10200 makes on any given call attempt, after which it releases the call; the default is 5.

```
add ca-config type=MAX-REATTEMPT-COUNT; datatype=INTEGER; value=4;
```

**Step 3** If necessary, set the 3XX reroute parameter for call redirection. If you want to force the system to perform fresh routing (reroute) using the dial plan of the terminating trunk, use the following command to set the 3XX reroute parameter to Y.

```
add ca-config type=SIP-3XX-REROUTE-ON-LOCAL-DOMAIN; datatype=BOOLEAN; value=Y;
```

**Note** By default, SIP-3XX-REROUT-ON-LOCAL-DOMAIN is set to N, and the system performs route advance when the redirection number is the same as the number in the original INVITE.

# Locating SIP Servers Using DNS Queries

The system can locate SIP servers using NAPTR and SRV DNS queries, or using A-Record DNS queries.

**Tip** For a description of how the system uses these queries to locate SIP servers, see the DNS query description in the *Cisco BTS 10200 Softswitch SIP Protocol User Guide*.

## Provisioning the System to Use NAPTR and SRV DNS Queries

Follow these steps to provision the system to use NAPTR and SRV DNS queries.

**Step 1** Enable NAPTR and SRV DNS queries.

```
change softsw_tg_profile id=<profile_id>; DNS_SRV_SUPP=RFC2782_LABELS;
```

**Step 2** Provision the TSAP address in the trunk group for the SIP server.

```
change trunk_grp id=<trunk group id>; softsw_tsap_addr=<see list of values below>;
```

Either of the following can be provisioned for softsw_tsap_addr:

- NAPTR name
- SRV name

**Note** The use of either NAPTR or SRV names requires correctly configured DNS servers.

### Provisioning Recommendations for NAPTR and SRV in the DNS Servers

- When using SRV, if a host name is provisioned in the TSAP address, include a port. This allows the application to identify the address as a host name and skip NAPTR and SRV queries.
- If an SRV name is required, provision NAPTR entries to provide SRV replacement strings instead of waiting for a failure on the NAPTR query to make an SRV query.

### Provisioning the System to Use A-Record DNS Queries

Follow these steps to provision the system to use A-record DNS queries.

**Step 1** Disable NAPTR and SRV DNS queries.

```
change softsw_tg_profile id=<profile_id>; dns_srv_supp=NONE;
```

> **Note** NONE is the default value for dns_srv_supp.

**Step 2** Provision the transport type.

```
change softsw_tg_profile id=<profile_id>; non_srv_transport=<see list of values below>;
```

Any one of the following can be provisioned for non_srv_transport:

- UDP (default)—If the message size is less than 1300 bytes as described in RFC 3261 and RFC 3263, the system uses UDP. If the message size is greater than 1300 bytes, the system uses TCP; however, if TCP fails, the system attempts to use UDP.
- UDP-ONLY—The initial outbound request uses UDP regardless of the message size. However, the transport used for subsequent outbound requests is based on the negotiated transport type exchanged in the Contact: header during dialog establishment.
- TCP—Use TCP only.

**Step 3** Provision the TSAP address in the trunk group for the SIP server.

```
change trunk_grp id=<trunk group id>; softsw_tsap_addr=<see list of values below>;
```

Any one of the following can be provisioned for softsw_tsap_addr.

- Host name
- Host name and port
- IP address
- IP address and port

> **Note** The use of host names requires correctly configured DNS servers.

## Type of Service

This section describes the provisionable options for TOS settings on SIP trunks.

> **Caution** If you change any parameters in the ca-config table, these changes do not take effect until the CA platform switches over or restarts.

### Provisioning SIP Trunks on the Cisco BTS 10200

The following commands set the Type of Service (ToS) default settings for all SIP trunks on the Cisco BTS 10200.

```
add ca_config TYPE=SIA-TRUNK-GRP-LEVEL-SIG-TOS; DATATYPE=BOOLEAN; VALUE=N;

add ca_config TYPE=SIA-SIG-TOS-PRECEDENCE; DATATYPE=INTEGER; VALUE=3;

add ca_config TYPE=SIA-SIG-TOS-RELIABILITY; DATATYPE=BOOLEAN; VALUE=N;

add ca_config TYPE=SIA-SIG-TOS-THROUGHPUT; DATATYPE=BOOLEAN; VALUE=N;

add ca_config TYPE=SIA-SIG-TOS-LOWDELAY; DATATYPE=BOOLEAN; VALUE=Y;
```

### Provisioning Type of Service Default Settings for SIP Trunks Associated to the SIP Trunk Profile

The following commands set the ToS default settings for SIP trunks associated to the SIP trunk profile <profile_id>:

```
add ca_config TYPE=SIA-TRUNK-GRP-LEVEL-SIG-TOS; DATATYPE=BOOLEAN; VALUE=Y;

change softsw_tg_profile id=<profile_id>; SIP_SIG_LOWDELAY=Y;

change softsw_tg_profile id=<profile_id>; SIP_SIG_THROUGHPUT=N;

change softsw_tg_profile id=<profile_id>; SIP_SIG_RELIABILITY=N;

change softsw_tg_profile id=<profile_id>; SIP_SIG_PRECEDENCE=FLASH;
```

**Note** Note that the ‘SIA-TRUNK-GRP-LEVEL-SIG-TOS’ flag in the call agent configuration is used to select between using ToS settings for all SIP trunks, or ToS settings for specific SIP trunks.

## Reliable Provisional Responses

The following commands control the reliable provisional response feature for regular SIP calls on all trunks associated to the SIP trunk profile <profile_id>.

**Step 1** The default for making reliable provisional responses not required for regular SIP calls sent or received over a SIP trunk is:

```
change softsw_tg_profile id=<profile_id>; PRACK_FLAG=N;
```

**Step 2** To make reliable provisional responses required for regular SIP calls sent or received over a SIP trunk, use the following command:

```
change softsw_tg_profile id=<profile_id>; PRACK_FLAG=Y;
```

**Note** When reliable provisional responses are not required, the Cisco BTS 10200 will not make them required on remote SIP entities. However, the reliable provisional responses may still occur if a remote SIP entity requires it of Cisco BTS 10200.

This flag must applies only to SIP calls on regular SIP trunks, and regular SIP calls received on SIP-T provisioned trunks. Consult the *Cisco BTS 10200 SIP User Guide* for details on the behavior of this feature.

# Diversion Indication

The following commands control the diversion feature for outgoing calls on all trunks associated to the SIP trunk profile <profile_id>.

**Step 1** Disable diversion headers for calls sent out the trunk.

The default is:

```
change softsw_tg_profile id=<profile_id>; DIVERSION_HEADER_SUPP=N;
```

**Step 2** Enable diversion headers for calls sent out the trunk.

```
change softsw_tg_profile id=<profile_id>; DIVERSION_HEADER_SUPP=Y;
```

**Note** This flag does not apply to incoming calls. If the diversion headers exists on the incoming call, the system interprets the information from the diversion header.

For a description of the diversion indication features, see the "Diversion Indication" section in the *SIP Protocol User Guide*.

# Carrier Identification Code over SIP

A carrier identification code (CIC) received in a SIP call on an incoming SIP trunk is automatically interpreted. No provisioning control is available. For outgoing SIP calls originated by a local Cisco BTS 10200 subscriber, the CIC may be provided by the subscriber record if provisioned. See the CIC selection rules in the trunk-grp table and the send-cic-param token in the softsw-tg-profile table in Chapter 5, "Database Tables" of the *Cisco BTS 10200 Softswitch SIP Protocol User Guide*.

For additional description of the CIC options, see the Carrier Identification Code Over SIP section in the *Cisco BTS 10200 Softswitch SIP Protocol User Guide*.

# Number Portability Information over SIP

The following commands control number portability information for calls sent out on the SIP trunk group <tg_id>.

**Step 1** The following command allow for sending number portability information if the information is available.

The default is:

```
change trunk_grp id=<tg_id>; SIGNAL_PORTED_NUMBER=N;
```

**Step 2** The following command disables the addition of number portability information to SIP calls sent out a SIP trunk group:

```
change trunk_grp id=<tg_id>; SIGNAL_PORTED_NUMBER=Y;
```

**Note** Note that number portability information received in a SIP call on an incoming SIP trunk is automatically interpreted. No provisioning control is available.

# SIP Trunk Sub-Groups

These steps illustrate how to provide multiple trunks toward a remote SIP entity for additional network-specific or application-specific properties for calls to and from the Cisco BTS 10200. One example: the identification of which rate center the call originated.

The following information is required at the time of provisioning:

- Associate a unique trunk group identifier for each rate center. For example: ‘rc1,’ ‘rc2,’ and ‘rc3’ for three rate centers.
- Identify the fully qualified domain name (FQDN) and port of the remote SIP server used for SIP message exchange. For example: ‘sipserver:5060.’
- Create a dial plan for calls received on the SIP trunks, to route the calls based on the called party number. For example: the identifier for this dial plan is ‘dp.’

**Step 1** Add a SIP trunk profile for the SIP trunks. Set the trunk sub-group type to indicate the trunk group identifier use:

```
add softsw_tg_profile ID=<profile_id>; PROTOCOL_TYPE=SIP; TRUNK_SUB_GRP_TYPE=TGID;
```

**Step 2** Add a SIP trunk for each trunk group identifier. Each trunk points to the address of the voicemail sever:

```
add trunk_grp ID=<trk_grp_id1>; TG_TYPE=SOFTSW; TG_PROFILE_ID=<profile_id>;
SOFTSW_TSAP_ADDR=sipserver:5060; DIAL_PLAN_ID=dp; TRUNK_SUB_GRP=rc1;
```

and:

```
add trunk_grp ID=<trk_grp_id2>; TG_TYPE=SOFTSW; TG_PROFILE_ID=<profile_id>;
SOFTSW_TSAP_ADDR=sipserver:5060; DIAL_PLAN_ID=dp; TRUNK_SUB_GRP=rc2;
```

and:

```
add trunk_grp ID=<trk_grp_id3>; TG_TYPE=SOFTSW; TG_PROFILE_ID=<profile_id>;
SOFTSW_TSAP_ADDR=sipserver:5060; DIAL_PLAN_ID=dp; TRUNK_SUB_GRP=rc3;
```

Routing and dial plan tables are provisioned (not shown) so that calls originating from a specific rate center are sent out the SIP trunk with the trunk group identifier representing that rate center.

## Session Timers

Use the commands in this section to provision session timers on the Cisco BTS 10200 Softswitch.

**Note** To configure SIP protocol and session timers in Release 4.5.x, you must use the new sip-timer-profile table. For customers upgrading to Release 4.5.x: SIP session timer values configured in the ca-config table prior to Release 4.5.x are reset to the default values after upgrading to Release 4.5.x. In Release 4.5.x and later, the session timer values are provisioned through the MIN-SE and SESSION-EXPIRES-DELTA-SECS tokens in the sip-timer-profile table. The id of the sip-timer-profile table record is then specified as the Value for the ca-config record of Type=sip_timer_profile_id. The id of the sip-timer-profile table can also be associated with a softsw-tg-profile record for SIP trunks. If you provision the timer values for a specific trunk, that overrides the ca-config default.

**Step 1** Adjust the session timer values in the sip-timer-profile table if necessary.

**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 on the applicable softswitch trunk group profile, and assign the sip-timer-profile-id:

```
change softsw_tg_profile id=<profile_id>; session_timer_allowed=Y;
sip_timer_profile_id=<timer_profile_id>;
```

**Step 3** For a switch-wide default for SIP trunks (if the trunk is not specifically provisioned), add a default sip-timer-profile-id to the ca-config table as follows:

```
add ca_config type=sip_timer_profile_id; datatype=string; value=<sip_timer_profile_id>;
```

## SIP Timer Values

Release 4.5.x enhances support for customizing SIP timers through the new sip-timer-profile table. A record in this table can be configured to apply to one or more SIP trunks or to apply switch-wide. A sip-timer-profile record can be associated with a specific softsw-tg-profile record and/or to a ca-config record. On a fresh software installation, and after a upgrade, the system operates with default SIP protocol timer values, as specified by the SIP specification. These default values are adequate for many installations. If customization is required, then a sip-timer-profile table can be provisioned and associated with all calls, or with calls on specific trunks.

**Note** The session timer parameters, MIN-SE and SESSION-EXPIRES-DELTA-SECS, have been consolidated into this new table and, unlike prior releases, are no longer configurable directly on the ca-config table.

**Note** To configure SIP protocol and session timers in Release 4.5.x, you must use the new sip-timer-profile table. For customers upgrading to Release 4.5.x: SIP timer values configured in the ca-config table prior to Release 4.5.x are reset to the default values after upgrading to Release 4.5.x. In Release 4.5.x and later, the timer values are provisioned in the sip-timer-profile table. The id of the sip-timer-profile table record is then specified as the Value for the ca-config record of Type=sip_timer_profile_id. The id of the sip-timer-profile table can also be associated with a softsw-tg-profile record for SIP trunks. If you provision the timer values for a specific trunk, that overrides the ca-config default.

**Step 1** Adjust the session 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** Enable session timers on the applicable softswitch trunk group profile, and assign the sip-timer-profile-id:

```
change softsw_tg_profile id=<profile_id>; session_timer_allowed=Y;
sip_timer_profile_id=<timer_profile_id>;
```

**Step 3** For a switch-wide default for SIP trunks (if the trunk is not specifically provisioned), add a default sip-timer-profile-id to the ca-config table as follows:

```
add ca_config type=sip_timer_profile_id; datatype=string; value=<sip_timer_profile_id>;
```

For a complete list of these timers, see the "SIP Timer Values" section on page 1-22.

# SIP-T, ISUP Version, ISUP-Transparency, and GTD

**Note** The values used in this section are examples. For a complete list of options, see the applicable table in the *Cisco BTS 10200 Softswitch Command Line Reference Guide.*

**Step 1** Provision a SIP-T trunk by setting the protocol type to SIP-T in the SIP trunk profile <profile_id> as follows.

**Note** Setting PROTOCOL_TYPE=SIP_T enables both SIP-T and SIP-GTD protocols.

**a.** If you want to review the valid SIP-T ISUP versions, enter the following command:

```
show sipt-isup-ver-base
```

**b.** For a SIP-T version of ANSI GR-317, provision as follows:

```
add softsw_tg_profile ID=<profile_id>; PROTOCOL_TYPE=SIP_T; PRACK_FLAG=Y;
SIPT_ISUP_VER=ANSI_GR317;
```

c. For a SIP-T version of GTD, provision as follows:

```
add softsw_tg_profile ID=<profile_id>; PROTOCOL_TYPE=SIP_T; PRACK_FLAG=Y;
SIPT_ISUP_VER=GTD; gtd_mode=<COMPACT or VERBOSE>; gtd_parms=ALL;
```

**Note** The version field (SIPT_ISUP_VER) is a user-provisioned alphanumeric in the SIP trunk profile required for SIP-T trunk types. The label represents the version of the ISUP as it is understood by the remote SIP-T entity for interworking. It is one of the following values: GTD, ANSI_GR317, or Q761_HONGKONG. If the remote SIP entity is looking for these ISUP versions but under a different name, the SIPT-ISUP-VER-ALIAS table can be used to provide a custom version name in the SIP message.
If it is desired to omit the base parameter from the SIP message (as defined in RFC 3204) for the ISUP version provisioned, the USE_SIPT_ISUP_BASE flag can be set to FALSE. It is TRUE by default.
The flag for controlling reliable provisionable responses (PRACK_FLAG) must be enabled.

**Step 2** Add a SIP trunk group associating it to the SIP trunk profile above as follows. The following example uses the dial plan identifier ‘dp,’ and the fully qualified domain name of the remote SIP-T entity ‘siptentity:5060.’

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

**Step 3** If you are using GTD, perform these additional substeps.

a. verify that the gtd-supp token in the call-agent-profile is set to Y, or set it to Y if not already done:

```
show call-agent-profile id=CA-146;
change call-agent-profile id=CA146; gtd-supp=Y;
```

b. If you are using GTD, enter the GTD parameter values, for example:

```
add gtd-parm-values id=ACL; description=Automatic Congestion Level;
```

**Note** GTD parameters can be used to support ISUP transparency between the Cisco BTS 10200 Softswitch and the Cisco PSTN Gateway (PGW) 2200. For more information on provisioning this feature, see the “ISUP Transparency on the BTS-PGW Interface” section in the *Cisco BTS 10200 Softswitch Provisioning Guide*. For a description of this feature, see the “ISUP Transparency with the Cisco PGW 2200” section in the *Cisco BTS 10200 Softswitch System Description*.

# DTMF SIP Signaling

The following command controls the DTMF SIP signaling feature on all SIP trunks associated to the SIP trunk profile <profile_id>.

**Step 1** Disable the DTMF SIP signaling feature.

The default is:

```
change softsw_tg_profile id=<profile_id>; DTMF_RELAY_METHOD=NONE;
```

**Step 2** Enable the DTMF SIP signaling feature.

Use the SIP INFO method to send unsolicited notification of telephone events (DTMF) toward the remote SIP entity provisioned in the trunk group:

```
change softsw_tg_profile id=<profile_id>; DTMF_RELAY_METHOD=INFO;
```

**Step 3** Enable the DTMF SIP signaling feature.

Use the SIP NOTIFY method to send solicited notification of telephone events (DTMF) toward the remote SIP entity provisioned in the trunk group. In this case, the remote SIP entity must subscribe to Cisco BTS 10200 for DTMF events:

```
change softsw_tg_profile id=<profile_id>; DTMF_RELAY_METHOD=NOTIFY;
```

# Asserted Identity and User-Level Privacy

The following command controls the p-asserted-id (PAI) header used to send and receive calling party information.

**Step 1** To set the system to derive calling party information exclusively from the PAI header on inbound calls, and always send for outbound calls, enter the command as follows:

```
change softsw_tg_profile id=<profile_id>; USE_PAI_HDR_FOR_ANI=Y;
```

**Step 2** To set the system to send or receive calling party information in the From: header, enter the command as follows. (This is the default setting.)

```
change softsw_tg_profile id=<profile_id>; USE_PAI_HDR_FOR_ANI=N;
```

The following command controls user-level privacy in the outbound SIP INVITE message.

**Step 1** To instruct the system to apply user-level privacy, enter the command as follows:

```
change softsw_tg_profile id=<profile_id>; APPLY-USER-PRIVACY=Y;
```

**Note** Setting this parameter to Y has the following effect—If the originator requested privacy, aspects of the calling party information (such as the calling name and number in the From: header) in the initial outbound SIP INVITE are hidden. Privacy is requested when either the calling party name or number have presentation restrictions.

**Step 2** To instruct the system to not apply user-level privacy, enter the command as follows. (This is the default setting.)

```
change softsw_tg_profile id=<profile_id>; APPLY-USER-PRIVACY=N;
```

**Note** A description of asserted identity and user-level privacy is provided in the *Cisco BTS 10200 Softswitch SIP Protocol User Guide*.

# ANI-Based Routing

The following rules apply when provisioning ANI-based routing for calls incoming on a SIP trunk:

- The softswitch trunk group on which the calls arrive must be have the "ANI_BASED_ROUTING" flag set to "Y."
- Office codes (NPA-NXX) must be provisioned for the calling party numbers.
- DN2Subscriber table must have the range of calling party numbers provisioned in it.
- A subscriber must be provisioned for a given range of DNs provisioned in DN2Subscriber. This subscriber's dial-plan and POP is then used to make call-type and routing decisions.

***Example 2-1 Example of ANI-Based Routing CLI***

```
add softsw-tg-profile ID=SS_PROFILE; PROTOCOL_TYPE=SIP;

add trunk-grp ID=157; CALL_AGENT_ID=CA146; TG_TYPE=SOFTSW;
SOFTSW_TSAP_ADDR=domainname.com; TG_PROFILE_ID=SS_PROFILE; POP_ID=1;
DIAL_PLAN_ID=BASIC_DPP; ANI_BASED_ROUTING=Y;

add subscriber-profile ID=sub_profile; DIAL_PLAN_ID=BASIC_DPP; POP_ID=1;

add subscriber ID=sub5; CATEGORY=INDIVIDUAL; NAME=sub5; TGN_ID=157;
SUB_PROFILE_ID=sub_profile; TERM_TYPE=TG;

add office-code DIGIT_STRING=214-555; OFFICE_CODE_INDEX=1;

add dn2subscriber FROM-DN=214-555-1231; TO-DN=214-555-1233; SUB_ID=sub5;
```

# Calling Name Delivery on Terminating SIP Trunks

This section describes how to provision the Calling Name Delivery (CNAM) feature on a terminating SIP trunk on the Cisco BTS 10200. When enabled on a SIP trunk, a local subscriber originating a call out this SIP trunk will have the originator name in the SIP message.

In the following provisioning example, if subscriber 'sub1' calls 469-555-2222, it is routed out a SIP trunk. The CNAM feature is invoked and adds 'john doe' to the display name of outgoing SIP call. To associate CNAM to the trunk, CNAM is associated to a virtual subscriber, and the virtual subscriber is associated to the SIP trunk.

```
add softsw-tg-profile ID=SS_PROFILE; PROTOCOL_TYPE=SIP;

add trunk-grp ID=157; CALL_AGENT_ID=CA146; TG_TYPE=SOFTSW; SOFTSW_TSAP_ADDR=TsapAddr.com;
TG_PROFILE_ID=SS_PROFILE; POP_ID=1; DIAL_PLAN_ID=BASIC_DPP; ANI_BASED_ROUTING=Y;

add subscriber-profile ID=sub_profile; DIAL_PLAN_ID=BASIC_DPP; POP_ID=1;

add subscriber ID=subcnam; CATEGORY=INDIVIDUAL; NAME=subcnam; TGN_ID=157;
SUB_PROFILE_ID=sub_profile; TERM_TYPE=TG; DN1=469-555-2222;
```

```
add feature FNAME=CNAM; TDP1=FACILITY_SELECTED_AND_AVAILABLE;
TID1=TERMINATION_RESOURCE_AVAILABLE; TTYPE1=R; FEATURE_SERVER_ID=FSPTC235;
DESCRIPTION=Calling Name; GRP_FEATURE=N

add service ID=3; FNAME1=CNAM;

add subscriber-service-profile SUB_ID=subcnam; SERVICE_ID=3;

change subscriber id=sub1; NAME=john doe;
```

# Trunk Group Audit for the SIP Adapter

The Trunk Group audit mechanism verifies the operational status of a trunk on a periodic basis. The mechanism is also triggered if communication problems are detected on the trunk.

When provisioning the Trunk Group audit mechanism, Cisco recommends provisioning only the STATUS-MONITORING flag in the Trunk Group table record.

The following fields should be left at the default settings:

In the SOFTSW_TG_PROFILE table:

- AUDIT-THRESHOLD

In the CA_CONFIG table:

- TRUNK-AUDIT-INTERVAL

# T.38 Fax Relay Call Agent Controlled Mode Across SIP Trunk Interface

The Cisco BTS 10200 Softswitch SIP interface always allows switching to T.38 fax when an incoming fax is detected from the SIP network. For additional guidance on interworking of SIP with other protocols for the T.38 fax features, see the T.38 fax relay information in the *Cisco BTS 10200 Softswitch SIP Protocol User Guide*.

CHAPTER 3

# Provisioning Voice Mail

**Revised: October 21, 2008, OL-5351-10**

The following example provisions a SIP trunk to a voice mail server located at 'vm.domainname.com:5060.' Typically, local subscriber dial plans have a route defined to this trunk when issuing forwarding calls to the voice mail server.

**Note** For a description of SIP VM features, see the "Voice-Mail Support" section in the *Cisco BTS 10200 Softswitch SIP PRotocol Provisioning Guide*. For general VM provisioning details, see the VM provisioning section in the *Cisco BTS 10200 Softswitch Provisioning Guide*.

## Provisioning Access to the Voice Mail Server

**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 Cisco BTS 10200 to forward incoming calls to the VM server, and (3) by the VM server to notify the Cisco 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 associated with the SIP trunk group. The value of dn1 is the DN that a subscriber can call to access the VM server.

```
add subscriber id=VMPilot; category=PBX; dn1=469-555-1001; tgn-id=80032;
sub-profile-id=sp1; term-type=TG;
```

**Step 6** If your voice-mail server does not support FQDN hostnames, a serving-domain-name record must be provisioned in the Cisco BTS 10200 Softswitch using the IP addresses resolved from the sia-xxxCAnnn.domain address. Otherwise, the VMWI status from SIP voice-mail platforms fails authentication with the Cisco BTS 10200 Softswitch.

**Note** This step is not necessary if your voice-mail server supports FQDN hostnames.

The address, sia-xxxCAnnn.domain, consists of the following parts:

- sia- is a required field
- xxx = site ID
- CAnnn = CA ID, such as CA146
- domain = a FQDN such as cisco.area777.com

**a.** Determine the two IP addresses associated with the sia-xxxCAnnn.domain address. These are available in the NIDS DNS table that was supplied with your system. You can also query the system for these two IP addresses through the **nslookup** command on the EMS host machine.

**b.** Add the two sia-xxxCAnnn.domain IP addresses to the serving-domain-name table:

```
add serving-domain-name domain-name=10.10.10.14; auth-reqd=n;
add serving-domain-name domain-name=10.10.11.14; auth-reqd=n;
```

# Provisioning Centrex Voice Mail

The following examples show commands for provisioning Centrex voice mail. Before you perform the following steps you must already have a Centrex group provisioned on your system. See the procedure in the "Provisioning a 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 Cisco BTS 10200 to forward incoming calls to the VM server, and (3) by the VM server to notify the Cisco 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=469-555-4444;
DN1=469-555-4444; 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=54444; CAT-CODE=1; SUB-ID=vmctxg1;
```

**Step 8** If your voice-mail server does not support FQDN hostnames, a serving-domain-name record must be provisioned in the Cisco BTS 10200 Softswitch using the IP addresses resolved from the sia-xxxCAnnn.domain address. Otherwise, the VMWI status from SIP voice-mail platforms fails authentication with the Cisco BTS 10200 Softswitch. The details for this step are provided in Step 6 of the "Provisioning Access to the Voice Mail Server" section on page 3-1,

# Provisioning Voice Mail Across Multiple Centrex Groups

The following provisioning steps illustrate how to provide voice mail service for Cisco BTS 10200 Centrex subscribers across multiple Centrex groups. The following information is required at the time of provisioning:

- Associate a unique business group identifier for each centrex group. For example:'bg1', 'bg2' and 'bg3', for three centrex groups.
- Identify the fully qualified domain name and port of the voice mail server used for SIP message exchange. For example:'vmserver:5060'.
- 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 this dial plan is 'dp'.

**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:

```
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;
```

Centrex group routing and dial plan tables are provisioned (not shown) 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.

APPENDIX A

# Sample Configuration Files for SIP Phones

**Revised: October 21, 2008, OL-5351-10**

For information about dial plans for other countries, contact Cisco support.

**Note** The provisioning in these examples is based on Cisco BTS 10200 Release 4.5.x.

## China Dial Plan

### Using a Cisco 7960 SIP Phone

A file (xxxx.xml) with the following dial_plan must be stored in the same directory as the SIP<mac>.cnf to support the China dialplan.

Add the dial_template: “china_dialplan.xml.”

```
china_dialplan.xml
<DIALTEMPLATE>
     <TEMPLATE MATCH="11."        Route="Default"    Timeout="0"    User="Phone"/> <!-- ->
    <TEMPLATE MATCH="........" Route="Default"    Timeout="0" User="Phone"/> <!-- -->
    <TEMPLATE MATCH="01.11."    Route="Default"    Timeout="0"    User="Phone"/>  <!-- -->
    <TEMPLATE MATCH="02.11."    Route="Default"    Timeout="0"    User="Phone"/>  <!-- -->
    <TEMPLATE MATCH="0...11."   Route="Default"    Timeout="0"    User="Phone"/> <!-- -->
    <TEMPLATE MATCH="00......*"  Route="Default"    Timeout="2"    User="Phone"/> <!-- -->
    <TEMPLATE MATCH="01........." Route="Default"    Timeout="0" User="Phone"/> <!-- -->
    <TEMPLATE MATCH="02........." Route="Default"    Timeout="0" User="Phone"/> <!-- -->
    <TEMPLATE MATCH="\*..\**"   Route="Default"    Timeout="2"    User="Phone"/>    <!--
-->
     <TEMPLATE MATCH="#..\**"    Route="Default"    Timeout="2"    User="Phone"/> <!-- -->
    <TEMPLATE MATCH="\*..#*"    Route="Default"    Timeout="0"    User="Phone"/> <!--  -->
    <TEMPLATE MATCH="#..#"       Route="Default"    Timeout="0"    User="Phone"/> <!-- -->
     <TEMPLATE MATCH="179.*"   Route="Default"    Timeout="2"    User="Phone"/> <!--  -->
     <TEMPLATE MATCH="............." Route="Default" Timeout="2" User="Phone"/> <!--  -->
     <TEMPLATE MATCH="*"      Route="Default"    Timeout="2"    User="Phone"/> <!--  -->
</DIALTEMPLATE>
```

### Using a Cisco 7905 SIP Phone

**Step 1** Add the following line to ld<macaddr>.txt file.

**Step 2** Convert it to binary by using cfgfmt.exe and upload the file to the TFTP server.

```
dial_plan:
11.|........|01.11.|02.11.|0...11.|00.....St2-|01.........|02.........|0..........|*..*St2
-|#..#|*#..*St2-|#..*St2-|179.St2-|......................
```

## North America Dial Plan

### Using a Cisco 7960 SIP Phone

Refer to the Cisco SIP IP Phone 7940/7960 Administrator Guide, Version 4.0.

### Using a Cisco 7905 SIP Phone

For further details refer to the Cisco IP Phone 7905 Series Administration Guide.

## Cisco IP Phone 7960 Sample Configuration File

```
##########################################################

SIPDefault.cnf file
##########################################################

# Image Version
image_version: "P0S3-WF-X-20"

# Proxy Server
proxy1_address: "10.89.224.18"
proxy2_address: ""
proxy3_address: ""
proxy4_address: ""
proxy5_address: ""
proxy6_address: ""

# Proxy Server Port (default - 5060)
proxy1_port:"5060"
proxy2_port:""
proxy3_port:""
proxy4_port:""
proxy5_port:""
proxy6_port:""

# Emergency Proxy info
proxy_emergency: "10.89.224.18"
proxy_emergency_port: "5060"

# Backup Proxy info
proxy_backup: "10.89.224.18"
proxy_backup_port: "5060"
```

```
# Proxy Registration (0-disable (default), 1-enable)
proxy_register: "1"

# Phone Registration Expiration [1-3932100 sec] (Default - 3600)
timer_register_expires: "3600"

# Codec for media stream (g711ulaw (default), g711alaw, g729)
preferred_codec: "g711ulaw"

# TOS bits in media stream [0-5] (Default - 5)
tos_media: "5"

# Enable VAD (0-disable (default), 1-enable)
enable_vad: "0"

# Inband DTMF Settings (0-disable, 1-enable (default))
dtmf_inband: "1"

# Out of band DTMF Settings (none-disable, avt-avt enable (default), avt_always - always
avt)
dtmf_outofband: "avt"

# DTMF dB Level Settings (1-6dB down, 2-3db down, 3-nominal (default), 4-3db up, 5-6dB up)
dtmf_db_level: "3"

# SIP Timers
timer_t1: "500"                  ; Default 500 msec
timer_t2: "4000"                 ; Default 4 sec
sip_retx: "10"                    ; Default 11
sip_invite_retx: "6"              ; Default 7
timer_invite_expires: "180"       ; Default 180 sec

# Setting for Message speeddial to UOne box
messages_uri: "9195551212"
#*********  Release 2 new config parameters **********

# TFTP Phone Specific Configuration File Directory
tftp_cfg_dir: "./sip_phone/"

# Time Server
sntp_mode: "directedbroadcast"
sntp_server: "171.68.10.150"
time_zone: "EST"
dst_offset: "+1"
dst_start_month: "April"
dst_start_day: ""
dst_start_day_of_week: "Sunday"
dst_start_week_of_month: "1"
dst_start_time: "02/00"
dst_stop_month: "Oct"
dst_stop_day: ""
dst_stop_day_of_week: "Sunday"
dst_stop_week_of_month: "8"
dst_stop_time: "02/00"
dst_auto_adjust: "1"

# Do Not Disturb Control (0-off, 1-on, 2-off with no user control, 3-on with no user
control)
dnd_control: "0"                 ; Default 0 (Do Not Disturb feature is off)

# Caller ID Blocking (0-disabled, 1-enabled, 2-disabled no user control, 3-enabled no user
con
trol)
callerid_blocking: "0"           ; Default 0 (Disable sending all calls as anonymous)
```

```
# Anonymous Call Blocking (0-disabled, 1-enabled, 2-disabled no user control, 3-enabled no
use
r control)
anonymous_call_block: "0"         ; Default 0 (Disable blocking of anonymous calls)

# DTMF AVT Payload (Dynamic payload range for AVT tones - 96-127)
dtmf_avt_payload: "101"           ; Default 100

# XML file that specifies the dialplan desired
dial_template: "dialplan"

# Network Media Type (auto, full100, full10, half100, half10)
network_media_type: "auto"

#Autocompletion During Dial (0-off, 1-on [default])
autocomplete: "1"

#Time Format (0-12hr, 1-24hr [default])
time_format_24hr: "0"

# Services URL points to XML on Stan-btc
services_url: "http://64.101.150.57/servlet/FeatureProvisioning"
# services_url: "http://64.101.150.57/servlet/CallForwarding"

# Enable telnet debugging
telnet_level: 2


##########################################################



##########################################################
# SIP0008ABC456.cnf
##########################################################

# SIP Confiuration Generic File (start)

# Line 1 Settings
line1_name: "9022551232"                     ; Line 1 Extension\User ID
line1_displayname: "SIP8"           ; Line 1 Display Name
line1_authname: "UNPROVISIONED"         ; Line 1 Registration Authentication
line1_password: "UNPROVISIONED"         ; Line 1 Registration Password
proxy1_address: 10.89.224.63
proxy1_port: 5060

# Line 2 Settings
line2_name: ""                    ; Line 2 Extension\User ID
line2_displayname: ""                ; Line 2 Display Name
line2_authname: "UNPROVISIONED"         ; Line 2 Registration Authentication
line2_password: "UNPROVISIONED"         ; Line 2 Registration Password

# Line 3 Settings
line3_name: ""                          ; Line 3 Extension\User ID
line3_displayname: ""                   ; Line 3 Display Name
line3_authname: "UNPROVISIONED"         ; Line 3 Registration Authentication
line3_password: "UNPROVISIONED"         ; Line 3 Registration Password

# Line 4 Settings
line4_name: ""                          ; Line 4 Extension\User ID
line4_displayname: ""                   ; Line 4 Display Name
line4_authname: "UNPROVISIONED"         ; Line 4 Registration Authentication
line4_password: "UNPROVISIONED"         ; Line 4 Registration Password
```

```
# Line 5 Settings
line5_name: ""                          ; Line 5 Extension\User ID
line5_displayname: ""                   ; Line 5 Display Name
line5_authname: "UNPROVISIONED"         ; Line 5 Registration Authentication
line5_password: "UNPROVISIONED"         ; Line 5 Registration Password

# Line 6 Settings
line6_name: ""                          ; Line 6 Extension\User ID
line6_displayname: ""                   ; Line 6 Display Name
line6_authname: "UNPROVISIONED"         ; Line 6 Registration Authentication
line6_password: "UNPROVISIONED"         ; Line 6 Registration Password

# Phone Label (Text desired to be displayed in upper right corner)
phone_label: "SIP Phone 8"            ; Has no effect on SIP messaging

# Time Zone phone will reside in
time_zone: CST

# XML file that specifies the dialplan desired
dial_template: "dialplan"
# SIP Configuration Generic File (stop)

##########################################################
```

# Cisco IP Phone 7905 Sample Configuration File

```
ld0008a3d31e4a.txt

#tx
upgradecode:3,0x501,0x0400,0x0100,2.3.4.5,69,0x030218A,LD0101SIP030218A.zup
UIPassword:Cisco
dhcp:1
Proxy:1.2.3.4
UID:4692557907
PWD:user
LoginID:user
UseLoginID:1
SIPRegOn:1
RxCodec:2
TxCodec:2
Timezone:20
DNS1IP:0.0.0.0
UseTftp:1
```

# GLOSSARY

Revised: October 21, 2008, OL-5351-10

## A

| | |
|---|---|
| **AC** | automatic callback |
| **AC_ACT** | automatic callback activation |
| **AC_DEACT** | automatic callback deactivation |
| **ACR** | anonymous call rejection |
| **ACR_ACT** | anonymous call rejection activation |
| **ACR_DEACT** | anonymous call rejection deactivation |
| **AI** | asserted identity |
| **ANI** | automatic number identification |
| **AOR** | address of record |
| **AOR2SUB** | Address of Record to Subscriber (refers to new table) |
| **AR** | automatic recall |
| **AR_ACT** | automatic recall activation |
| **AR_DEACT** | automatic recall deactivation |
| **ATA** | analog telephone adaptor |
| **AUTH-REALM** | Authentication Realm (refers to new table) |

## B

| | |
|---|---|
| **BCM** | Basic Call module |
| **BGID** | Business Group Identity |
| **BLV** | Busy Line Verification |
| **B911** | Basic 911 |

## C

| | |
|---|---|
| **CA** | Call Agent |
| **CA-CONFIG** | Call Agent configuration (refer to the SIP Adaptor Configuration Parameters table) |
| **CALEA** | Communications Assistance for Law Enforcement Act |
| **CAS** | channel-associated signaling |
| **CAT** | customer access treatment |
| **CBLK** | call block (reject caller) |
| **CCW** | cancel call waiting |
| **CDP** | custom dial plan |
| **CDR** | call detail record |
| **CF** | call forwarding |
| **CFB** | call forwarding on busy |
| **CFBI** | call forwarding on busy interrogation |
| **CFBVA** | call forwarding on busy variable activation |
| **CFBVD** | call forwarding on busy variable deactivation |
| **CFNA** | call forwarding on no answer |
| **CFNAI** | call forwarding on no answer interrogation |
| **CFNAVA** | call forwarding on no answer variable activation |
| **CFNAVD** | call forwarding on no answer variable deactivation |
| **CFU** | call forwarding unconditional |
| **CFUA** | call forwarding unconditional activation |
| **CFUD** | call forwarding unconditional deactivation |
| **CFUI** | call forwarding unconditional interrogation |
| **CHD** | call hold |
| **CIC** | circuit identification code, carrier identification code |
| **CID** | calling identity delivery, also caller ID (see also CND) |
| **CIDB** | calling identity delivery blocking |
| **CIDCW** | calling identity delivery on call waiting |

| | |
|---|---|
| **CIDS** | calling identity delivery and suppression (per call) |
| **CIDSD** | calling identity delivery and suppression (per call)—delivery part |
| **CIDSS** | calling identity delivery and suppression (per call)—suppression part |
| **CLASS** | custom local area signaling services |
| **CLI** | command-line interface |
| **CMSS** | Call Management System Signaling |
| **CNAB** | calling name delivery blocking |
| **CNAM** | calling name delivery |
| **CND** | calling number delivery, calling number display |
| **CNDB** | calling number delivery blocking |
| **CODEC** | coder/decoder, compression/decompression |
| **COS** | class of service |
| **COT** | customer-originated trace, continuity testing, central office termination |
| **CPT** | called party termination |
| **CPRK** | call park |
| **CPRK_RET** | call park retrieve |
| **CT** | call transfer, call type |
| **CW** | call waiting |
| **CWD** | call waiting deluxe |
| **CWDA** | call waiting deluxe activation |
| **CWDD** | call waiting deluxe deactivation |
| **CWDI** | call waiting deluxe interrogation |
| **CWI** | call waiting indication |

## D

| | |
|---|---|
| **DACWI** | distinctive alerting call waiting indication |
| **DPN** | directed call pickup without barge-in |
| **DPU** | directed call pickup with barge-in |

| | |
|---|---|
| **DID** | direct inward dialing |
| **DN** | directory number |
| **DND** | do not disturb |
| **DND_ACT** | do not disturb activation |
| **DND_DEACT** | do not disturb deactivation |
| **DNS** | domain name server |
| **DNS SRV** | domain name server services |
| **DOD** | direct outward dialing |
| **DP** | dial plan |
| **DPN** | directed call pickup without barge-in |
| **DPN_O** | directed call pickup without barge-in (originate) |
| **DPN_T** | directed call pickup without barge-in (terminate) |
| **DPU** | directed call pick-up with barge-in |
| **DPU_O** | directed call pickup with barge-in (originate) |
| **DPU_T** | directed call pickup with barge-in (terminate) |
| **DRCW** | distinctive ringing/call waiting |
| **DRCW_ACT** | distinctive ringing/call waiting activation |
| **DTMF** | dual tone multifrequency |

## E

| | |
|---|---|
| **E.164** | Telephone number standard of ITU |
| **E911** | Enhanced 911 |
| **EMS** | Element Management System |

## F

| | |
|---|---|
| **FDT** | Final Stage Dial Tone |
| **FS** | Feature Server |
| **FQDN** | fully qualified domain name |

## G

| | |
|---|---|
| **GAP** | generic address parameter |
| **GUI** | graphical user interface |
| **GUI FS** | graphical user interface feature server |

## H

| | |
|---|---|
| **H.323** | ITU-T recommendation adopted by the VoIP Forum as the call signaling protocol over LAN |
| **HOTLINE** | hotline |
| **HTML** | HyperText Markup Language |
| **HTTP** | Hypertext Transfer Protocol |

## I

| | |
|---|---|
| **IETF** | Internet Engineering Task Force |
| **INS** | in service |
| **IP** | Internet Protocol |
| **ISDN** | Integrated Services Digital Network |
| **ISFG** | Incoming simulated facility group |
| **ISUP** | ISDN user part |
| **ITP** | IP transfer point |
| **IVR** | interactive voice response |

## J

## K

## L

| | |
|---|---|
| **LATA** | local access and transport area |

| | |
|---|---|
| **LNP** | local number portability |
| **LSSGR** | LATA Switching Systems Generic Requirements |

## M

| | |
|---|---|
| **MAC2SUB** | MAC to Subscriber (refers to new table) |
| **MDN** | multiple directory numbers |
| **MF** | multifrequency |
| **MG (MGW)** | media gateway |
| **MGCP** | Media Gateway Control Protocol |
| **MGW (MG)** | media gateway |
| **MLHG** | multiline hunt group |
| **MWI** | message waiting indicator |

## N

| | |
|---|---|
| **NAPTR** | Naming Authority Pointer |
| **NP** | number portability |
| **NPDI** | number portability dip indication |

## O

| | |
|---|---|
| **OAM** | operations, administration, and maintenance, Operations administration module |
| **OCB** | outgoing call barring |
| **OCBA** | outgoing call barring activation |
| **OCBD** | outgoing call barring deactivation |
| **OCBI** | outgoing call barring interrogation |
| **OOS** | out of service |
| **OSFG** | outgoing simulated facility group |

## P

| | |
|---|---|
| **POTS** | plain old telephone service |
| **PRACK** | provisional response acknowledgement |
| **PRI** | primary rate interface |
| **PSTN** | public switched telephone network |

## Q

| | |
|---|---|
| **QoS** | quality of service |

## R

| | |
|---|---|
| **RACF** | remote activation of call forwarding |
| **RACF-PIN** | remote activation of call forwarding personal ID number |
| **RFC** | Request for Comment (IETF) |
| **RQNT** | request for notification |
| **RN** | routing number |

## S

| | |
|---|---|
| **SC1D** | speed call 1-digit |
| **SC1D_ACT** | speed call 2-digit activation |
| **SC2D** | speed call 1-digit |
| **SC2D_ACT** | speed call 2-digit activation |
| **SCA** | selective call acceptance |
| **SCA_ACT** | selective call acceptance activation |
| **SCF** | selective call forwarding |
| **SCF_ACT** | selective call forwarding activation |
| **SCR** | selective call rejection |
| **SCR_ACT** | selective call rejection activation |
| **SDP** | Session Description Protocol |

| | |
|---|---|
| **SIA** | SIP adapter |
| **single-stage digit collection** | Used with SIP subscriber features. Refers to when there is no tone provided for SIP users to prompt for forwarding digits. The SIP users enter the forwarding digits immediately after the VSC. |
| **SIP** | Session Initiation Protocol |
| **SIP-T** | SIP for telephones |
| **SLE** | screening list editing |
| **SP** | service provider |
| **SPCS** | stored program control system |
| **SRV** | server resource records |
| **SS7** | Signaling System 7 |

## T

| | |
|---|---|
| **TCP** | Transmission Control Protocol |
| **TF** | toll free |
| **TG** | trunk group |
| **TGID** | trunk group identity |
| **TGW** | trunking gateway |
| **ToS** | type of service |
| **TSAP** | Transport Service Access Point |
| **TWC** | three-way calling |
| **TWCD** | three-way calling deluxe |

## U

**UAC** user agent client

**UAS** user agent server

**UDP** User Datagram Protocol

**UI** user interface

**URI** uniform resource identifier

**URL** universal resource locator

**USER-AUTH** User Authentication (refers to new table)

**USTWC** usage-sensitive three-way calling

## V

**VM** voice mail

**VoIP** voice over IP

**VSC** vertical service code

## W

**WARMLINE** warmline

**WFI** waiting for instruction

## X

***x*DSL** (generic) digital subscriber line

## Y

## Z