Cisco BTS 10200 Softswitch SIP Protocol Provisioning Guide

Release 4.1

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Preface

The *Cisco BTS 10200 Softswitch SIP Protocol Provisioning Guide* details support for Session Initiation Protocol (SIP) devices and trunks in Release 4.1. 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.

Organization

This document is organized as follows:

- Chapter 1, "Provisioning SIP Devices"
- Chapter 2, "Provisioning SIP Trunks"
- Chapter 3, "Provisioning Voice-Mail"
- Appendix A, "Sample Configuration Files"
- 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 4.1 SIP Protocol Support feature module should be used in conjunction with the previous Cisco BTS 10200 Softswitch documents:

- *Cisco BTS 10200 Softswitch Release Notes for Release 4.1*
- *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 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*

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If you have a priority level 3 (P3) or priority level 4 (P4) problem, contact TAC by going to the TAC website:

http://www.cisco.com/tac

P3 and P4 level problems are defined as follows:

- P3—Your network performance is degraded. Network functionality is noticeably impaired, but most business operations continue.
- P4—You need information or assistance on Cisco product capabilities, product installation, or basic product configuration.

In each of the above cases, use the Cisco TAC website to quickly find answers to your questions.

To register for Cisco.com, go to the following website:

http://www.cisco.com/register/

If you cannot resolve your technical issue by using the TAC online resources, Cisco.com registered users can open a case online by using the TAC Case Open tool at the following website:

http://www.cisco.com/tac/caseopen

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http://www.cisco.com/warp/public/687/Directory/DirTAC.shtml

P1 and P2 level problems are defined as follows:

- P1—Your production network is down, causing a critical impact to business operations if service is not restored quickly. No workaround is available.
- P2—Your production network is severely degraded, affecting significant aspects of your business operations. No workaround is available.

CHAPTER 1

Provisioning SIP Devices

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

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 4.1 SIP Protocol Support 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 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 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 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 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 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 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 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 add AUTH_REALM_ID=ciscolab;	NA	NA	NA
Serving Domain Name 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: sia-domainname.com proxy1_port : 5060
Subscriber – DN 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 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) UseLoginID: 1 (To use login ID for authentication. LoginID is used if authenticate user information is different from UID.) PWD:cisco	LoginID: SIP_7940_ONE (same as 7960) UseLoginID: 1 (To use login ID for authentication. LoginID is used if authenticate user information is different from UID.) PWD:cisco	line1_authname:“SIP_7940_ONE” line1_password:“cisco”

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

Services Key: Enabling Feature Activation or Deactivation

This feature can only be used with phones that support HTM services using softkey. The Cisco 7960 is an example of such a phone.

Install Mini Browser Adaptor

SIP phones interface via the IP network with the Mini Browser Adaptor (MBA) for services. The user accesses service functions via the "Services" key on the SIP phone. A GUI on the SIP phone allows users to self-provision certain features. The MBA supports these services, and performs GUI management for the GUI-enabled SIP-phone handsets.

The GUI feature server (GFS) in the Feature Server for POTS, Tandem, and Centrex features (FSPTC) is the feature server data access component for GUI management and is responsible for subscriber data access and northbound updates into the EMS.

Signaling between the MBA and SIP phones uses Cisco CMXML protocol over HTTP. Internal signaling between the MBA and the GFS is via GUI control protocol (GCP), which is an XML-based protocol over SCTP links.

The following steps allow to install the MBA.

Step 1 Locate **mba_install.sh** in the same location as the Cisco BTS 10200 "install.sh" script.

Step 2 Execute the script.

Step 3 Answer the script prompts (local and remote DNS names).

MBA automatically starts when the installation script is finished.

Step 4 Verify MBA execution by monitoring the MBA.log file located in /opt/ems/log.

GUI Feature Server Provisioning

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 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 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 11 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;
```

To use the CLI command combination, see Step 11.

Step 4 (Optional) Add AOR2SUB entry.

The Domain portion of the Host ID should be provisioned in the Server Domain table.

```
add aor2sub
aor_id=4695551111@domainname.com; sub_id= sip_sub1;
```

Step 5 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 6 If the device is not capable of registering itself, a static contact may be used.

```
add static_contact
static_contact_host=3.4.5.6;static_contact_port=5060; aor_id=4695551111@domainname.com;
user_type=phone;
```

Step 7 Add MAC2SUB.

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

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

Step 8 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 Adaptor 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 9 Set the AORID in the Subscriber table.

```
change subscriber id= sip_sub1;
aor_id=4695551111@domainname.com;
```

Step 10 Put AOR in Service.

```
control aor2sub
aor_id=4695551111@domainname.com; target-state=INS;
```

Step 11 Step 3 (Subscriber), Step 4 (AOR2SUB), Step 6 (STATIC_CONTACT), Step 7 (MAC2SUB), and Step 9 (Set AOR in SUB table) 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;
static_contact_host=3.4.5.6;static_contact_port=5060; user_type=phone; mac_id=
SIP0008A3D31E4A;
```

Provisioning Subscriber Features

This section describes how to provision Subscriber features. Existing features introduced prior to Release 4.1 are hyperlinked to the *Cisco BTS 10200 Softswitch Release 4.1 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 4.1 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 4.1, 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 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*.

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. 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.

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.

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

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, Release 4.1 also offers jointly-provided features. These are features provided jointly by the phone and by the Cisco BTS 10200. To use these features, you must provision both the phone and the Cisco BTS 10200.

Session Timer

Step 1 Change the softswitch trunk group profile ID.

```
softsw_tg_profile id=<profile_id>; SESSION_TIMER_ALLOWED=Y;
```

Step 2 Add the CA-CONFIG session-expires value.

```
add ca-config  type=session-expires;data-type=INT; value=3600;
```

Step 3 Add the CA-CONFIG min-se value.

```
add ca-config type=min-se;data-type=INT;value=900;
```

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

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.

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;
```

DNS SRV

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

Step 1 Disable DNS SRV.

The default is:

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

Step 2 Enable DNS SRV.

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

Note When a DNS SRV query fails and the target destination is resolved using an A-record query, the SIP trunk provisioning characteristics determine the transport used. Consult the TCP/UDP section for provisioning details for the transport options available.

The softsw_tsap_addr of the softswitch trunk group can now be provisioned as one of the following:

a. NAPTR name

b. SRV name

c. Host name

d. Host name and port

e. IP address

f. IP address and port

Step 3 If the dns_srv_supp field of the softsw_tg_profile table is set to NONE, NAPTR and SRV names are not supported.

The use of either NAPTR, SRV or Host names requires correctly configured DNS servers.

Provisioning recommendations:

1. 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.
2. 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.
3. Provision DNS entries so that the same TSAP address is used to perform A-record queries or SRV quires.

Note This can lead to problems when troubleshooting DNS configuration issues. If an A-record is set up for x.y.com and an SRV record is set up for _sip._udp.x.y.com, the system can successfully resolve the TSAP address x.y.com using either an SRV lookup or an A-record lookup. The system uses an SRV lookup if SRV support is turned on and an A-record lookup if SRV support is turned off. An investigator would likely verify the DNS configuration using nslookup to perform A-record queries, even if SRV queries were being used by the Cisco BTS 10200. Avoid confusion by setting up DNS entries so that the same TSAP address cannot be resolved by SRV queries and A record queries.

Type of Service

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=2;

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.

TCP/UDP

Use the following steps to provision a non-SRV transport.

Step 1 Ensure the TSAP address for the trunk group is either a host name or an IP address. NAPTR and SRV TSAP addresses are not supported on non-SRV trunk groups.

Step 2 Set the dns_srv_supp field in the softsw_tg_profile table to **NONE**.

Step 3 Set the non-srv-transport field of the softsw_tg_profile table to the desired transport. The options are TCP, UDP, UDP_ONLY.

If TCP is not supported at the remote end of the trunk group, use UDP_ONLY to prevent using TCP for large messages.

For SRV trunk groups, the transport protocol is determined by the DNS configuration.

Refer to the SRV feature provisioning for details on provisioning SRV trunk groups and important notes on performance impacts.

Reliable Provisional Responses

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

Step 1 The default for making reliable provisional responses not required for 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 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 from 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 be set to 'Y' if the PROTOCOL_TYPE of the SIP trunk profile is set to 'SIP_T' or 'CMSS.'

Diversion Indication

The following commands control the diversion feature 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;
```

and:

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

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

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

or:

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

Note Note that DIVERSION_HEADER_SUPP and CC_DIVERSION_SUPP cannot both be set to 'Y.'

These flags do not apply to incoming calls. If the diversion headers exist, the information from them is interpreted automatically.

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. Refer to the *Cisco BTS 10200 Softswitch SIP Protocol User Guide* for more details.

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 voice-mail sever:

```
add trunk_grp ID=<trk_grp_id1>; TG_TYPE=SOFTSW; TG_PROFILE_ID=<profile_id>;
SOFTSW_TSAP_ADDR=sipser ver: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 Timer

The following commands control the session timer feature on all SIP trunks associated to the SIP trunk profile <profile_id>.

Step 1 Disable session timer.

The default is:

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

Step 2 Enable session timer.

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

Step 3 The following command sets the session duration when the duration is controlled by Cisco BTS 10200 for a given call.

The value specified is in seconds with a range of 1800 to 7200.

```
add ca_config TYPE=SESSION_EXPIRES; DATATYPE=INTEGER; VALUE=7200;
```

Step 4 The following command sets the minimum tolerable session duration if the session duration is negotiated down by other SIP entities along the signaling path.

The value specified is in seconds with a range of 900 to 1800.

```
add ca_config TYPE=MIN_SE; DATATYPE=INTEGER; VALUE=1800;
```

SIP-T

Step 1 A SIP-T trunk is provisioned by setting the protocol type to SIP-T in the SIP trunk profile <profile_id> as follows:

```
add softsw_tg_profile ID=<profile_id>; PROTOCOL_TYPE=SIP_T; PRACK_FLAG=Y;
SIPT_ISUP_VER=gr317;
```

Step 2 Next, 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;
```

The version label (SIPT_ISUP_VER) is a user-provisioned alphanumeric in the SIP trunk profile required for SIP-T trunk types. The label represents the version name for gr317 ISUP as it is understood by the remote SIP-T entity for interworking. Usually, it is set to 'gr317.' The flag for controlling reliable provisionable responses (PRACK_FLAG) must be enabled.

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

The Asserted Identity feature is controlled by the Send CPN flag in the SIP trunk profile with the identifier <profile_id>. When disabled, asserted identity and privacy headers are not populated into the SIP INVITE message out the SIP trunk. In this case, the ANI information is populated into the From header.

Step 1 Disable the asserted Identity feature.

The default is:

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

Step 2 Enable the asserted identity feature.

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

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;
```

TCP/UDP

If the dns_srv_supp field in the SIP trunk profile <profile_id> is set to NONE, the transport for outgoing calls is selected based on the non-srv-transport field. Possible values for this field are TCP, UDP, UDP_ONLY. These values are set as follows:

`change softsw_tg_profile id=<profile_id>; NON_SRV_TRANSPORT=TCP;`
or:

`change softsw_tg_profile id=<profile_id>; NON_SRV_TRANSPORT=UDP;`
or:

`change softsw_tg_profile id=<profile_id>; NON_SRV_TRANSPORT=UDP_ONLY;`
If UDP is provisioned as the transport and a SIP message is greater than 1300 bytes, Cisco BTS 10200 attempts to send it over TCP. If this fails, the Cisco BTS 10200 falls back to UDP and attempts to send the message. UDP_ONLY is used to prevent this attempt if it is known that TCP is not supported at the destination.

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;
```

Provisioning Voice-Mail

The following example provisions a SIP trunk toward a voice mail server located at 'vm.domainname.com:5060.' In this example, local subscribers of the Cisco BTS10200 originating or forwarding a call to number 816-222-2001 will get voice mail service.

The SIP trunk has diversion enabled to allow the voice mail server determine the last forwarded party and select the target inbox. A dial plan 'tb1' is associated to the SIP trunk to allow the voice mail server to originate a call toward the Cisco BTS 10200. The voice mail may originate a call using a prefix of '216-351' as provisioned in the following example. This is used for voice mail features that involve originating calls toward the Cisco BTS 10200.

The provisioning sample commands show how to provision voice mail.

Note Provisioning of subscribers is not shown here.

Step 1 Add dial plan for calling back from VM trunk.

```
add dial-plan-profile id=tb16;nanp-dial-plan=y; description=north america local
add dial-plan id=tb16;digit-string=216-351;dest-id=tb16-local;min-digits=10;max-digits=10
add dial-plan id=tb16;digit-string=216-352;dest-id=tb16-local;min-digits=10;max-digits=10
```

Step 2 Add the Softswitch trunk group for voice-mail.

```
add softsw-tg-profile id=8003X;protocol-type=SIP; diversion_header_supp=Y;
voice_mail_trunk_grp=Y;
```

Step 3 Add the trunk group for VM (dial-plan determines who may be called back from VM).

```
add trunk-grp id=80032;softsw-tsap-addr=vm.domainname.com:5060;
call-agent-id=CA146;tg-type=softsw;tg-profile-id=8003X;dial-plan-id=tb16
```

Provisioning Non-Centrex Voice-Mail

Step 1 Add the destination ID.

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

Step 2 Add the dial plan.

```
add dial-plan
id=BASIC_DPP;digit-string=469-255;reqd-digits=10;dest-id=DESTLOC;
```

Step 3 Add the subscriber.

```
add subscriber
id=VMPilot;category=PBX;dn1=469-255-1001;tgn-id=80032;sub-profile-id=sp1; term-type=TG;
```

Provisioning Centrex Voice-Mail

The following examples show commands for provisioning Centrex voice-mail.

Step 1 Add the dial plan profile.

```
add dial-plan-profile id=cdp1;DESCRIPTION=centrex dial plan; NANP_DIAL_PLAN=y;
add dial-plan id=cdp1; digit-string=5555; min-digits=4; max-digits=10; dest-id=DESTLOC;
NOA=UNKNOWN;
```

Step 2 Add the Centrex trunk group profile for voice-mail.

```
add softsw_tg_profile id=SS_PRO24; protocol_type=SIP; trunk_sub_grp_type=BGID;
diversion_header_supp=Y; voice_mail_trunk_grp=Y;
```

Step 3 Add the voice-mail Centrex trunk group for voice-mail.

```
add
trunk-grp;id=24;trunk_sub_grp_type=bg1;softsw_tsap_addr=vm.cisco.com:5060;tg-type=softsw;t
g-profile-id=SS_PRO24;dial-plan-id=cdp1;CALL_AGENT_ID=CA146;
```

Step 4 Add the voice-mail extension as a Centrex subscriber.

```
add subscriber id=vmctxg1; CATEGORY=ctxg; NAME=vmctxg1; STATUS=ACTIVE; LANGUAGE=english;
BILLING-DN=469-255-5555; DN1=469-255-5555; PRIVACY=NONE; RING-TYPE-DN1=1;
SUB-PROFILE-ID=sp2; TERM-TYPE=TG; POLICY-ID=NULL;ctxg_id=ctxgsip1;tgn_id=24;
```

Step 5 Associate the voice-mail Centrex trunk group with a Centrex extension subscriber.

```
change trunk-grp; id=24;main_sub_id=vmctxg1;
```

Step 6 Map the voice-mail's Centrex extension to a subscriber.

```
add ext2subscriber CTXG-ID=ctxgsip1; EXT=55555; ASSIGNED=Y; CTX-RESTRICT=NONE; CAT-CODE=1;
SUB-ID=vmctxg1;
```

Voice-Mail over SIP: Cisco BTS 10200 Centrex Subscribers

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;
```

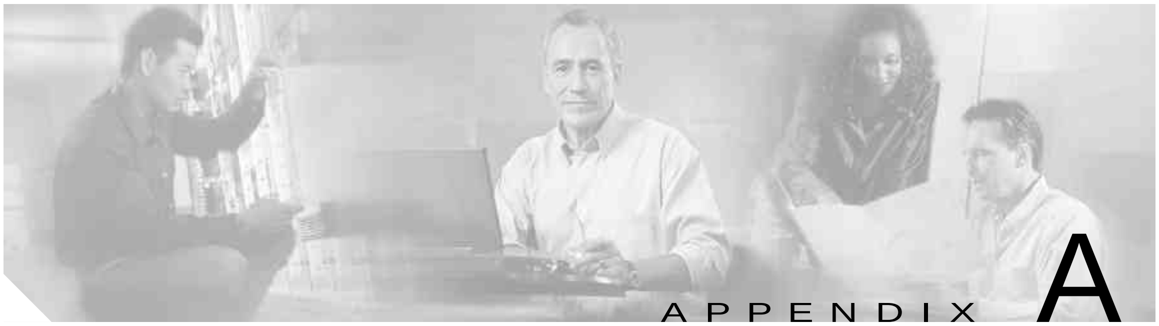
Step 2 Add a SIP trunk for each business group identifier. Each trunk points to the address of the voice mail sever:

```
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.



Sample Configuration Files

SIP Phone Market-Specific Configuration

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

Note The provisioning in these examples is based on Cisco BTS 10200 Release 4.1.

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

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