CHAPTER 1

Provisioning SIP Devices and SIP Subscribers

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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 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”

<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.
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.
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;</td><td>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.</td><td>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.</td><td>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.</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>#.-
Add the dial_template defined for regional settings as necessary.</td><td>DialPlan:*St4-|#St4-|911|1>#t8.r9t2-|0>#t811.rat4-|^1t4>#.-
Add the dial_template defined for regional settings as necessary.</td><td>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.</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 Default = 500	T1 timer (in milliseconds) for RTT estimate.
TIMER-T2-SECS	RANGE(1 – 10) seconds 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 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 DEFAULT = 0[1]	Timer A specifies the INVITE request retransmit interval (in milliseconds) for UDP only.
TIMER-B-SECS	RANGE(1 - 3600) seconds DEFAULT = 0*	Timer B specifies INVITE transaction timeout in seconds.
TIMER-D-SECS (0 for TCP)	RANGE(33 – 65) seconds DEFAULT = 33	Timer D (in seconds) specifies wait time for response retransmits.
TIMER-E-MILLI	RANGE(100 – 5000) ms DEFAULT = 0*	Timer E (in milliseconds) specifies non-INVITE request retransmit interval, UDP only.
TIMER-F-SECS	RANGE(1 - 3600) seconds 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 DEFAULT = 0*	Timer G (in milliseconds) specifies INVITE response retransmit interval.
TIMER-H-SECS	RANGE(1 - 3600) seconds DEFAULT = 0*	Timer H (in seconds) specifies Wait time for ACK receipt.
TIMER-I-SECS (0 for TCP)	RANGE(1 - 10) seconds DEFAULT = 0*	Timer I (in seconds) specifies Wait time for ACK retransmits.
TIMER-J-SECS (0 for TCP)	RANGE(1 – 3600) seconds DEFAULT = 0*	Timer J in seconds specifies Wait time for non-INVITE request.
INVITE-INCOMPLETE-TIMER-SECS	RANGE(15 – 600) seconds 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 DEFAULT = 900	This is minimum acceptable value of session-expires in seconds. **Note** This parameter is a session timer.
SESSION-EXPIRES-DELTA-SECS	RANGE(100 – 7200) seconds DEFAULT = 1800	This value (in seconds) is sent in the session-expires header. **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.