Configuring Non-Cisco SIP Phones

Cisco Unified CallManager, Release 5.0 supports Cisco SIP IP Phones as well as RFC3261-compliant SIP phones from third-party companies. This chapter describes how to configure the third-party SIP phones by using Cisco Unified CallManager Administration.

This chapter contains the following sections:

SIP Phone Configuration Differences

Table 4-1 provides a comparison overview of the configuration differences between Cisco SIP IP Phones and third-party SIP phones.

Table 4-1 SIP Phone Model Configuration Comparison

SIP Phone	Integrated with Centralized TFTP	Sends MAC Address	Downloads Softkey File	Downloads Dial Plan File	Supports Cisco Unified CallManager Failover and Fallback	Supports Reset and Restart
Cisco SIP IP Phone model 7911, 7941, 7961, 7970, 7971	Yes	Yes	Yes	Yes	Yes	Yes
Cisco SIP IP Phone model 7940, 7960	Yes	Yes	No	Yes	Yes	Yes
Cisco SIP IP Phone model 7905, 7912	Yes	Yes	No	No	Yes	Yes
Third-party SIP Phone	No	No	No	No	No	No

Use Cisco Unified CallManager Administration to configure third-party SIP phones (see the "Third-Party SIP Phone Configuration Checklist" section on page 4-4). The administrator must also perform configuration steps on the third-party SIP phone; see following examples:

- Ensure proxy address in the phone is the IP or Fully Qualified Domain Name (FQDN) of Cisco Unified CallManager
- Ensure directory number(s) in the phone match the directory number(s) that are configured for the device in Cisco Unified CallManager Administration
- Ensure digest user ID (sometimes referred to as Authorization ID) in the phone matches the Digest User ID in Cisco Unified CallManager Administration

Consult the documentation that came with the third-party SIP phone for more information.

How Cisco Unified CallManager Identifies a Third-Party Phone

Because third-party SIP phones do not send a MAC address, they must identify themselves by using digest authentication.

The REGISTER message includes the following header:

Authorization: Digest
username="swhite",realm="ccmsipline",nonce="GBauADss2qoWr6k9y3hGGVDAqnLfoLk5",uri="sip:172.18.197.224",algorithm=MD5,response="126c0643a4923359ab59d4f53494552e"

The username, swhite, must match an end user that is configured in the End User Configuration window of Cisco Unified CallManager Administration (see Adding an End User in the *Cisco Unified CallManager Administration Guide, Release 5.0(4)*). The administrator configures the SIP third-party phone with the user; for example, swhite, in the Digest User field of Phone Configuration window (see Configuring Cisco Unified IP Phones, in the *Cisco Unified CallManager Administration Guide, Release 5.0(4)*).

Note You can assign each end user ID to only one third-party phone (in the Digest User field of the Phone Configuration window). If the same end user ID is assigned as the Digest User for multiple phones, the third-party phones to which they are assigned will not successfully register.

Third-Party SIP Phones and TFTP

Third-party SIP phones do not get configured by using the Cisco Unified CallManager TFTP server. The customer configures them by using the native phone configuration mechanism (usually a web page or tftp file). The customer must keep the device and line configuration in the Cisco Unified CallManager database synchronized with the native phone configuration (for example, extension 1002 on the phone and 1002 in Cisco Unified CallManager). Additionally, if the directory number of a line is changed, ensure that it gets changed in both Cisco Unified CallManager Administration and in the native phone configuration mechanism.

Enabling Digest Authentication for Third-Party SIP Phones

To enable digest authentication for third-party SIP phones, the administrator must create a Phone Security Profile. (See Phone Security Profile Configuration in the *Cisco Unified CallManager Administration Guide, Release 5.0(4)*.) On the Phone Security Profile Configuration window, check the Enable Digest Authentication check box. After the security profile is configured, the administrator must assign that security profile to the SIP phone by using the Phone Configuration window. If this check box is not checked, Cisco Unified CallManager will use digest authentication for purposes of identifying the phone by the end user ID, and it will not verify the digest password. If the check box is checked, Cisco Unified CallManager will verify the password.

Note Cisco Unified CallManager does not support Transport Layer Security (TLS) from third-party SIP phones.

DTMF Reception

To require DTMF reception, check the Require DTMF Reception check box that displays on the Phone Configuration window in Cisco Unified CallManager Administration.

Migrating from Cisco Unified CallManager Release 5.0(1), and above, to Cisco Unified CallManager, Release 5.1(1)

In Cisco Unified CallManager, Release 5.1(1) and above, certain characteristics for Basic and Advanced Third-Party SIP Phones changed. These characteristics include changes to the Maximum Number of Calls per Device, Default Maximum number of calls per DN, and Default Busy Trigger per DN fields that display on the Directory Number Configuration window in Cisco Unified CallManager Administration. See Table 4-2 and Table 4-3 for more information.

Table 4-2 Directory Number Migration Changes for Basic Third-Party SIP Phones

Field Name	Old Value	New Value
Maximum Number of Calls Per Device	8	2
Default Maximum Number of Calls per DN	4	2
Default Busy Trigger per DN	2	2

Table 4-3 Directory Number Migration Changes for Advanced Third-Party SIP Phones

Field Name	Old Value	New Value
Maximum Number of Calls Per Device	64	16
Default Maximum Number of Calls per DN	4	2
Default Busy Trigger per DN	2	2

For users that have third-party SIP phones that are configured on release 5.0(1) and above that are migrating/upgrading to release 5.1(1) or above, be aware after the upgrade, these devices retain their release 5.0 configured values. However, if users need to make changes to DN configuration values, users must change Maximum Number of Calls and Default Busy Trigger values on each DN.

For basic third-party SIP phones, only one line value needs to be modified. However, for advanced third-party SIP phones, users potentially must disassociate lines on the device before they can make any DN-related configuration changes. This situation potentially can happen if more than four lines are configured. An example scenario follows:

- Advanced phone configured with 6 lines with Maximum number of calls = 4 and Busy Trigger = 2 for each of the lines.
- After upgrade to release 5.1 or above, ensure maximum number of calls on the device is reduced to 16 or below before any DN changes. The current value on this phone equals 24 (6 lines * 4). The device essentially exists in a negative zone (16-24).
- User would disassociate two lines from the device.
- After the user disassociates those lines from the device, you can modify the DN characteristics for the remaining four lines by setting Maximum Number of Calls and Busy Trigger to an appropriate value.
- User reassociates the disassociated lines.

Third-Party SIP Phone Configuration Checklist

Table 4-4 provides steps to manually configure a third-party SIP phone by using Cisco Unified CallManager Administration.

Table 4-4 Third-Party SIP Phone Configuration Checklist

Configuration Steps		Procedures and Related Topics
Step 1	Gather the following information about the phone: • MAC address • Physical location of the phone • Cisco Unified CallManager user to associate with the phone • Partition, calling search space, and location information, if used • Number of lines and associated DNs to assign to the phone	
Step 2	Configure the end user that will be the Digest User. **Note** If the third-party SIP phone does not support an authorization ID (digest user), create a user with a user ID that matches the DN of the third-party phone. For example, create an end user named 1000 and create a DN of 1000 for the phone. Assign this user to the phone (see Step 8).	Adding an End User, *Cisco Unified CallManager Administration Guide, Release 5.0(4)*
Step 3	Configure the SIP Profile or use the default profile. The SIP Profile gets added to the SIP phone by using the Phone Configuration window. **Note** Third-party SIP phones use only the SIP Profile Information section of the SIP Profile Configuration window.	Configuring SIP Profiles, *Cisco Unified CallManager Administration Guide, Release 5.0(4)* Configuring Cisco Unified IP Phones, *Cisco Unified CallManager Administration Guide, Release 5.0(4)*

Table 4-4 Third-Party SIP Phone Configuration Checklist (continued)

Configuration Steps		Procedures and Related Topics
Step 4	Configure the Phone Security Profile. To use digest authentication, you must configure a new phone security profile. If you use one of the standard, nonsecure SIP profiles that are provided for auto-registration, you cannot enable digest authentication.	Enabling Digest Authentication for Third-Party SIP Phones, page 4-2 Phone Security Profile Configuration, *Cisco Unified CallManager Administration Guide, Release 5.0(4)* *Cisco Unified CallManager Security Guide, Release 5.0(4)*
Step 5	Add and configure the third-party SIP phone by choosing Third-party SIP Device (Advanced) or (Basic) from the Add a New Phone Configuration window. **Note** Third-party SIP Device (Basic) supports one line and consumes three license units, and Third-party SIP Device (Advanced) supports up to eight lines and video and consumes six license units.	Configuring Cisco Unified IP Phones, *Cisco Unified CallManager Administration Guide, Release 5.0(4)*
Step 6	Add and configure lines (DNs) on the phone.	Directory Number Configuration, *Cisco Unified CallManager Administration Guide, Release 5.0(4)*
Step 7	In the End User Configuration window, associate the third-party SIP phone with the user by using Device Association and choosing the SIP phone.	Associating Devices to an End User, *Cisco Unified CallManager Administration Guide, Release 5.0(4)*
Step 8	In the Digest User field of the Phone Configuration window, choose the end user that you created in Step 2.	Phone Configuration Settings, *Cisco Unified CallManager Administration Guide, Release 5.0(4)*
Step 9	Provide power, install, verify network connectivity, and configure network settings for the third-party SIP phone.	Refer to the administration guide that was provided with your SIP phone.
Step 10	Make calls with the third-party SIP phone.	Refer to the user guide that came with your third-party SIP phone.

Where to Find More Information

- Directory Number Configuration, *Cisco Unified CallManager Administration Guide, Release 5.0(4)*
- Cisco Unified IP Phone Configuration, *Cisco Unified CallManager Administration Guide, Release 5.0(4)*
- SIP Profile Configuration, *Cisco Unified CallManager Administration Guide, Release 5.0(4)*
- End User Configuration, *Cisco Unified CallManager Administration Guide, Release 5.0(4)*
- Cisco Unified IP Phones, *Cisco Unified CallManager System Guide, Release 5.0(4)*