Non-Cisco SIP phones setup

This appendix provides information about Configuring Non-Cisco Phones That Are Running SIP.

About non-Cisco SIP phone setup

Cisco Unified Communications Manager supports Cisco Unified IP Phones with SIP as well as RFC3261-compliant phones that are running SIP from third-party companies. This appendix describes how to configure the third-party phones that are running SIP by using Cisco Unified Communications Manager Administration.

Third-party SIP phone setup process

Cisco Unified Communications Manager supports Cisco Unified IP Phones with SIP as well as RFC3261-compliant phones that are running SIP from third-party companies. You can manually configure a third-party phone that is running SIP by using Cisco Unified Communications Manager Administration.

Procedure

Step 1 Gather the information about the phone.

- MAC address
- Physical location of the phone
- Cisco Unified Communications Manager 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 Determine whether sufficient Device License Units are available. If not, purchase and install additional Device License Units. Third-Party SIP Devices (Basic) and (Advanced) consume three and six Device License Units each, respectively.
See topics related to calculating the number of required licenses and obtaining a license in the *Cisco Unified Communications Manager Features and Services Guide*.

Step 3 Configure the end user that will be the Digest User.

Note If the third-party phone that is running SIP 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 9, on page 2).

Step 4 Configure the SIP Profile or use the default profile. The SIP Profile gets added to the phone that is running SIP by using the Phone Configuration window.

Note Third-party phones that are running SIP use only the SIP Profile Information section of the SIP Profile Configuration window.

Step 5 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.

Step 6 Add and configure the third-party phone that is running SIP 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.

Step 7 Add and configure lines (DNs) on the phone.

Step 8 In the End User Configuration window, associate the third-party phone that is running SIP with the user by using Device Association and choosing the phone that is running SIP.

Step 9 In the Digest User field of the Phone Configuration window, choose the end user that you created in Step 3, on page 2.

Step 10 Provide power, install, verify network connectivity, and configure network settings for the third-party phone that is running SIP.
See the administration guide that was provided with your phone that is running SIP.

Step 11 Make calls with the third-party phone that is running SIP.
See the user guide that came with your third-party phone that is running SIP.

Related Topics

- Phone security profile setup
- Set up Cisco Unified IP Phone
- Set up speed-dial buttons or abbreviated dialing
- About SIP profile setup
- About end user setup
- Associate devices to end user
- Enable digest authentication for third-party SIP phones, on page 4

Different setups for SIP phones

The following table provides a comparison overview of the configuration differences between Cisco Unified IP Phones and third-party phones that are running SIP.

Table 1: Model configuration comparison for phones that are running SIP

Phone That Is Running SIP	Integrated with Centralized TFTP	Sends MAC Address	Downloads Softkey File	Downloads Dial Plan File	Supports Cisco Unified Communications Manager Failover and Fallback	Supports Reset and Restart
Cisco Unified IP Phone 7911, 7941, 7961, 7970, 7971	Yes	Yes	Yes	Yes	Yes	Yes
Cisco Unified IP Phone 7940, 7960	Yes	Yes	No	Yes	Yes	Yes
Cisco Unified IP Phone 7905, 7912	Yes	Yes	No	No	Yes	Yes
Third-party phone that is running SIP	No	No	No	No	No	No

Use Cisco Unified Communications Manager Administration to configure third-party phones that are running SIP. The administrator must also perform configuration steps on the third-party phone that is running SIP; see following examples:

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

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

Related Topics

Where to find more information, on page 6

How Cisco Unified Communications Manager identifies third-party phones

Because third-party phones that are running SIP do not send a MAC address, they must identify themselves by using username.

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 Communications Manager Administration. The administrator configures the SIP third-party phone with the user; for example, swhite, in the Digest User field of Phone Configuration window.

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.

Related Topics

Set up Cisco Unified IP Phone
About end user setup

Third-party phones running SIP and TFTP

Third-party phones that are running SIP do not get configured by using the Cisco Unified Communications Manager 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 Communications Manager database synchronized with the native phone configuration (for example, extension 1002 on the phone and 1002 in Cisco Unified Communications Manager). Additionally, if the directory number of a line is changed, ensure that it gets changed in both Cisco Unified Communications Manager Administration and in the native phone configuration mechanism.

Enable digest authentication for third-party SIP phones

To enable digest authentication for third-party phones that are running SIP, the administrator must create a Phone Security Profile. See the *Cisco Unified Communications Manager Security Guide* for details. 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 phone that is running SIP by using the Phone Configuration window. If this check box is not checked, Cisco Unified Communications Manager 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 Communications Manager will verify the password.

> **Note** Cisco Unified Communications Manager does not support Transport Layer Security (TLS) from third-party phones that are running SIP.

Related Topics

Phone security profile setup

DTMF reception

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

Licensing third-party SIP phones

Licensing of third-party phones that are running SIP enforces the following limitations:

- Third-party SIP Device (Basic)—Video calls do not get supported. Video enforcement occurs as part of the offer/answer process. If video-related media is provided as part of an offer or answer from a SIP device that is not permitted to negotiate video, only the non-video-related parts of the call get extended to the destination party. Similarly, a SIP endpoint that is not permitted to negotiate media will not receive any video-related media in the SDP that is sent from Cisco Unified Communications Manager.
- Third-party SIP Device (Advanced) and (Basic)—Cisco-specific SIP extensions do not get supported. Some Cisco-specific SIP extensions that are not supported include service URIs, header extensions, dialog subscriptions, and remote call control proprietary mime types. Cisco Unified Communications Manager will reject any request from a phone that is running SIP that is not permitted to use an advanced feature that uses a service request URI (such as Call Pickup URI, Meet Me Service URI). The SIP profile specifies service URIs. The profile gets assigned to SIP devices. Cisco Unified Communications Manager will block features that require the use of Cisco-specific SIP extensions.

> **Note** Ensure that any wireless third-party SIP client or device is configured as a Third-Party SIP Device (Advanced) in conformance with Cisco Unified Communications Manager licensing policy.

For more information about Cisco SIP Extensions, contact your Cisco representative.

In Cisco Unified Communications Manager, Release 5.1(1) and above, certain characteristics for Basic and Advanced Third-Party phones that are running SIP 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 Communications Manager Administration. The following tables provide more information.

Table 2: Directory number migration changes for basic third-party phones that are running SIP

Field Name	Old Value	New Value
Maximum Number of Calls Per Device	8	2

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

Table 3: Directory number migration changes for advanced third-party phones that are running SIP

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 phones that are running SIP that are configured on any version of release 5.0 that are migrating/upgrading to release 6.0(1) or above, be aware that, 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 phones that are running SIP, only one line value needs to be modified. However, for advanced third-party phones that are running SIP, 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 line.
- After upgrade to release 6.1, 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.

Where to find more information

Related Topics

Third-party SIP phone setup process, on page 1