SIP Profile Configuration

A SIP profile comprises the set of SIP attributes that are associated with SIP trunks and SIP endpoints. SIP profiles include information such as name, description, timing, retry, call pickup URI, and so on. The profiles contain some standard entries that cannot be deleted or changed.

Use the following topics to configure and locate SIP profiles:

Finding SIP Profiles

This topic describes how to use the Find and List SIP Profile window. The function searches every type of SIP profile against the following categories:

- Profile name
- Description

Procedure

Step 1 Choose **Device > Device Settings > SIP Profile**.

The Find and List SIP Profile window displays.

Step 2 From the drop-down lists, choose your search text for the type of SIP profiles that you want listed and click **Find**.

Note To find all SIP profiles that are registered in the database, click **Find** without entering any text.

The window refreshes and then displays the SIP profiles that match your search criteria.

Step 3 From the list of records that match your search criteria, choose the SIP profile.

Step 4 To delete multiple SIP profiles from the Find and List SIP Profiles window, check the check boxes next to the appropriate SIP profiles and click **Delete Selected**. To choose all the SIP profiles in the window, click the **Select All** button and then click **Delete Selected**.

Additional Information

See the "Related Topics" section on page 79-9.

Configuring SIP Profiles

Perform the following procedure to add, copy, or update a SIP profile.

Procedure

Step 1 Choose **Device > Device Settings > SIP Profile**.

The Find and List SIP Profile window displays.

Step 2 Perform one of the followings tasks:

- To copy an existing SIP profile, locate the appropriate SIP profile as described in "Finding SIP Profiles" section on page 79-1, click the **Copy** button next to the SIP profile that you want to copy and continue with Step 3.
- To add a new SIP profile, click the **Add New** button and continue with Step 3.
- To update an existing SIP profile, locate the appropriate SIP profile as described in "Finding SIP Profiles" section on page 79-1 and continue with Step 3.

Step 3 Enter the appropriate settings as described in Table 79-1.

Step 4 Click **Save**.

Additional Information

See the "Related Topics" section on page 79-9.

SIP Profile Configuration Settings

Table 79-1 describes the available settings in the SIP Profile Configuration window. For more information about related procedures, see the "Related Topics" section on page 79-9.

Table 79-1 SIP Profile Configuration Settings

Field	Description
SIP Profile Information	
Name	Enter a name to identify the SIP profile; for example, SIP_7905. The value can include 1 to 50 characters, including alphanumeric characters, dot, dash, and underscores.

Table 79-1 SIP Profile Configuration Settings (continued)

Field	Description
Description	This field identifies the purpose of the SIP profile; for example, SIP for Model 7970.
Default Telephony Event Payload Type	This field specifies the default payload type for RFC2833 telephony event. See RFC 2833 for more information. In most cases, the default value specifies the appropriate payload type. Be sure that you have a firm understanding of this parameter before changing it, as changes could result in DTMF tones not being received or generated. The default value specifies 101 with range from 96 to 127. The value of this parameter affects calls with the following conditions: • The call is an outgoing SIP call from Cisco Unified CallManager. • For the calling SIP trunk, the Media Termination Point Required check box is checked on the SIP Trunk Configuration window.
Redirect by Application	Checking this check box and configuring this SIP Profile on the SIP trunk allows the Cisco Unified CallManager administrator to • Apply a specific calling search space to redirected contacts that are received in the 3xx response. • Apply digit analysis to the redirected contacts to make sure that the call get routed correctly. • Prevent DOS attack by limiting the number of redirection (recursive redirection) that a service parameter can set. • Allow other features to be invoked while the redirection is taking place. Getting redirected to a restricted phone number (such as an international number) means that handling redirection at the stack level will cause the call to be routed instead of being blocked. This represents the behavior you will get if the Redirect by Application check box is unchecked. See the "Redirection" section on page 41-12.
Disable Early Media on 180	By default, Cisco Unified CallManager will signal the calling phone to play local ringback if SDP is not received in the 180 or 183 response. If SDP is included in the 180 or 183 response, instead of playing ringback locally, Cisco Unified CallManager will connect media and the calling phone will play whatever the called device is sending (such as ringback or busy signal). If you do not receive ringback, the device to which you are connecting may be including SDP in the 180 response, but it is not sending any media before the 200OK response. In this case, check this check box to play local ringback on the calling phone and connect the media upon receipt of the 200OK response **Note** Even though the phone that is receiving ringback is the calling phone, you need the configuration on the called device's profile because it determines the behavior. See the "Use of Early Media" section on page 41-6.

Table 79-1 SIP Profile Configuration Settings (continued)

Field	Description
Parameters used in Phone	
Timer Invite Expires (seconds)	This field specifies the time, in seconds, after which a SIP INVITE expires. The Expires header uses this value. Valid values include any positive number; 180 specifies the default.
Timer Register Delta (seconds)	Use this parameter in conjunction with the Timer Register Expires setting. The phone will reregister Timer Register Delta seconds before the registration period ends. The registration period gets determined by the value of the SIP Station Keepalive Interval service parameter. Valid values range from 32767 to 0. Default specifies 5.
Timer Register Expires (seconds)	This field specifies the value that the SIP phone will send in the Expires header of the REGISTER message. Valid values include any positive number; however, 3600 (1 hour) specifies the default value. In the 200OK response to REGISTER, Cisco Unified CallManager will include an Expires header with the configured value of the SIP Station KeepAlive Interval service parameter. This value in the 200OK determines the time, in seconds, after which the registration expires. The phone will refresh the registration Timer Register Delta seconds before the end of this interval.
Timer T1 (msec)	This field specifies the lowest value, in milliseconds, of the retransmission timer for SIP messages. Valid values include any positive number. Default specifies 500.
Timer T2 (msec)	This field specifies the highest value, in milliseconds, of the retransmission timer for SIP messages. Valid values include any positive number. Default specifies 4000.
Retry INVITE	This field specifies the maximum number of times that an INVITE request will be retransmitted. Valid values include any positive number. Default specifies 6.
Retry Non-INVITE	This field specifies the maximum number of times that a SIP message other than an INVITE request will be retransmitted. Valid values include any positive number. Default specifies 10.
Start Media Port	This field designates the start real-time protocol (RTP) port for media. Media port ranges from 16384 to 32766. Default specifies 16384.
Stop Media Port	This field designates the stop real-time protocol (RTP) port for media. Media port ranges from 16384 to 32766. Default specifies 32766.
Call Pickup URI	This URI provides a unique address that the SIP phone will send to Cisco Unified CallManager to invoke the call pickup feature.
Call Pickup Group Other URI	This URI provides a unique address that the SIP phone will send to Cisco Unified CallManager to invoke the call pickup group other feature.
Call Pickup Group URI	This URI provides a unique address that the SIP phone will send to Cisco Unified CallManager to invoke the call pickup group feature.
Meet Me Service URI	This URI provides a unique address that the SIP phone will send to Cisco Unified CallManager to invoke the meet me conference feature.
Call Forward URI	This URI provides a unique address that the SIP phone will send to Cisco Unified CallManager to invoke the call forward feature.

Table 79-1 SIP Profile Configuration Settings (continued)

Field	Description
Abbreviated Dial URI	This URI provides a unique address that the SIP phone will send to Cisco Unified CallManager to invoke the abbreviated dial feature. Speed dials that are not associated with a line key (abbreviated dial indices) will not download to the phone. The phone will use the feature indication mechanism (INVITE with Call-Info header) to indicate when an abbreviated dial number has been entered. The request URI will contain the abbreviated dial digits (for example, 14), and the Call-Info header will indicate the abbreviated dial feature. Cisco Unified CallManager will translate the abbreviated dial digits into the configured digit string and extend the call with that string. If no digit string has been configured for the abbreviated dial digits, a 404 Not Found response gets returned to the phone.
User Info	This field configures the user= parameter in the REGISTER message. Valid values follow: • none—No value gets inserted. • phone—The value user=phone gets inserted in the To, From, and Contact Headers for REGISTER. • ip—The value user=ip gets inserted in the To, From, and Contact Headers for REGISTER.
DTMF DB Level	This field specifies in-band DTMF digit tone level. Valid values follow: • 1 to 6 dB below nominal • 2 to 3 dB below nominal • 3 nominal • 4 to 3 dB above nominal • 5 to 6 dB above nominal
KPML	This field specifies key press markup language. Choices include Signal only, DTMF only, None, and Both. • None - KPML is off. • Signal only - Digit collection from the phone that is needed to place the call • DTMF only - Transport of out-of-band DTMF digits after the call is established; for example, to access an IVR system • Both - Signal and DTMF
Call Hold Ring Back	If you have a call on hold and are talking on another call, when you hang up the call, this parameter causes the phone to ring to let you know that you still have another party on hold. Valid values follow: • Off permanently and cannot be turned on and off locally by using the user interface. • On permanently and cannot be turned on and off locally by using the user interface.

Table 79-1 SIP Profile Configuration Settings (continued)

Field	Description
Anonymous Call Block	This field configures anonymous call block. Valid values follow: • Off—Disabled permanently and cannot be turned on and off locally by using the user interface. • On—Enabled permanently and cannot be turned on and off locally by using the user interface.
Caller ID Blocking	This field configures caller ID blocking. When blocking is enabled, the phone blocks its own number or e-mail address from phones that have caller identification enabled. Valid values follow: • Off—Disabled permanently and cannot be turned on and off locally by using the user interface. • On—Enabled permanently and cannot be turned on and off locally by using the user interface.
Do Not Disturb Control	This field sets the Do Not Disturb (DND) feature. Valid values follow: • User controlled (Default)—The phones dndControl parameter should be 0. • Admin Controlled—The phones dndControl parameter should be 2.
Telnet Level for 7940 and 7960	Cisco SIP IP Phone models 7940 and 7960 do not support ssh for login access or http that is used to collect logs; however, these phones support telnet, which lets the user control the phone, collect debugs, and look at configuration settings. This field controls the telnet_level configuration parameter with the following possible values: • Disabled (no access) • Limited (some access but cannot run privileged commands) • Enabled (full access)
Timer Keep Alive Expires (seconds)	Cisco Unified CallManager requires a keepalive mechanism to support redundancy. This field specifies the interval between keepalive messages sent to the backup Cisco Unified CallManager to ensure it is available in the event a failover is required.
Timer Subscribe Expires (seconds)	This field specifies the time, in seconds, after which a subscription expires. This value gets inserted into the Expires header field. Valid values include any positive number; however, 120 specifies the default value.
Timer Subscribe Delta (seconds)	Use this parameter in conjunction with the Timer Subscribe Expires setting. The phone will resubscribe Timer Subscribe Delta seconds before the subscription period ends, as governed by Timer Subscribe Expires. Valid values range from 3 to 15. Default specifies 5.
Maximum Redirections	Use this configuration variable to determine the maximum number of times that the phone will allow a call to be redirected before dropping the call. Default specifies 70 redirections.
Off Hook to First Digit Timer (microseconds)	This field specifies the time in microseconds that passes when the phone goes off hook and the first digit timer gets set. The value ranges from 0 - 15,000 microseconds. Default specifies 15,000 microseconds.

Table 79-1 SIP Profile Configuration Settings (continued)

Field	Description
Conference Join Enabled	This check box determines whether the SIP IP phone 7940 or 7960, when the conference initiator that is using that phone hangs up, should attempt to join the remaining conference attendees. Check the check box if you want to join the remaining conference attendees; leave it unchecked if you do not want to join the remaining conference attendees. **Note** This check box applies to the Cisco SIP IP phones 7941/61/70/71/11 when they are in SRST mode only.
RFC 2543 Hold	Check this check box to enable setting connection address to 0.0.0.0 per RFC2543 when call hold is signaled to Cisco Unified CallManager. This allows backward compatibility with endpoints that do not support RFC3264.
Semi Attended Transfer	This check box determines whether the Cisco SIP IP phone 7940 and 7960 caller can transfer the second leg of an attended transfer while the call is ringing. Check the check box if you want semi-attended transfer enabled; leave it unchecked if you want semi-attended transfer disabled. **Note** This check box applies to the Cisco SIP IP phones 7941/61/70/71/11 when they are in SRST mode only.
Enable VAD	Check this check box if you want voice activation detection (VAD) enabled; leave it unchecked if you want VAD disabled. When VAD is enabled, no media gets transmitted when voice is detected.

Table 79-1 SIP Profile Configuration Settings (continued)

Field	Description
Stutter Message Waiting	Check this check box if you want stutter dial tone when the phone goes off hook and a message is waiting; leave unchecked if you do not want a stutter dial tone when a message is waiting.
Call Stats	Check this check box if you want RTP statistics in BYE requests and responses enabled; leave unchecked if you want RTP statistics in BYE requests and responses disabled. If this check box is checked, the phone inserts the headers RTP-RxStat and RTP-TxStat as follows: • RTP-RxStat:Dur=a,Pkt=b,Oct=c,LatePkt=d,LostPkt=e,AvgJit=f • RTP-TxStat: Dur=g,Pkt=h,Oct=i where: • Dur—Total number of seconds since the beginning of reception or transmission. • Pkt—Total number of RTP packets received or transmitted. • Oct—Total number of RTP payload octets received or transmitted (not including RTP header). • LatePkt—Total number of late RTP packets received. • LostPkt—Total number of lost RTP packets received (not including the late RTP packets). • AvgJit—Average jitter, which is an estimate of the statistical variance of the RTP packet interarrival time, measured in timestamp unit and calculated according to RFC 1889. • a, b, c, d, e, f, g, h, and i—Integers

Deleting SIP Profiles

This section describes how to delete a SIP profile.

Before You Begin

To find out which devices are using the SIP profile, choose **Dependency Records** link from the Related Links drop-down list box in the SIP Profile Configuration window. If the dependency records are not enabled for the system, the dependency records summary window displays a message. For more information about dependency records, see the "Accessing Dependency Records" section on page A-2.

Procedure

Step 1 Locate the SIP profile that you want to delete. See the "Finding SIP Profiles" section on page 79-1.

Step 2 From the SIP Profile Configuration window, click **Delete**.

A message displays that states that this you cannot undo this action.

Step 3 To delete the SIP profile, click **OK** or, to cancel the deletion, click **Cancel**.

Additional Information

See the "Related Topics" section on page 79-9.

Resetting a SIP Profile

Perform the following procedure to reset a SIP profile.

Procedure

Step 1 From Cisco Unified CallManager Administration, choose **Device > Device Settings> SIP Profile**.

Step 2 Locate the SIP profile that you want to reset. See the "Finding SIP Profiles" section on page 79-1.

Step 3 Click the SIP profile that you want to reset.

The SIP Profile Configuration window displays.

Step 4 Click **Reset**.

The Device Reset dialog displays.

Step 5 Click one of the following choices:

- **Restart—**Restarts the chosen devices without shutting them down (reregisters the phones and trunks with Cisco Unified CallManager).
- **Reset**—Shuts down, then restarts, the device.
- **Close**—Closes the Reset Device dialog without performing any action.

Additional Information

See the "Related Topics" section on page 79-9.

Related Topics

- Finding SIP Profiles, page 79-1
- Configuring SIP Profiles, page 79-2
- SIP Profile Configuration Settings, page 79-2
- Deleting SIP Profiles, page 79-8
- Resetting a SIP Profile, page 79-9

- Configuring Cisco Unified IP Phones, page 70-2
- Configuring a Trunk, page 71-2
- Understanding Session Initiation Protocol (SIP), *Cisco Unified CallManager System Guide*