SIP profile setup

This chapter provides information to configure and locate SIP profiles. 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.

About SIP profile setup

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SIP profile reset

For instructions on how to reset a SIP profile, see the descriptions of the Reset Selected and Reset buttons in the *Cisco Unified Communications Manager System Guide*.

Related Topics

Synchronize SIP profile settings with SIP devices, on page 18

SIP profile deletion

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 *Cisco Unified Communications Manager System Guide*.

SIP profile settings

The following table describes the available settings in the SIP Profile Configuration window.

Table 1: SIP profile 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.
Description	This field identifies the purpose of the SIP profile; for example, SIP for 7970. The description can include up to 50 characters in any language, but it cannot include double-quotes ("), percentage sign (%), ampersand (&), back-slash (\\), or angle brackets (<>).
Default MTP 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 Communications Manager. • For the calling SIP trunk, the Media Termination Point Required check box is checked on the SIP Trunk Configuration window.
Resource Priority Namespace List	Select a configured Resource Priority Namespace list from the drop-down menu. Configure the lists in the Resource Priority Namespace List menu that is accessed from **System** > **MLPP** > **Namespace**.

Field	Description
Early Offer for G.Clear Calls	The Early Offer for G.Clear Calls feature supports both standards-based G.Clear (CLEARMODE) and proprietary Cisco Session Description Protocols (SDP). To enable or disable Early Offer for G.Clear Calls, choose one of the following options: • Disabled • CLEARMODE • CCD • G.nX64 • X-CCD
SDP Session-level Bandwidth Modifier for Early Offer and Re-invites	The Session Level Bandwidth Modifier specifies the maximum amount of bandwidth needed when all the media streams are used. There are three Session Level Bandwidth Modifiers: Transport Independent Application Specific (TIAS), Application Specific (AS), and Conference Total (CT). Select one of the following options to specify which Session Level Bandwidth Modifier to include in the SDP portion of SIP Early Offer or Reinvite requests. • TIAS and AS • TIAS only • AS only • CT only
User-Agent and Server header information	This feature indicates how Cisco Unified Communications Manager handles the User-Agent and Server header information in a SIP message. Choose one of the following three options: • Send Unified CM Version Information as User-Agent Header—For INVITE requests, the User-Agent header is included with the CM version header information. For responses, the Server header is omitted. Cisco Unified Communications Manager passes through any contact headers untouched. This is the default behavior. • Pass Through Received Information as Contact Header Parameters —If this option is selected, the User-Agent/Server header information is passed as Contact header parameters. The User-Agent/Server header is derived from the received Contact header parameters, if present. Otherwise, they are taken from the received User-Agent/Server headers. • Pass Through Received Information as User-Agent and Server Header—If this option is selected, the User-Agent/Server header information is passed as User-Agent/Server headers. The User-Agent/Server header is derived from the received Contact header parameters, if present. Otherwise, they are taken from the received User-Agent/Server headers.

Field	Description
Accept Audio Codec Perferences in Received Offer	Select **On** to enable Cisco Unified Communications Manager to honor the preference of audio codecs in received offer and preserve it while processing. Select **Off** to enable CUCM to ignore the preference of audio codecs in received offer and apply the locally configured Audio Codec Preference List. The default will select the service parameter configuration.
Dial String Interpretation	Cisco Unified Communications Manager uses the Dial String Interpretation policy to determine if the SIP identity header is a directory number or directory URI. Because directory numbers and directory URIs are saved in different database lookup tables, Cisco Unified Communications Manager examines the characters in the SIP identity header's user portion, which is the portion of the SIP address that is before the @ sign (for example, user@IP address or user@domain). To configure the Dial String Interpretation, choose one of the following options from the drop-down list box: • Always treat all dial strings as URI addresses—Cisco Unified Communications Manager treats the address of incoming calls as if they were URI addresses. • Phone number consists of characters 0–9, A–D, *, and + (others treated as URI addresses)—Cisco Unified Communications Manager treats the incoming call as a directory number if all the characters in the user portion of the SIP identity header fall within this range. If the user portion of the address uses any characters that do not fall within this range, the address is treated as a URI. • Phone number consists of characters 0-9, *, and + (others treated as URI addresses)—Cisco Unified Communications Manager treats the incoming call as a directory number if all the characters in the user portion of the SIP identity header fall within this range. If the user portion of the address uses any characters that do not fall within this range, the address is treated as a URI. **Note** If the user=phone tag is present in the Request URI, Cisco Unified Communications Manager always treats the dial string as a number regardless of what option you choose for the Dial String Interpretation field.

Field	Description
Redirect by Application	Checking this check box and configuring this SIP Profile on the SIP trunk allows the Cisco Unified Communications Manager 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 causes the call to be routed instead of being blocked. This behavior occurs if the Redirect by Application check box is unchecked.
Disable Early Media on 180	By default, Cisco Unified Communications Manager signals 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 Communications Manager connects media, and the calling phone plays 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 profile because it determines the behavior.
Outgoing T.38 INVITE Include Audio mline	The parameter allows the system to accept a signal from Microsoft Exchange that causes it to switch the call from audio to T.38 fax. To use this feature, you must also configure a SIP trunk with this SIP profile. For more information, see Chapter 68, Trunk "Configuration." **Note** The parameter applies to SIP trunks only, not phones that are running SIP or other endpoints.

Field	Description
Enable ANAT	This option allows a dual-stack SIP trunk to offer both IPv4 and IPv6 media. When you check both the Enable ANAT and the MTP Required check boxes, Cisco Unified Communications Manager inserts a dual-stack MTP and sends out an offer with two m-lines, one for IPv4 and another for IPv6. If a dual- stack MTP cannot be allocated, Cisco Unified Communications Manager sends an INVITE without SDP. When you check the Enable ANAT check box and the Media Termination Point Required check box is unchecked, Cisco Unified Communications Manager sends an INVITE without SDP. When the Enable ANAT and Media Termination Point Required check boxes display as unchecked (or when an MTP cannot be allocated), Cisco Unified Communications Manager sends an INVITE without SDP. When you uncheck the Enable ANAT check box but you check the Media Termination Point Required check box, consider the information, which assumes that an MTP can be allocated: • Cisco Unified Communications Manager sends an IPv4 address in the SDP for SIP trunks with an IP Addressing Mode of IPv4 Only. • Cisco Unified Communications Manager sends an IPv6 address in the SDP for SIP trunks with an IP Addressing Mode of IPv6 Only. • For dual-stack SIP trunks, Cisco Unified Communications Manager determines which IP address type to send in the SDP based on the configuration for the IP Addressing Mode Preference for Media enterprise parameter.
Require SDP Inactive Exchange for Mid-Call Media Change	This feature designates how Cisco Unified Communications Manager handles mid-call updates to codecs or connection information such as IP address or port numbers. If the box is checked, during mid-call codec or connection updates Cisco Unified Communications Manager sends an INVITE a=inactive SDP message to the endpoint to break the media exchange. This is required if an endpoint is not capable of reacting to changes in the codec or connection information without disconnecting the media. This applies only to audio and video streams within SIP-SIP calls. **Note** For early offer enabled SIP trunks, this parameter will be overridden by the Send send-receive SDP in mid-call INVITE parameter. If the box is unchecked, Cisco Unified Communications Manager passes the mid-call SDP to the peer leg without sending a prior Inactive SDP to break the media exchange. This is the default behavior.

Field	Description
Use Fully Qualified Domain Name in SIP Requests	This feature enables Cisco Unified Communications Manager to relay an alphanumeric hostname of a caller by passing it through to the called device or outbound trunk as a part of the SIP header information. • If the box is unchecked, the IP address for Cisco Unified Communications Manager will be passed to the line device or outbound trunk instead of the user's hostname. This is the default behavior. • If the box is checked, Cisco Unified Communications Manager will relay an alphanumeric hostname of a caller by passing it through to the called endpoint as a part of the SIP header information. This enables the called endpoint to return the call using the received or missed call list. If the call is originating from a line device on the Cisco Unified Communications Manager cluster, and is being routed on a SIP trunk then the configured Organizational Top-Level Domain (e.g., cisco.com) will be used in the Identity headers, such as From, Remote-Party-ID, and P-Asserted-ID. If the call is originating from a trunk on Cisco Unified Communications Manager and is being routed on a SIP trunk then: ◦ If the inbound call provides a host or domain in the caller's information, the outbound SIP trunk messaging will preserve the hostname in the Identity headers, such as From, Remote-Party-ID, and P-Asserted-ID ◦ If the inbound call does not provide a host or domain in the caller's information, the configured Organizational Top-Level Domain will be used in the Identity headers, such as From, Remote-Party-ID, and P-Asserted-ID
Assured Services SIP conformance	This checkbox should be checked for third-party AS-SIP endpoints as well as AS-SIP trunks to ensure proper Assured Service behavior. This setting provides specific Assured Service behavior that affects services such as Conference factory and SRTP.
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)	This field is intended to be used by SIP endpoints only. The endpoint receives this value via a tftp config file. The end point reregisters 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 for Timer Register Delta range from 32767 to 0. The default value is 5.

Field	Description
Timer Register Expires (seconds)	This field is intended to be used by SIP endpoints only. The SIP endpoint receives the value via a tftp config file. This field specifies the value that the phone that is running SIP sends in the Expires header of the REGISTER message. Valid values include any positive number; however, 3600 (1 hour) specifies the default value. If the endpoint sends a shorter Expires value than the value of the SIP Station Keepalive Interval service parameter, Cisco Unified Communications Manager responds with a 423 "Interval Too Brief". If the endpoint sends an Expires value that is greater than the SIP Station Keepalive Interval service parameter value, Cisco Unified Communications Manager responds with a 200 OK that includes the Keepalive Interval value for Expires. **Note** For mobile phones that are running SIP, Cisco Unified Communications Manager uses the value in this field instead of the value that the SIP Station KeepAlive Interval service parameter specifies to determine the registration period. **Note** For TCP connections, the value for the Timer Register Expires field must be lower than the value for the SIP TCP Unused Connection service parameter.
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 gets 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 gets 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 phone that is running SIP sends to Cisco Unified Communications Manager to invoke the call pickup feature.
Call Pickup Group Other URI	This URI provides a unique address that the phone that is running SIP sends to Cisco Unified Communications Manager to invoke the call pickup group other feature.

Field	Description
Call Pickup Group URI	This URI provides a unique address that the phone that is running SIP sends to Cisco Unified Communications Manager to invoke the call pickup group feature.
Meet Me Service URI	This URI provides a unique address that the phone that is running SIP sends to Cisco Unified Communications Manager to invoke the meet me conference feature.
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
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.
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.

Field	Description
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—The dndControl parameter for the phone should specify 0. • Admin—The dndControl parameter for the phone should specify 2.
Telnet Level for 7940 and 7960	Cisco Unified IP Phones 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)
Resource Priority Namespace	This field enables the admin to select one of the cluster's defined Resource Priority Namespace network domains for assignment to a line via its SIP Profile.
Timer Keep Alive Expires (seconds)	Cisco Unified Communications Manager requires a keepalive mechanism to support redundancy. This field specifies the interval between keepalive messages that are sent to the backup Cisco Unified Communications Manager to ensure that it is available in the event that 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 resubscribes 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 allows a call to be redirected before dropping the call. Default specifies 70 redirections.

Field	Description
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.
Call Forward URI	This URI provides a unique address that the phone that is running SIP sends to Cisco Unified Communications Manager to invoke the call forward feature.
Abbreviated Dial URI	This URI provides a unique address that the phone that is running SIP sends to Cisco Unified Communications Manager to invoke the abbreviated dial feature. Speed dials that are not associated with a line key (abbreviated dial indices) do not download to the phone. The phone uses the feature indication mechanism (INVITE with Call-Info header) to indicate when an abbreviated dial number has been entered. The request URI contains the abbreviated dial digits (for example, 14), and the Call-Info header indicates the abbreviated dial feature. Cisco Unified Communications Manager translates 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.
Conference Join Enabled	This check box determines whether the Cisco Unified IP Phones 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 Unified 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 Communications Manager. This allows backward compatibility with endpoints that do not support RFC3264.
Semi Attended Transfer	This check box determines whether the Cisco Unified IP Phones 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 Unified 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.
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. This setting supports Cisco Unified IP Phones 7960 and 7940 that run SIP.

Field	Description
MLPP User Authorization	Check this box to enable MLPP User Authorization. MLPP User Authorization requires the phone to send in an MLPP username and password.
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 that are received or transmitted. • Oct—Total number of RTP payload octets that are received or transmitted (not including RTP header). • LatePkt—Total number of late RTP packets that are received. • LostPkt—Total number of lost RTP packets that are 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
Normalization Script	
Normalization Script	From the drop-down list box, choose the script that you want to apply to this SIP profile. To import another script, go to the SIP Normalization Script Configuration window (**Device** > **Device Settings** > **SIP Normalization Script**), and import a new script file.
Parameter Name/Parameter Value	Optionally, enter parameter names and parameter values. Valid values include all characters except equals signs (=), semi-colons (;), and non-printable characters, such as tabs. You can enter a parameter name with no value. To add another parameter line, click the + (plus) button. To delete a parameter line, click the - (minus) button. **Note** You must choose a script from the Normalization Script drop-down list box before you can enter parameter names and values.

Field	Description
Enable Trace	Check this check box to enable tracing within the script or uncheck this check box to disable tracing. When checked, the trace.output API provided to the Lua scripter produces SDI trace **Note** Cisco recommends that you only enable tracing while debugging a script. Tracing impacts performance and should not be enabled under normal operating conditions.
Incoming Requests FROM URI Settings	
Caller ID DN	Enter the pattern that you want to use for calling line ID, from 0 to 24 digits. For example, in North America: • 555XXXX = Variable calling line ID, where X equals an extension number. The CO appends the number with the area code if you do not specify it. • 55000 = Fixed calling line ID, where you want the Corporate number to be sent instead of the exact extension from which the call is placed. The CO appends the number with the area code if you do not specify it. You can also enter the international escape character +.
Caller Name	Enter a caller name to override the caller name that is received from the originating SIP Device.
Trunk Specific Configuration	

Field	Description
Reroute Incoming Request to new Trunk based on	Cisco Unified Communications Manager only accepts calls from the SIP device whose IP address matches the destination address of the configured SIP trunk. In addition, the port on which the SIP message arrives must match the one that is configured on the SIP trunk. After Cisco Unified Communications Manager accepts the call, Cisco Unified Communications Manager uses the configuration for this setting to determine whether the call should get rerouted to another trunk. From the drop-down list box, choose the method that Cisco Unified Communications Manager uses to identify the SIP trunk where the call gets rerouted: • Never—If the SIP trunk matches the IP address of the originating device, choose this option, which equals the default setting. Cisco Unified Communications Manager, which identifies the trunk by using the source IP address of the incoming packet and the signaling port number, does not route the call to a different (new) SIP trunk. The call occurs on the SIP trunk on which the call arrived. • Contact Info Header—If the SIP trunk uses a SIP proxy, choose this option. Cisco Unified Communications Manager parses the contact header in the incoming request and uses the IP address or domain name and signaling port number that is specified in the header to reroute the call to the SIP trunk that uses the IP address and port. If no SIP trunk is identified, the call occurs on the trunk on which the call arrived. • Call-Info Header with purpose=x-cisco-origIP—If the SIP trunk uses a Customer Voice Portal (CVP) or a Back-to-Back User Agent (B2BUA), choose this option. When the incoming request is received, Cisco Unified Communications Manager parses the Call-Info header, looks for the parameter, purpose=x-cisco-origIP, and uses the IP address or domain name and the signaling port number that is specified in the header to reroute the call to the SIP trunk that uses the IP address and port. If the parameter does not exist in the header or no SIP trunk is identified, the call occurs on the SIP trunk on which the call arrived. **Tip** This setting does not work for SIP trunks that are connected to a Cisco Unified Presence proxy server or SIP trunks that are connected to originating gateways in different Cisco Unified CM groups.
RSVP Over SIP	This field configures RSVP over SIP trunks. From the drop-down list box, choose the method that Cisco Unified Communications Manager uses to configure RSVP over SIP trunks: • Local RSVP—In a local configuration, RSVP occurs within each cluster, between the end point and the local SIP trunk, but not on the WAN link between the clusters. • E2E—In an end-to-end (E2E) configuration, RSVP occurs on the entire path between the end points, including within the local cluster and over the WAN.

Field	Description
Resource Priority Namespace List	Select a configured Resource Priority Namespace list from the drop-down menu. Configure the lists in the Resource Priority Namespace List menu that is accessed from **System** > **MLPP** > **Namespace**.
Fall back to local RSVP	Check this box if you want to allow failed end-to-end RSVP calls to fall back to local RSVP to establish the call. If this box is not checked, end-to-end RSVP calls that cannot establish an end-to-end connection fail.
SIP Rel1XX Options	This field configures SIP Rel1XX, which determines whether all SIP provisional responses (other than 100 Trying messages) get sent reliably to the remote SIP endpoint. Valid values follow: • Disabled—Disables SIP Rel1XX. • Send PRACK if 1XX contains SDP—Acknowledges a 1XX message with PRACK, only if the 1XX message contains SDP. • Send PRACK for all 1XX messages —Acknowledges all1XX messages with PRACK. If you set the RSVP Over SIP field to E2E, you cannot choose Disabled.
Video Call Traffic Class	Video Call Traffic Class determines the type of video endpoint or trunk that the SIP Profile is associated with. From the drop-down list box, select one of the following three options • Immersive—High-definition immersive video. • Desktop—Standard desktop video. • Mixed—A mix of immersive and desktop video. Cisco Unified Communications Manager Locations Call Admission Control (CAC) reserves bandwidth from two Locations video bandwidth pools, "Video Bandwidth" and/or "Immersive Bandwidth", depending on the type of call determined by the Video Call Traffic Class. Please refer to the Call Admission Control chapter of the *Cisco Unified Communications Manager System Guide* for more information.
Calling Line Identification Presentation	Select **Strict From URI presentation Only** to select the network provided identity. Select **Strict Identity Headers presentation Only** to select the user provided identity.

Field	Description
Deliver Conference Bridge Identifier	Check this check box for the SIP trunk to pass the b-number that identifies the conference bridge across the trunk instead of changing the b-number to the null value. The terminating side does not require that this field be enabled. Checking this check box is not required for Open Recording Architecture (ORA) SIP header enhancements to the Recording feature to work. Enabling this check box allows the recorder to coordinate recording sessions where the parties are participating in a conference.
Early Offer support for voice and video calls (insert MTP if needed)	Check this check box if you want to create a trunk that supports early offer. Early Offer configurations on SIP profile apply to SIP trunk calls. These configurations do not affect SIP line side calls. If this profile is shared between a trunk and a line, only the SIP trunk that uses the profile provides early offer. Because E2E RSVP provides an early offer by including an SDP in the initial INVITE, the early offer and E2E RSVP features are mutually exclusive on the SIP Profile Configuration window. When you choose E2E from the RSVP Over SIP drop-down list box, the Early Offer support for voice and video calls (insert MTP if needed) check box gets disabled. **Note** When checked, the Media Termination Required check box on the Trunk Configuration window overrides the early offer configuration on the associated SIP profile. The Cisco Unified Communications Manager sends the MTP IP address and port with a single codec in the SDP in the initial INVITE.
Send send-receive SDP in mid-call INVITE	Check this check box to prevent Cisco Unified Communications Manager from sending an INVITE a=inactive SDP message during call hold or media break during supplementary services. **Note** This check box applies only to early offer enabled SIP trunks and has no impact on SIP line calls. When you enable Send send-receive SDP in mid-call INVITE for an early offer SIP trunk in tandem mode, Cisco Unified Communications Manager inserts MTP to provide sendrecv SDP when a SIP device sends offer SDP with a=inactive or sendonly or recvonly in audio media line. In tandem mode, Cisco Unified Communications Manager depends on the SIP devices to initiate reestablishment of media path by sending either a delayed INVITE or mid-call INVITE with send-recv SDP. When you enable both Send send-receive SDP in mid-call INVITE and Require SDP Inactive Exchange for Mid-Call Media Change on the same SIP Profile, the Send send-receive SDP in mid-call INVITE overrides the Require SDP Inactive Exchange for Mid-Call Media Change, so Cisco Unified Communications Manager does not send an INVITE with a=inactive SDP in mid-call codec updates. For SIP line side calls, the Require SDP Inactive Exchange for Mid-Call Media Change check box applies when enabled. **Note** To prevent the SDP mode from being set to inactive in a multiple-hold scenario, set the Duplex Streaming Enabled clusterwide service parameter (**System** > **Service Parameters**) to True.

Field	Description
Allow Presentation Sharing using BFCP	If the box is checked, Cisco Unified Communications Manager is configured to allow supported SIP endpoints to use the Binary Floor Control Protocol to enable presentation sharing. The use of BFCP creates an additional media stream in addition to the existing audio and video streams. This additional stream is used to stream a presentation, such as a PowerPoint presentation from someone's laptop, into a SIP videophone. If the box is unchecked, Cisco Unified Communications Manager rejects BFCP offers from devices associated with the SIP profile by setting the BFCP application line and associated media line ports to 0 in the answering SDP message. This is the default behavior. **Note** BFCP is only supported on SIP networks. BFCP must be enabled on all SIP trunks, lines, and endpoints for presentation sharing to work. BFCP is not supported if the SIP line or SIP trunk uses MTP, RSVP, TRP or Transcoder. For more information on BFCP, refer to the *Cisco Unified Communications Manager System Guide*.
Allow iX Application Media	Check this check box to enable support for iX media channel.
Allow Passthrough of Configured Line Device Caller Information	Check this box to allow passthrough of configured line device caller information from the SIP trunk.
Reject Anonymous Incoming Calls	Check this box to reject anonymous incoming calls.
Reject Anonymous Outgoing Calls	Check this box to reject anonymous outgoing calls.
SIP OPTIONS Ping	
Enable OPTIONS Ping to monitor destination status for Trunks with service type "None (Default)"	Check this check box if you want to enable the SIP OPTIONS feature. SIP OPTIONS are requests to the configured destination address on the SIP trunk. If the remote SIP device fails to respond or sends back a SIP error response such as 503 Service Unavailable or 408 Timeout, Cisco Unified Communications Manager tries to reroute the calls by using other trunks or by using a different address. If this check box is not checked, the SIP trunk does not track the status of SIP trunk destinations. When this check box is checked, you can configure two request timers.

Field	Description
Ping Interval for In-service and Partially In-service Trunks (seconds)	This field configures the time duration between SIP OPTIONS requests when the remote peer is responding and the trunk is marked as In Service. If at least one IP address is available, the trunk is In Service; if all IP addresses are unavailable, the trunk is Out of Service. The default value specifies 60 seconds. Valid values range from 5 to 600 seconds.
Ping Interval for Out-of-service SIP Trunks (seconds)	This field configures the time duration between SIP OPTIONS requests when the remote peer is not responding and the trunk is marked as Out of Service. The remote peer may be marked as Out of Service if it fails to respond to OPTIONS, if it sends 503 or 408 responses, or if the Transport Control Protocol (TCP) connection cannot be established. If at least one IP address is available, the trunk is In Service; if all IP addresses are unavailable, the trunk is Out of Service. The default value specifies 120 seconds. Valid values range from 5 to 600 seconds.
Ping Retry Timer (milliseconds)	This field specifies the maximum waiting time before retransmitting the OPTIONS request. Valid values range from 100 to 1000 milliseconds. The default value specifies 500 milliseconds.
Ping Retry Count	This field specifies the number of times that Cisco Unified Communications Manager resends the OPTIONS request to the remote peer. After the configured retry attempts are used, the destination is considered to have failed. To obtain faster failure detection, keep the retry count low. Valid values range from 1 to 10. The default value specifies 6.

Synchronize SIP profile settings with SIP devices

To synchronize SIP devices with a SIP profile that has undergone configuration changes, perform the following procedure, which applies any outstanding configuration settings in the least-intrusive manner possible. (For example, a reset/restart may not be required on some affected devices.)

Procedure

Step 1 Choose **Device** > **Device Settings** > **SIP Profile**.
The Find and List SIP Profiles window displays.

Step 2 Choose the search criteria to use.

Step 3 Click Find.
The window displays a list of SIP Profiles that match the search criteria.

Step 4 Click the SIP profile to which you want to synchronize applicable SIP devices. The SIP Profile Configuration window displays.

Step 5 Make any additional configuration changes.

Step 6 Click Save.

Step 7 Click Apply Config.
The Apply Configuration Information dialog displays.

Step 8 Click OK.