Documentation Update for Defects

Command Line Interface Reference Guide

utils dbreplication clusterreset

This documentation update resolves CSCvf93618.

The **utils dbreplication clusterreset** command is deprecated, instead run **utils dbreplication reset** command to repair replication.

```
admin:utils dbreplication clusterreset

********************************************************************************************
This command is deprecated, please use 'utils dbreplication reset' to repair replication!
********************************************************************************************

Executed command unsuccessfully
```

For more details on **utils dbreplication reset** command, see the "Utils Commands" chapter in the *Command Line Interface Guide for Cisco Unified Communications Solutions* at https://www.cisco.com/c/en/us/support/unified-communications/unified-communications-manager-callmanager/products-maintenance-guides-list.html.

Security Guide

Certificates

This documentation update resolves CSCvg10775.

The following note is omitted from the "Security Overview" chapter in *Security Guide for Cisco Unified Communications Manager*.

Note The maximum supported size of certificate for DER or PEM is 4096 bits.

System Error Messages

CSCvg70867 Documentation Defect Update

The *System Error Messages for Cisco Unified Communications Manager* file is missing the following ENUM definitions for the 78XX and 88xx phones.

Value	Device Type
508	Cisco IP Phone 7821
509	Cisco IP Phone 7841
510	Cisco IP Phone 7861
544	Cisco IP Phone 8831
568	Cisco IP Phone 8841
569	Cisco IP Phone 8851
570	Cisco IP Phone 8861
36665	Cisco IP Phone 7811
36669	Cisco IP Phone 8821
36670	Cisco IP Phone 8811
36677	Cisco IP Phone 8845
36678	Cisco IP Phone 8865
36686	Cisco IP Phone 8851NR
36701	Cisco IP Phone 8865NR

CSCvd71818 Documentation Defect Update

The *System Error Messages for Cisco Unified Communications* file is missing some ENUM values for the **Reason For Out Of Service** parameter within the **LastOutOfServiceInformation** alarm. Following is a complete list:

Reason Code	Description
10	TCPtimedOut - The TCP connection to the Cisco Unified Communication Manager experienced a timeout error

Reason Code	Description
12	TCPucmResetConnection - The Cisco Unified Communication Manager reset the TCP connection
13	TCPucmAbortedConnection - The Cisco Unified Communication Manager aborted the TCP
14	TCPucmClosedConnection - The Cisco Unified Communication Manager closed the TCP connection
15	SCCPKeepAliveFailure - The device closed the connection due to a SCCP KeepAlive failure
16	TCPdeviceLostIPAddress - The connection closed due to the IP address being lost. This may be due to the DHCP Lease expiring or the detection of IP address duplication. Check that the DHCP Server is online and that no duplication has been reported by the DHCP Server
17	TCPdeviceLostIPAddress - The connection closed due to the IP address being lost. This may be due to the DHCP Lease expiring or the detection of IP address duplication. Check that the DHCP Server is online and that no duplication has been reported by the DHCP Server
18	TCPclosedConnectHighPriorityUcm - The device closed the TCP connection in order to reconnect to a higher priority Cisco Unified CM
20	TCPclosedUserInitiatedReset - The device closed the TCP connection due to a user initiated reset
22	TCPclosedUcmInitiatedReset - The device closed the TCP connection due to a reset command from the Cisco Unified CM
23	TCPclosedUcmInitiatedRestart - The device closed the TCP connection due to a restart command from the Cisco Unified CM
24	TCPClosedRegistrationReject - The device closed the TCP connection due to receiving a registration rejection from the Cisco Unified CM
25	RegistrationSuccessful - The device has initialized and is unaware of any previous connection to the Cisco Unified CM
26	TCPclosedVlanChange - The device closed the TCP connection due to reconfiguration of IP on a new Voice VLAN
27	Power Save Plus
30	Phone Wipe (wipe from CUCM)
31	Phone Lock (lock from CUCM)
32	TCPclosedPowerSavePlus - The device closed the TCP connection in order to enter Power Save Plus mode

Reason Code	Description
100	ConfigVersionMismatch - The device detected a version stamp mismatch during registration Cisco Unified CM
101	Config Version Stamp Mismatch
102	Softkeyfile Version Stamp Mismatch
103	Dial Plan Mismatch
104	TCPclosedApplyConfig - The device closed the TCP connection to restart triggered internally by the device to apply the configuration changes
105	TCPclosedDeviceRestart - The device closed the TCP connection due to a restart triggered internally by the device because device failed to download the configuration or dial plan file
106	TCPsecureConnectionFailed - The device failed to setup a secure TCP connection with Cisco Unified CM
107	TCPclosedDeviceReset - The device closed the TCP connection to set the inactive partition as active partition, then reset, and come up from the new active partition
108	VpnConnectionLost - The device could not register to Unified CM because VPN connectivity was lost 109 IP Address Changed
109	IP Address Changed
110	Application Requested Stop (service control notify to stop registering)
111	Application Requested Destroy
114	Last Time Crash
200	ClientApplicationClosed - The device was unregistered because the client application was closed
201	OsInStandbyMode - The device was unregistered because the OS was put in standby mode
202	OsInHibernateMode - The device was unregistered because the OS was put in hibernate mode
203	OsInShutdownMode - The device was unregistered because the OS was shut down
204	ClientApplicationAbort - The device was unregistered because the client application crashed
205	DeviceUnregNoCleanupTime - The device was unregistered in the previous session because the system did not allow sufficient time for cleanup
206	DeviceUnregOnSwitchingToDeskphone - The device was unregistered because the client requested to switch from softphone to deskphone control

Reason Code	Description
207	DeviceUnregOnSwitchingToSoftphone - The device is being registered because the client requested to switch from deskphone control to softphone
208	DeviceUnregOnNetworkChanged - The device is being unregistered because the client detected a change of network
209	DeviceUnregExceededRegCount - The device is being unregistered because the device has exceeded the maximum number of concurrent registrations
210	DeviceUnregExceededLoginCount - The device is being unregistered because the client has exceeded the maximum number of concurrent logons

Missing Device Type ENUM Values

This update is for CSCvg70867.

The *System Error Messages for Cisco Unified Communications Manager* file is missing the following ENUM definitions for the 78XX and 88xx phones.

Value	Device Type
508	Cisco IP Phone 7821
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36677	Cisco IP Phone 8845
36678	Cisco IP Phone 8865
36686	Cisco IP Phone 8851NR
36701	Cisco IP Phone 8865NR

Missing Reason Codes for LastOutOfServiceInformation Alarms

This update is for CSCvd71818.

The *System Error Messages for Cisco Unified Communications* file is missing some ENUM values for the **Reason For Out Of Service** parameter within the **LastOutOfServiceInformation** alarm. Following is a complete list:

Reason Code	Description
10	TCPtimedOut - The TCP connection to the Cisco Unified Communication Manager experienced a timeout error
12	TCPucmResetConnection - The Cisco Unified Communication Manager reset the TCP connection
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14	TCPucmClosedConnection - The Cisco Unified Communication Manager closed the TCP connection
15	SCCPKeepAliveFailure - The device closed the connection due to a SCCP KeepAlive failure
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25	RegistrationSuccessful - The device has initialized and is unaware of any previous connection to the Cisco Unified CM
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Reason Code	Description
27	Power Save Plus
30	Phone Wipe (wipe from CUCM)
31	Phone Lock (lock from CUCM)
32	TCPclosedPowerSavePlus - The device closed the TCP connection in order to enter Power Save Plus mode
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104	TCPclosedApplyConfig - The device closed the TCP connection to restart triggered internally by the device to apply the configuration changes
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106	TCPsecureConnectionFailed - The device failed to setup a secure TCP connection with Cisco Unified CM
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114	Last Time Crash
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201	OsInStandbyMode - The device was unregistered because the OS was put in standby mode
202	OsInHibernateMode - The device was unregistered because the OS was put in hibernate mode
203	OsInShutdownMode - The device was unregistered because the OS was shut down

Reason Code	Description
204	ClientApplicationAbort - The device was unregistered because the client application crashed
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206	DeviceUnregOnSwitchingToDeskphone - The device was unregistered because the client requested to switch from softphone to deskphone control
207	DeviceUnregOnSwitchingToSoftphone - The device is being registered because the client requested to switch from deskphone control to softphone
208	DeviceUnregOnNetworkChanged - The device is being unregistered because the client detected a change of network
209	DeviceUnregExceededRegCount - The device is being unregistered because the device has exceeded the maximum number of concurrent registrations
210	DeviceUnregExceededLoginCount - The device is being unregistered because the client has exceeded the maximum number of concurrent logons

Online Help for Cisco Unified Communications Manager

DHCP Subnet Setup Tips

This documentation update resolves CSCve07463.

The DHCP subnet setup tip is incorrect in the *Cisco Unified CM Administration Online Help*. The correct information for “DHCP Subnet Setup Tips” is as follows:

Changes to the server configuration do not take effect until you restart DHCP Monitor Service.

Insufficient Information About Opus Codec

This documentation update resolves CSCva48193.

The “System Menu” chapter in *Cisco Unified CM Administration Online Help* contains insufficient information about the **Opus Codec** field. The following note is omitted from the guide.

Note The Advertise G.722 Codec service parameter in the **Enterprise Parameters Configuration** window should be set to **Enabled** for the SIP devices to use Opus codec. For more information on enterprise parameters, see the *System Configuration Guide for Cisco Unified Communications Manager* at http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/11_5_1/sysConfig/CUCM_BK_SE5DAF88_00_cucm-system-configuration-guide-1151.html.

Incorrect Time Period Example

This documentation update resolves CSCvb74432.

The time period documentation contains an incorrect example that can cause configuration problems. It suggests to use a date range for a single day time period: "Choose a Year on value of Jan and 1 and an until value of Jan and 1 to specify January 1st as the only day during which this time period applies."

That is incorrect; please avoid using this example for the "Year on...until" option for time periods.

Insufficient Information About Time Schedule

This documentation update resolves CSCvd75418.

The Time Schedule Settings topic in the “Call Routing Menu” chapter of the *Cisco Unified CM Administration Online Help* contains insufficient information about the selected time period for a day. The following scenario is omitted from the guide:

Table 1: Time Schedule Settings

Field	Description
Time Period Information	

Field	Description
Selected Time Periods	**Scenario:** If multiple time periods are associated to a time schedule and the time periods does not overlap. However, overlap in a day, then the single day period takes precedence and other time periods for that day is ignored. Example 1: Three time periods are defined in the time schedule: Range of Days: Jan 1 - Jan 31: 09:00 - 18:00 Day of Week: Mon - Fri: 00:00 - 08:30 Day of Week: Mon - Fri: 18:30 - 24:00 In this case, even though the times are not overlapping, Range of Days is ignored for a call on Wednesday at 10:00. Example 2: Three time periods are defined in the time schedule: Single Day: Jan 3 2017 (Tues): 09:00 - 18:00 Day of Week: Mon - Fri: 00:00 - 08:30 Day of Week: Mon - Fri: 18:30 - 24:00 In this case, even though the times are not overlapping, Day of Week is ignored for a call on Jan 3 at 20:00. **Note** If Day of Year settings is configured, then the Day of Year settings is considered for the entire day (24 hours) and Day of Week settings, Range of Days settings for that particular day is ignored.

Insufficient Information on LDAP User Authentication

This documentation update resolves CSCvc30013.

The *LDAP Authentication Settings* in the *System Menu* chapter in *Cisco Unified CM Administration Online Help* contains insufficient information about LDAP User Authentication. The following note is omitted from the guide:

Note You can do LDAP User Authentication using the IP address or the hostname. When IP address is used while configuring the LDAP Authentication, LDAP configuration needs to be made the IP address using the command `utils ldap config ipaddr`. When hostname is used while configuring the LDAP Authentication, DNS needs to be configured to resolve that LDAP hostname.

Remote Destination Configuration Page In the OLH Needs To Be Updated

This documentation update resolves CSCvb88447.

The "Device Menu" chapter in Cisco Unified CM Administration Online Help contains incorrect information in the “Remote Destination Configuration Settings” help page. The following information was either incorrect or omitted in the relevant fields.

- The **Timer Information** field has incorrect information in the help page. It states the time in “milliseconds”, the correct time is set in “seconds”.
- The **Timer Information**section lists incorrect order in the help page. The correct orders of the fields are: **Delay Before Ringing Timer**, **Answer Too Soon Timer**, and **Answer Too Late Timer**.
- The **Owner User ID** field is omitted. Following is the description for this field:
    - **Owner User ID**— From drop-down list, choose the appropriate end user profile to which the remote destination profile can be associated later.

SIP Profile Field Descriptions Are Missing

The online help in Cisco Unified Communications Manager Releases 11.5(1)SU3 and SU4 contains an error in the SIP Profile Settings topic for the online help. This topic may be missing the SIP Profile field descriptions. If this is the case, refer to the following topic for the list of field descriptions.

SIP Profile Settings

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

Table 2: 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	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 (<>).

Field	Description
Default MTP Telephony Event Payload Type	Specifies the default payload type for RFC2833 telephony event. See RFC 2833 for more information. Usually, the default value specifies the appropriate payload type. Ensure 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 Unified Communications Manager. • For the calling SIP trunk, the Media Termination Point Required check box is checked on the **SIP Trunk Configuration** window.
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	Specifies the maximum amount of bandwidth that is 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

Field	Description
User-Agent and Server header information	Indicates how Unified Communications Manager handles the User-Agent and Server header information in a SIP message. Choose one of the following three options: • Send Unified Communications Manager 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. 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.

<table>
<tr><th>Field</th><th>Description</th></tr>
<tr><td>Dial String Interpretation</td><td>Determine if the SIP identity header is a directory number or directory URI.

As directory numbers and directory URIs are saved in different database lookup tables, 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 list:

• Always treat all dial strings as URI addresses—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 that are treated as URI addresses)—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 that are treated as URI addresses)—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, Unified Communications Manager always treats the dial string as a number regardless of what option you choose for the Dial String Interpretation field.</td></tr>
<tr><td>Accept Audio Codec Preferences in Received Offer</td><td>Allows to select On to enable Unified Communications Manager to honor the preference of audio codecs in the received offer and preserve it while processing. Select Off to enable Unified Communications Manager to ignore the preference of audio codecs in the received offer and apply the locally configured Audio Codec Preference List. The default will select the service parameter configuration.

Note If this is enabled in both incoming and outgoing trunks then the same codec preference list should be associated with both trunks else it might result in a different codec being negotiated towards both sides leading to audio issues.</td></tr>
</table>

Field	Description
Require SDP Inactive Exchange for Mid-Call Media Change	Designates how Unified Communications Manager handles mid-call updates to codecs or connection information such as IP address or port numbers. If the check box is selcted, during mid-call codec or connection updates 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. If the check box is unchecked, 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. **Note** For early offer or best effort early offer enabled SIP trunks, this parameter will be overridden by the Send send-receive SDP in mid-call INVITE parameter.
Confidential Access Level Headers	Determines the inclusion of Confidential Access Level headers in INVITE and 200 OK messages. Valid values are as follows: • Disabled—CAL headers are not included. • Preferred—CAL headers are included and confidential-access-level tag is added in the Supported header. • Required— CAL headers are included and confidential-access-level tag is added in the Require and Proxy-Require headers.
SDP Transparency Profile	Allows you to choose one of the following options for SIP profile : • **None**—Choose this option for Unified Communications Manager to filter out known SDP attributes only. By default, this option is selected. • **Pass all unknown SDP attributes**—Choose this option for media adaptation and resilience (MARI). To ensure that the session level MARI attributes pass the unknown attributes through Unified Communications Manager, choose this value on the SIP profile, which is associated with both the originating device and the terminating device.

Field	Description
Redirect by Application	Checking this check box and configuring this SIP Profile on the SIP trunk allows the 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, Unified Communications Manager signals the calling phone to play local ringback if SDP is not received in the 180 response. If SDP is included in the 180 response, instead of playing ringback locally, 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	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.
Offer valid IP and Send/Receive mode only for T.38 Fax Relay	If this checkbox is checked, this SIP profile on the trunk allows you to send a fax offer with a valid IP address and with Send Receive SDP mode. If this checkbox is not checked, this SIP profile on the trunk allows you to send a fax offer with a null IP address and with Send Receive SDP mode. This parameter applies only to trunks, not phones that are running SIP or other endpoints. It applies only for T38 fax relay and, by default, this checkbox is unchecked.

Field	Description
Enable ANAT	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, 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, 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, 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), 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: • Unified Communications Manager sends an IPv4 address in the SDP for SIP trunks with an IP Addressing Mode of IPv4 Only. • 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, 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. • For dual-stack SIP trunks, 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	Designates how 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 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, 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	Enables 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 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, 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 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 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	Specifies to check this box 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.
Enable External QoS	Specifies to check this box to configure this SIP Profile for external QoS support. With this feature enabled, you can use an APIC-EM Controller to manage QoS for SIP media flows for devices that use this SIP Profile. The default value is unchecked. **Note** This check box appears only if the **External QoS Enable** service parameter is set to **True**.
Parameters Used in Phone	
Timer Invite Expires (seconds)	Sspecifies 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)	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)	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, 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, Unified Communications Manager responds with a 200 OK that includes the Keepalive Interval value for Expires. **Note** For mobile phones that are running SIP, 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)	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)	Specifies the highest value, in milliseconds, of the retransmission timer for SIP messages. Valid values include any positive number. Default specifies 4000.
Retry INVITE	Specifies the maximum number of times that an INVITE request gets retransmitted. Valid values include any positive number. Default specifies 6.
Retry Non-INVITE	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.
Media Port Ranges	Specifies to click the radio button that corresponds to how you want to manage QoS for audio and video calls for devices that are associated to this SIP Profile • **Common Port Range for Audio and Video**—Choose this option if you want to use a common port range that can handles both the audio and video media stream. • **Separate Port Ranges for Audio and Video**—Choose this option if you want to set up a distinct port range for the audio stream and a distinct port range for the video stream.
Start Media Port	Designates the start real-time protocol (RTP) port for media. Media port ranges from 2048 to 65535. Default specifies 16384. This field appears when you select **Common Port Range for Audio and Video** as the **Media Port Range**.

Field	Description
Stop Media Port	Designates the stop real-time protocol (RTP) port for media. Media port ranges from 2048 to 65535. Default specifies 32766. This field appears when you select **Common Port Range for Audio and Video** for the **Media Port Range**.
Start Audio Port	Allows you to create a port range for audio by entering the start of the port range. For example, 16384. The audio port range cannot overlap the video port range. This field appears when you select **Separate Port Ranges for Audio and Video** for the **Media Port Range**.
Stop Audio Port	Allows you to enter the ending of the port range for audio calls. The audio port range must not overlap the video port range. For example, 32766. This field appears when you select **Separate Port Ranges for Audio and Video** for the **Media Port Range**.
Start Video Port	Allows you to create a port range for the video stream of a video call by entering the beginning of the port range. For example, 32767. The video port range cannot overlap with the audio port range. This field appears when you select **Separate Port Ranges for Audio and Video** for the **Media Port Range**.
Stop Video Port	Allows you to enter the ending of the port range for audio calls. The audio port range must not overlap the video port range. This field appears when you select **Separate Port Ranges for Audio and Video** for the **Media Port Range**.
DSCP for Audio Calls	Allows you to select the value that you want to assign as the DSCP value for audio-only calls. The Default Option is to use the value of the DSCP for Audio Calls service parameter.
DSCP for Video Calls	Allows you to select the value that you want to assign as the DSCP value for video calls. The Default Option is to use the value of the DSCP for Video Calls service parameter.
DSCP for Audio Portion of Video Calls	Allows you to select the value that you want to assign as the DSCP value for audio portion of a video call. The default option is to use the value that is configured in the DSCP for Audio Portion of Video Calls service parameter. **Note** If you choose a different DSCP value for audio portion of video calls than you configured for DSCP Video Calls, it could mean that the audio and video streams within a single video call could have different DSCP markings and different QoS policy control, which could result in lip sync issues that result from network bandwidth issues.
DSCP for TelePresence Calls	Allows you to select the value that you want to assign as the DSCP value for TelePresence calls. The default option is to use the value of the DSCP for TelePresence Calls service parameter.

Field	Description
DSCP for Audio Portion of TelePresence Calls	Allows you to select the value that you want to assign as the DSCP value for the audio portion of TelePresence calls. The default option is to use the value of the DSCP for TelePresence Calls service parameter.
Call Pickup URI	Provides a unique address that the phone that is running SIP sends to Unified Communications Manager to invoke the call pickup feature.
Call Pickup Group Other URI	Provides a unique address that the phone that is running SIP sends to Unified Communications Manager to invoke the call pickup group other feature.
Call Pickup Group URI	Provides a unique address that the phone that is running SIP sends to Unified Communications Manager to invoke the call pickup group feature.
Meet Me Service URI	Provides a unique address that the phone that is running SIP sends to Unified Communications Manager to invoke the meet me conference feature.
User Info	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	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	Indicates the call on hold status. For example, 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.

Field	Description
Anonymous Call Block	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	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	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	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)	Specifies the interval between keepalive messages that are sent to the backup Unified Communications Manager to ensure that it is available in the event that a failover is required. Unified Communications Manager requires a keepalive mechanism to support redundancy.
Timer Subscribe Expires (seconds)	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.

Field	Description
Timer Subscribe Delta (seconds)	Allows you to 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	Allows you to 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.
Off Hook to First Digit Timer (microseconds)	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 - 150,000 microseconds. Default specifies 15,000 microseconds.
Call Forward URI	Provides a unique address that the phone that is running SIP sends to Unified Communications Manager to invoke the call forward feature.
Abbreviated Dial URI	Provides a unique address that the phone that is running SIP sends to 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. 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	Determines whether the Unified Communications Managers 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 IM and Presence Services 7941/61/70/71/11 when they are in SRST mode only.
RFC 2543 Hold	Enables setting connection address to 0.0.0.0 per RFC2543 when call hold is signaled to Unified Communications Manager. This allows backward compatibility with endpoints that do not support RFC3264.
Semi Attended Transfer	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	Enables Voice Activation Detection (VAD). When VAD is enabled, media is not transmitted until the voice is detected.

<table>
<tr><th>Field</th><th>Description</th></tr>
<tr><td>Stutter Message Waiting</td><td>Enables 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.</td></tr>
<tr><td>MLPP User Authorization</td><td>Enable MLPP User Authorization. MLPP User Authorization requires the phone to send in an MLPP username and password.</td></tr>
<tr><td colspan="2">Normalization Script</td></tr>
<tr><td>Normalization Script</td><td>Alows you to 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.

Caution A normalization script in the SIP profile is only valid for non-trunk devices.</td></tr>
<tr><td>Parameter Name/Parameter Value</td><td>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 list before you can enter parameter names and values.</td></tr>
<tr><td>Enable Trace</td><td>Enables 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 We recommend that you only enable tracing while debugging a script. Tracing impacts performance and should not be enabled under normal operating conditions.</td></tr>
<tr><td colspan="2">Incoming Requests FROM URI Settings</td></tr>
<tr><td>Caller ID DN</td><td>Allows you to 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 +.</td></tr>
<tr><td>Caller Name</td><td>Allows you to enter a caller name to override the caller name that is received from the originating SIP Device.</td></tr>
</table>

Field	Description
Trunk Specific Configuration	
Reroute Incoming Request to new Trunk based on	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 configured on the SIP trunk. After the Unified Communications Manager accepts the call, it uses the configuration for this setting to determine whether the call should get rerouted to another trunk. You can select any of the following method that Unified Communications Manager uses to identify the SIP trunk where the call is rerouted: • Never—If the SIP trunk matches the IP address of the originating device, choose this option, which equals the default setting. The Unified Communications Manager identifies the trunk by using the source IP address of the incoming packet and the signaling port number, do 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. The Unified Communications Manager analyzes 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 number. 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, the Unified Communications Manager analyzes the Call-Info header, search for the parameter purpose=x-cisco-origIP, and uses the IP address or domain name specified in the header to reroute the call to the SIP trunk that uses the IP address and port. The listening port on the inbound trunk and the trunk targeted by the x-cisco-origIP value need to match for the targeted trunk to be used in the call . 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. **Note** You cannot set these parameters as they are not supported in Unified Communications Manager for secure calls. This setting does not work for SIP trunks that are connected to a IM and Presence Service proxy server or SIP trunks that are connected to originating gateways in different Unified CM groups.
Resource Priority Namespace List	Allows you to select a configured Resource Priority Namespace list. Configure the lists in the Resource Priority Namespace List menu that is accessed from **System** > **MLPP** > **Namespace**.

Field	Description
SIP Rel1XX Options	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. **Note** You need not configure the above field if **Connect Inbound Call before Playing Queuing Announcement** checkbox is checked in the Trunk Specific Configuration.
Session Refresh Method	Session Timer with Update: The session refresh timer allows for periodic refresh of SIP sessions, which allows the Unified Communications Manager and remote agents to determine whether the SIP session is still active. Prior to Release 10.01, when the Unified Communications Manager received a refresh command, it supported receiving either Invite or Update SIP requests to refresh the session. When the Unified Communications Manager initiated a refresh, it supported sending only Invite SIP requests to refresh the session. With Release 10.01, this feature extends the refresh capability so that Unified Communications Manager can send both Update and Invite requests. Specify whether Invite or Update should be used as the Session Refresh Method. **Invite** (default): **Note** Sending a mid-call Invite request requires that an offer SDP be specified in the request. This means that the far end must send an answer SDP in the Invite response. **Update**: Unified Communications Manager sends a SIP Update request, if support for the Update method is specified by the far end of the SIP session either in the Supported or Require headers. When sending the Update request, the Unified Communications Manager includes an SDP. This simplifies the session refresh since no SDP offer/answer exchange is required. **Note** If the Update method is not supported by the far end of the SIP session, the Unified Communications Manager continues to use the Invite method for session refresh.

Field	Description
Early Offer support for voice and video calls	Configures Early Offer support for voice and video calls. When enabled, Early Offer support includes a session description in the initial INVITE for outbound calls. Early Offer configuration settings on SIP profile apply only to SIP trunk calls. These configuration settings do not affect SIP line side calls. If this profile is shared between a trunk and a line, only a SIP trunk that uses the profile is affected by these settings. The Media Transfer Point (MTP) Required check box on the Trunk Configuration window, if enabled, overrides the early offer configuration on the associated SIP profile. Unified Communications Manager sends the MTP IP address and port with a single codec in the SDP in the initial INVITE. Select one of the following three options: • Disabled (Default value) - Disables Early Offer; no SDP will be included in the initial INVITE for outbound calls. • Best Effort (no MTP inserted) • Provide Early Offer for the outbound call only when caller side's media port, IP and codec information is available. • Provide Delayed Offer for the outbound call when caller side's media port, IP and codec information is not available. No MTP is inserted to provide Early Offer in this case. • Mandatory(insert MTP if needed) - Provide Early Offer for all outbound calls and insert MTP when caller side's media port, IP and codec information is not available.
Video Call Traffic Class	Determines the type of video endpoint or trunk that the SIP Profile is associated with. From the list, select one of the following three options • Immersive—High-definition immersive video. • Desktop—Standard desktop video. • Mixed—A mix of immersive and desktop video. 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.
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.

<table>
<tr><th>Field</th><th>Description</th></tr>
<tr><td>Deliver Conference Bridge Identifier</td><td>Allows 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.</td></tr>
<tr><td>Early Offer support for voice and video calls (insert MTP if needed)</td><td>Allows 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.

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 Unified Communications Manager sends the MTP IP address and port with a single codec in the SDP in the initial INVITE.</td></tr>
<tr><td>Send send-receive SDP in mid-call INVITE</td><td>Allows you to prevent 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 or best 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 or best early offer SIP trunk in tandem mode, 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, 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 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.</td></tr>
</table>

Field	Description
Allow Presentation Sharing using BFCP	Allows the 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, 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.
Allow iX Application Media	Enables support for iX media channel.
Allow Passthrough of Configured Line Device Caller Information	Allows passthrough of configured line device caller information from the SIP trunk.
Reject Anonymous Incoming Calls	Allows to reject anonymous incoming calls.
Reject Anonymous Outgoing Calls	Allows to reject anonymous outgoing calls.

Field	Description
Allow multiple codecs in answer SDP	Applies when incoming SIP signals do not indicate support for multiple codec negotiation and Unified Communications Manager can finalize the negotiated codec. When this check box is checked, the endpoint behind the trunk is capable of handling multiple codecs in the answer SDP. For example, an endpoint that supports multiple codec negotiation calls the SIP trunk and Unified Communications Manager sends a Delay Offer request to a trunk. The endpoint behind the trunk returns all support codecs without the Contact header to indicate the support of multiple codec negotiation. In this case, Unified Communications Manager identifies the trunk as capable of multiple codec negotiation and sends SIP response messages back to both endpoints with multiple common codecs. When this check box is unchecked, Unified Communications Manager identifies the endpoint behind the trunk as incapable of multiple codec negotiation, unless indicated otherwise by SIP contact header URI. Unified Communications Manager continues the call with single codec negotiation. Configure **Allow multiple codecs in answer SDP** for the following: • Third-party SIP endpoints that support this capability • SIP trunks to third-party call controls servers that uniformly support this capability for all endpoints Do not configure this capability for SIP intercluster trunks to Cisco SME or other Unified Communications Manager systems.
Send ILS Learned Destination Route String	Allows the calls that Unified Communications Manager routes to a learned directory URI, learned number, or learned pattern, Unified Communications Manager adds the *x-cisco-dest-route-string* header to outgoing SIP INVITE and SUBSCRIBE messages and inserts the destination route string into the header. When this check box is unchecked, Unified Communications Manager does not add the *x-cisco-dest-route-string* header to any SIP messages. The *x-cisco-dest-route-string* header allows Unified Communications Manager to route calls across a Unified Border Element.
Connect Inbound Call before Playing Queuing Announcement	Allows you to send the carrier a CONNECT message before playing the hunt group announcements. You should enable this feature if the carrier trunk does not support in-band call status updates or if external callers report that they are unable to hear hunt group announcements.
SIP OPTIONS Ping	

Field	Description
Enable OPTIONS Ping to monitor destination status for Trunks with service type “None (Default)”	Allows you 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, Unified Communications Manager reroute the calls using other trunks or using a different address. The OPTIONS ping interval value for In-service and Partially In-service ranges from 5 to 600 seconds. The default value is 60 seconds. A SIP trunk is set to In-service when it receives a success response from the peer. If the peer fails to respond due to some errors, then the status is set to Out-of-service. The SIP trunk does not know the peer status until the next time OPTIONS ping is sent. If the SIP trunk sends any message between the ping interval and if the peer destination is Out-of service because of any error, the message results in failure. You can change the ping timer to a smaller value if required. If this check box is unchecked, the SIP trunk does not track the status of SIP trunk destinations. If this check box is checked, you can change the ping timer to a smaller value if required.
Ping Interval for In-service and Partially In-service Trunks (seconds)	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)	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)	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	Specifies the number of times that 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.