Cisco Unified SRST Administration Guide (All Versions)

First Published: 2022-12-14

Last Modified: 2024-03-30

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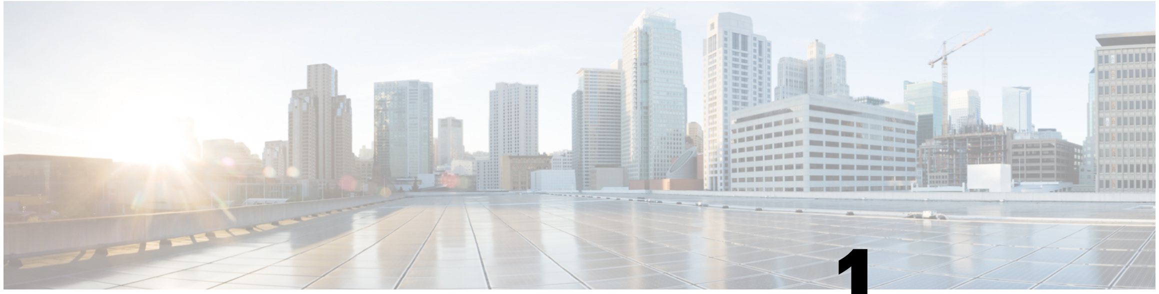


CHAPTER 1

Cisco Unified Survivable Remote Site Telephony Feature Roadmap

This chapter contains a list of Cisco Unified Survivable Remote Site Telephony (Cisco Unified SRST) features and the location of feature documentation.

Use Cisco Feature Navigator to find information about platform support and Cisco IOS software image support. Access Cisco Feature Navigator at http://www.cisco.com/go/fn. You must have an account on Cisco.com. If you do not have an account or have forgotten your username or password, click **Cancel** at the login dialog box and follow the instructions that appear.

Documentation Organization

This document consists of the following chapters or appendixes as shown in the following table .

Chapter or Appendix	**Description**
Cisco Unified SRST Feature Overview, on page 41	Gives a brief description of Cisco Unified SRST and provides information on the supported platforms and Cisco Unified IP Phones. In addition, it describes any prerequisites or restrictions that should be addressed before Cisco Unified SIP SRST is configured.
Setting Up the Network, on page 153	Describes how to set up a Cisco Unified SRST system to communicate with your network.
Enhanced SRST, on page 121	Describes how to configure the Cisco Unified Enhanced SRST feature in your network.
Cisco Unified SIP SRST, on page 163	Describes the features for Cisco Unified SIP SRST Version 4.1 and provides the associated configuration procedures.
Setting Up Cisco Unified IP Phones using SCCP, on page 173	Describes how to set up the basic Cisco Unified SRST phone configuration.

Chapter or Appendix	Description
Setting Up Cisco Unified IP Phones using SIP, on page 193	Describes features available in Version 3.0 that are also necessary for Version 3.4. Features include instructions on how to provide a backup to an external SIP call control (IP-PBX) by providing basic registrar services. These services are used by a SIP IP phone in the event of a WAN connection outage when the SIP phone is unable to communicate with its primary SIP proxy.
Configuring Call Handling, on page 213	Describes how to configure incoming and outgoing calls.
Configure Secure SRST for SCCP and SIP, on page 265	Describes the Secure SRST security functionality to the Cisco Unified SRST.
Integrating Voice Mail with Cisco Unified SRST, on page 363	Describes how to set up voicemail.
Setting Video Parameters, on page 385	Describes how to set up video parameters.
Monitoring and Maintaining Cisco Unified SRST, on page 399	Provides a list of useful show commands for monitoring and maintaining Cisco Unified SRST.
Appendix A: Configuring Cisco Unified SIP SRST Features Using Redirect Mode, on page 401	Describes features using redirect mode, which applies to version 3.0 only.
Appendix B: Integrating Cisco Unified Communications Manager and Cisco Unified SRST to Use Cisco Unified SRST as a Multicast MOH Resource, on page 409	Describes how to configure Cisco Unified CM and Cisco Unified SRST to enable multicast music-on-hold (MOH).

Feature Roadmap

The following table provides a feature history summary of Unified SRST features.

Cisco Unified SRST	Cisco IOS Release	Enhancements or Modifications
Version 14.4	Cisco IOS XE 17.14.1a	TLS version 1.3 support—TLS version 1.3 and associated ciphers support for secure SIP and SCCP SRST: • AES128_GCM_SHA256 • AES256_GCM_SHA384 • CHACHA20_POLY1305_SHA256

Cisco Unified SRST	Cisco IOS Release	Enhancements or Modifications
Version 14.3	Cisco IOS XE Dublin 17.11.1a Cisco IOS XE Cupertino 17.9.3a	Webex Survivability Gateway Mode, on page 45—Webex Survivability Gateway mode now supports survivability for Webex Calling endpoints and Unified SRST-based survivability for on-premises endpoints that register to Cisco Unified Communications Manager. Site Survivability for Webex Calling—Configure a Webex Survivability Gateway to provide a calling fallback service for Webex Calling endpoints.
Version 14.3	Cisco IOS XE Dublin 17.10.1a	YANG model enhancements for Unified SRST: Programmability Guide for Cisco IOS XE Unified Communications VoIP Products https://www.cisco.com/c/en/us/td/docs/routers/sdwan/command/sdwan-cr-book.html
Version 14.3	Cisco IOS XE Cupertino 17.9.1a	Webex Managed Gateway Command Reference Introduced the following commands as part of Cisco Webex Calling Branch Survivability: • **mode webex-sgw** • **voice register webex-sgw sync** • **show voice register webex-sgw users** Modified the following command as part of Cisco Webex Calling Branch Survivability: • **id (voice register pool)** modified to include phone-number *e164-number* and extension-number *extension-number*
Version 14.3	Cisco IOS XE Cupertino 17.9.1a	CDR Accounting Overview Configuring File Accounting gw-accounting
Version 14.2	Cisco IOS XE Cupertino 17.8.1a	SHA2-Cipher-Only Mode for Unified Secure SRST, on page 287
Version 14.2	Cisco IOS XE Cupertino 17.8.1a	SIP OAuth Client Registration for Unified Secure SRST, on page 279
Version 14.1	Cisco IOS XE Bengaluru 17.6.1a	Programmability Guide for Cisco IOS XE Unified Communications VoIP Products
Version 14.1	Cisco IOS XE Bengaluru 17.5.1a Release	• Support for Unified SRST on Cisco 1100 Integrated Services Router • Support for Unified SRST and E-SRST on Cisco 8200L Catalyst Series Edge Platforms

Cisco Unified SRST	Cisco IOS Release	Enhancements or Modifications
Version 14.1	Cisco IOS XE Bengaluru 17.4.1a Release	• Support for Unified SRST and E-SRST on Cisco 8200 Catalyst Series Edge Platforms • Smart Licensing Using Policy—Licensing • Smart Licensing Using Policy—Licensing
Version 14.1	Cisco IOS XE Amsterdam 17.3.2 Release	• Support for Unified SRST and E-SRST on Cisco 8300 Catalyst Series Edge Platforms • Smart Licensing Using Policy—Licensing • Smart Licensing Using Policy—Licensing
Version 12.8	Cisco IOS XE Amsterdam 17.2.1r	• Cisco Jabber with Unified SRST • VRF Support for Unified SRST • Support for YANG Models in Unified SRST
Version 12.7	Cisco IOS XE Amsterdam 17.1.1	Support for maximum number of devices in Cisco 4451 and 4461 Integrated Services Routers was increased from 1500 to 2000
Version 12.6	Cisco IOS XE Gibraltar 16.11.1a	• Simple Network Management Protocol (SNMP) Support for Unified SRST • Toll Fraud Prevention for SIP Line Side on Unified SRST • Unified SRST, Unified E-SRST, and Unified Secure SRST Password Policy
Version 12.5	Cisco IOS XE Gibraltar 16.10.1a	Support for Unified SRST on Cisco 4461 Integrated Services Routers
Version 12.3	Cisco IOS XE Fuji 16.9.1	Secure SCCP SRST Support
Version 12.2	Cisco IOS XE Fuji 16.8.1	Unified E-SRST with Support for Voice Hunt Group
Version 12.1	Cisco IOS XE Fuji 16.7.1	• Licensing • Secure SCCP SRST Support • Unified SRST and Unified Border Element Co-location
Version 12.0	Cisco IOS XE Everest 16.6.1	IPv6 Support for Unified SRST SIP IP Phones

Cisco Unified SRST	Cisco IOS Release	Enhancements or Modifications
Version 11.0	15.6(1)T	• Support for Cisco IP Phone 7811 • Support for Cisco IP Phones 8811, 8831, 8841, 8845, 8865, 8851, 8851NR, 8861 • Support for Cisco ATA-190 Phones
Version 10.5	15.4(3)M	• Setting Up the Network • Support for Cisco Unified DX650 SIP IP Phones • Support for Cisco Unified 78xx SIP IP Phones • Support for Cisco IP Phones 88xx, 8941, 8945, and 8961
Version 10.0	15.3(3)M	• Cisco Jabber for Windows • SIP: Configure Unified E-SRST
Version 9.5	15.3(2)T	• After-hour Pattern Blocking Support for Regular Expressions • Call Park Recall Enhancement • Display Support for Name of Called Voice Hunt Groups • Preventing Local-Call Forwarding to Final Agent in Voice Hunt Groups • Trunk-to-Trunk Transfer Blocking for Toll Fraud Prevention on Cisco Unified SIP IP Phones
Version 9.1	15.2(4)M	• Key Expansion Module Support for Cisco Unified SIP IP Phones • Enhancement in Speed-Dial Support • Voice Hunt Group Support
Version 9.0	15.2(2)T	• Support for Cisco Unified 6901 and 6911 SIP IP Phones • Support for Cisco Unified 6921, 6941, 6945, and 6961 SIP IP Phones • Support for Cisco Unified 8941 and 8945 SIP IP Phones • Multiple Calls Per Line • Voice and Fax Support on Cisco ATA-187
Version 8.8	15.2(1)T	Support for Cisco Unified 6945, 8941, and 8945 SCCP IP Phones
Version 8.6	15.1(4)M	Support for Cisco Unified 8941 and 8945 SCCP IP Phones were introduced. For more information, see Configuring Cisco Unified 8941 and 8945 SCCP IP Phones.

Cisco Unified SRST	Cisco IOS Release	Enhancements or Modifications
Version 8.0	15.1(1)T	Beginning with Cisco IP Phone firmware 8.5(3) and Cisco IOS Release 15.1(1)T, Cisco SRST supports SIP signaling over UDP, TCP, and TLS connections, providing both RTP and SRTP media connections based on the security settings of the IP phone. For more information, see the following sections: • SRST Routers and the TLS Protocol • Media Security on Unified SRST - SRTP • Configuring Secure SIP Call Signaling and SRTP Media with Cisco SRST
Version 7.0/4.3	See Cisco Feature Navigator for compatibility.	• Configuring Eight Calls per Button (Octo-Line) • Configuring Consultative Transfer
Version 4.2(1)	See Cisco Feature Navigator for compatibility.	Enhanced 911 Services. The following new features are included: • Assigning ERLs to zones to enable routing to the PSAP that is closest to the caller. • Customizing E911 by defining a default ELIN, identifying a designated number if the 911 caller cannot be reached on callback, specifying the expiry time for data in the Last Caller table, and enabling syslog messages that announce all emergency calls. • Expanding the E911 location information to include name and address. • Adding new permanent call detail records.
Version 4.1	12.4(15)T	• Enabling KPML for SIP Phones • Disabling SIP Supplementary Services for Call Forward and Call Transfer • Configuring Idle Prompt Status for SIP Phones • Enhanced 911 Services
Version 4.0	12.4(4)XC	• Cisco IP Communicator Support • Fax Pass-through using SCCP and ATAs Support • H.323 VoIP Call Preservation Enhancements for WAN Link Failures for SCCP Phones • Video Support

Cisco Unified SRST	Cisco IOS Release	Enhancements or Modifications
Version 3.4	12.4(4)T	• Cisco SIP SRST 3.4 • Appendix A: Configuring Cisco Unified SIP SRST Features Using Redirect Mode • Configuring Call Handling (see Back-to-Back User Agent Mode)
Version 3.3		• Secure SRST • Cisco Unified IP Phone 7970G and Cisco Unified 7971G-GE Support • Enhancement to the show ephone Command
Version 3.2	12.3(11)T	• Enhancement to the alias Command • Enhancement to the pickup Command • Enhancement to the user-locale Command • Increased the Number of Cisco Unified IP Phones Supported on the Cisco 3845 • MOH Live-Feed Support • No Timeout for Call Preservation • RFC 2833 DTMF Relay Support • Translation Profile Support
Version 3.1	12.3(7)T	• Cisco Unified IP Phone 7920 Support • Cisco Unified IP Phone 7936 Support

Cisco Unified SRST	Cisco IOS Release	Enhancements or Modifications
Version 3.0	12.2(15)ZJ 12.3(4)T	• Additional Language Options for IP Phone Display • Consultative Call Transfer and Forward Using H.450.2 and H.450.3 for SCCP Phones • Customized System Message for Cisco Unified IP Phones • Dual-Line Mode • E1 R2 Signaling Support • European Date Formats • Huntstop for Dual-Line Mode • Music On Hold for Multicast from Flash Files • Ringing Timeout Default • Secondary Dial Tone • Enhancement to the Show ephone Command • System Log Messages for Phone Registrations • Three-Party G.711 Ad Hoc Conferencing • Support for Cisco VG248 Analog Phone Gateway 1.2(1) and Higher Versions
Version 2.1		• Cisco Unified IP Phone 7902G Support • Cisco Unified IP Phone 7912G Support • Additional Language Options for IP Phone Display • Cisco Unified SRST Aggregation • Cisco ATA 186 and ATA 188 Support • Cisco Unified IP Phone 7905G Support • Cisco Unified IP Phone Expansion Module 7914 Support • Enhancement to the Dial Plan-Pattern Command

Cisco Unified SRST	Cisco IOS Release	Enhancements or Modifications
Version 2.02		• Cisco Unified IP Phone Conference Station 7935 Support • Increase in Directory Numbers • Cisco Unity Voicemail Integration Using In-Band DTMF Signaling Across the PSTN and BRI/PRI, on page 40 • Cisco Unified SRST was implemented on the Cisco Catalyst 4500 access gateway module and Cisco 7200 routers (NPE-225, NPE-300, and NPE400). • Support was removed for the Cisco MC3810-V3 concentrator.
Version 2.01		• Cisco Unified SRST was implemented on the Cisco 1760 routers, and support for the Cisco 1750 was removed. • Support was added for additional connected Cisco IP phones. • Support was added for additional directory numbers or virtual voice ports on Cisco IP phones.
Version 2.0		• Cisco Unified SRST was implemented on the Cisco 2600XM and Cisco 2691 routers. • Cisco Unified SRST was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745 routers and the Cisco MC3810-V3 concentrators. • Cisco Unified SRST was implemented on the Cisco 1750 and Cisco 1751 routers. • Huntstop support. • Class of restriction (COR). • Translation rule support. • MOH and tone on hold. • Distinctive ringing. • Forward to a central voicemail or auto-attendant (AA) through PSTN during Cisco Unified Communications Manager fallback. • Phone number alias support during Cisco Unified Communications Manager fallback: enhanced default destination support. • List-based call restrictions for Cisco Unified Communications Manager fallback.

Cisco Unified SRST	Cisco IOS Release	Enhancements or Modifications
Version 1.0		• Support was added for 144 Cisco IP phones on the Cisco 3660 multiservice routers. • Cisco Unified SRST introduced on the Cisco 2600 series and Cisco 3600 series multiservice routers and the Cisco IAD2420 series integrated access devices. • Cisco IP phones able to establish a connection with an SRST router in the event of a WAN link to Cisco Unified Communications Manager failure. • Dimming of all Cisco Unified IP Phone function keys that are not supported during Cisco Unified SRST operation. • Extension-to-extension dialing. • Direct Inward Dialing (DID). • Direct Outward Dialing (DOD). • Calling party ID (Caller ID/ANI) display. • Last number redial. • Preservation of local extension-to-extension calls when WAN link fails. • Preservation of local extension to PSTN calls when WAN link fails. • Preservation of calls in progress when failed WAN link is re-established. • Blind transfer of calls within IP network. • Multiple lines per Cisco IP phone. • Multiple-line appearance across telephones. • Call hold (shared lines). • Analog Foreign Exchange Station (FXS) and Foreign Exchange Office (FXO) ports. • BRI support for EuroISDN. • PRI support for NET5 switch type.

Information About New Features in Cisco Unified SRST

New Features for Cisco Unified SRST Version 14.4

Cisco Unified SRST 14.4 Release introduces support for the following new features:

- Secure SIP SRST and Secure SCCP SRST supports TLS version 1.3 ciphers.
- SHA2 ciphers support with TLS version 1.3 for secure SCCP SRST.

Note Only SCCP Analog Voice Gateways support TLS version 1.3. The SCCP IP phone endpoints do not support TLS version 1.3.

For configuration information, see Configure Secure SRST for SCCP and SIP.

New Features for Cisco Unified SRST Version 14.3

Cisco Unified SRST 14.3 Release introduces support for the following new features:

- Webex Survivability Gateway—From Cisco IOS XE 17.9.3 and Cisco IOS XE Dublin 17.11.1a onward, configure a Webex Survivability Gateway to provide an on-site calling fallback service for Webex Calling endpoints. This feature also supports the colocation of a Webex Calling survivability configuration and a Unified SRST configuration on the same router.

  For general information, see Webex Survivability Gateway Mode, on page 45.

  To configure a Survivability Gateway, see Site Survivability for Webex Calling.

- SFTP CDR Transfer for File Accounting — Allows transfer of SRST CDRs using SFTP. See Configuring File Accounting.

New Features for Cisco Unified SRST Version 14.2

Cisco Unified SRST 14.2 Release introduces support for the following new features:

- Restrict Secure SIP SRST and Secure SCCP SRST to only using TLS 1.2 SHA2 Cipher Suites—SHA2-Cipher-Only Mode for Unified Secure SRST, on page 287
- SIP OAuth Support for Secure SRST—SIP OAuth Client Registration for Unified Secure SRST, on page 279

New Features for Unified SRST Version 14.1

Unified SRST 14.1 Release introduces support for the following new features:

- Voice: Class of Restriction YANG Configuration Model—Programmability Guide for Cisco IOS XE Unified Communications VoIP Products

- Smart Licensing Using Policy—Cisco Smart Licensing for Unified SRST
- Smart Licensing Using Policy—Cisco Smart Licensing for Unified E-SRST

New Features for Unified SRST Version 12.7

Unified SRST 12.7 Release introduces support for the following new feature:

- Support for maximum number of devices in Cisco 4451 and 4461 Integrated Services Routers was increased from 1500 to 2000.

New Features for Cisco Unified SRST Version 12.6

Cisco Unified SRST 12.6 Release introduces support for the following new features:

- Simple Network Management Protocol (SNMP) Support for Unified SRST
- Toll Fraud Prevention for SIP Line Side on Unified SRST
- Unified SRST, Unified E-SRST, and Unified Secure SRST Password Policy

New Features for Cisco Unified SRST Version 12.3

Cisco Unified SRST 12.3 Release introduces support for Secure SCCP SRST Support.

New Features for Cisco Unified SRST Version 12.2

Cisco Unified SRST 12.2 Release introduces support for Unified E-SRST with Support for Voice Hunt Group.

New Features for Cisco Unified SRST Version 12.1

Cisco Unified SRST 12.1 introduces support for the following new features:

- Licensing
- Secure SCCP SRST Support
- Unified SRST and Unified Border Element Co-location

New Feature for Cisco Unified SRST Version 12.0

Cisco Unified SRST 12.0 introduces support for IPv6 protocols on SIP IP Phones. For more information on IPv6 Support introduced for Cisco Unified SRST, see IPv6 Support for Unified SRST SIP IP Phones.

New Features for Cisco Unified SRST Version 11.0

Cisco Unified SRST 11.0 supports the following new Cisco IP phones and adapters:

- Support for Cisco IP Phone 7811

- Support for Cisco IP Phones 8811, 8831, 841, 8851, 8851NR, 8861
- Support for Cisco ATA-190

For information on the phones supported in Cisco Unified SRST 11.0, see Phone Feature Support Guide for Cisco Unified Communications Manager Express, Cisco Unified SRST, Unified E-SRST, and Unified Secure SRST.

New Features for Cisco Unified SRST Version 10.5

Cisco Unified SRST 10.5 supports the following features:

- Where to Go Next, Setting Up the Network, on page 153

For more information on the Cisco Unified SRST 10.5 supported feature, see the SCCP: Configure Unified E-SRST.

Cisco Unified SRST 10.5 supports the following new Cisco Unified SIP IP phones:

- Support for Cisco Unified DX650 SIP IP Phones
- Support for Cisco Unified 78xx SIP IP Phones

Support for Cisco Unified DX650 SIP IP Phones

For information on feature support for the Cisco Unified DX650 SIP IP Phones in Cisco Unified SRST 10.5, see Phone Feature Support Guide for Unified CME, Unified SRST, Unified E-SRST, and Unified Secure SRST.

Support for Cisco Unified 78xx SIP IP Phones

For information on feature support for the Cisco Unified 78xx SIP IP Phones in Cisco Unified SRST 10.5, see Phone Feature Support Guide for Unified CME, Unified SRST, Unified E-SRST, and Unified Secure SRST.

New Features in Cisco Unified SRST Version 10.0

Cisco Unified SRST 10.0 supports the following new features:

- Cisco Jabber for Windows
- SIP: Configure Unified E-SRST

To obtain an account on Cisco.com, go to www.cisco.com and click Register at the top of the screen.

Cisco Jabber for Windows

Cisco Jabber for Windows client is supported from Cisco Unified CME Release 10 onwards.Cisco Jabber for Windows supports the visual voicemail functionality integrated with the Cisco Unity connection. Cisco Jabber for Windows is a SIP-based soft client with integrated Instant Messaging and presence functionality, and uses the new Client Services Framework 2nd Generation (CSF2G) architecture.

CSF is a unified communications engine that is reused by multiple Cisco PC-based clients. The Cisco Jabber client has to be registered with a presence server such as cloud-based Cisco Webex server, or Cisco Unified

Presence server to avail the standard XMPP-based instant messaging functionalities. The client is identified by a device ID name that can be configured under the voice register pool in Cisco Unified CME. You should configure the username and password under voice register pool to identify the user logging into Cisco Unified CME through Cisco Jabber for Windows client. The device discovery process uses HTTPS connection. Therefore, you should configure the secure HTTP on Cisco Unified CME. A new phone type, Jabber-Win has been added to configure the voice register pool for Cisco Jabber for Windows client.

Restrictions

- The Cisco Jabber for Windows client version should be version 9.1.0 and later version.
- The Cisco Jabber for Windows client should register with a presence server such as cloud-based Webex server, or a Cisco Unified Presence server to enable the telephony features on the Jabber client.
- The Cisco Jabber for Windows client supports only the visual voicemail functionality using Internet Message Access Protocol (IMAP) on the Cisco Unity Connection.
- The Cisco Jabber for Windows client does not support software-based conferencing and supports only the softphone mode with Cisco Unified CME.
- Desk phone models are not supported.

For configuration information, see the "Cisco Jabber for Windows" section of Cisco Unified Communications Manager Administration Guide.

Version Negotiation for Cisco Unified SIP IP Phones

The version negotiation for Cisco Unified SIP IP Phones was introduced in Cisco Unified SRST 10.0 release. For more information on the Cisco Unified SRST 10.0 supported features, see the SIP: Configure Unified E-SRST section.

New Features in Cisco Unified SRST Version 9.5

After-hour Pattern Blocking Support for Regular Expressions

In Cisco Unified SRST 9.5, support for afterhours pattern blocking is extended to regular expression patterns for dial plans on Cisco Unified SIP and Cisco Unified SCCP IP phones. With this support, users can add a combination of fixed dial plans and regular expression-based dial plans.

When a call is initiated after hours, the dialed number is matched against a combination of dial plans. If a match is found, the call is blocked.

To enable regular expression patterns to be included when configuring afterhours pattern blocking, the **after-hours block pattern** command is modified to include regular expressions as a value for the *pattern* argument in the following command syntax:

after-hours block pattern *pattern-tag pattern*

This command is available in the following configuration modes:

- telephony-service—For both SCCP and SIP Phones.
- ephone-template—For SCCP phones only.

Note The maximum length of a regular expression pattern is 32 for both Cisco Unified SIP and Cisco Unified SCCP IP phones.

If calls to the following numbers are to be blocked after hours:

- numbers beginning with ‘0’ and ‘00’
- numbers beginning with 1800, followed by four digits
- numbers 9876512340 to 9876512345

then the following configurations can be used:

- after-hours block pattern 1 0*
- after-hours block pattern 2 00*
- after-hours block pattern 3 1800….
- after-hours block pattern 4 987651234[0-5]

Note There is no change in the number of afterhours patterns that can be added. The maximum number is still 100.

For more information on configuration examples, see the “Configuring Afterhours Block Patterns of Regular Expressions: Example” section of Cisco Unified Communications Manager Administration Guide.

For a summary of the basic Cisco IOS regular expression characters and their functions, see the Cisco Regular Expression Pattern Matching Characters section of *Terminal Services Configuration Guide*.

Call Park Recall Enhancement

Before Cisco Unified CME 9.5, a parked call could not be recalled by or transferred to the phone that put the call in park or the original phone that transferred the call when the destination phone was offhook or ringing.

In Cisco Unified CME 9.5, the **recall force** keyword is added to the **call-park system** command in telephony-service configuration mode to allow a user to force the recall or transfer of a parked call to the phone that put the call in park or the phone with the reserved-for number as its primary DN when the destination phone is available to answer the call.

In Cisco Unified CME 10.5, a new ring tone is introduced for park recall to assist the phone user to distinctly identify the type of call.

This feature is supported on all phone families for SCCP endpoints and on 89XX and 99XX phone families for SIP endpoints. No configurations are required to activate this feature.

The following example configures the Call Park Recall:

```
Router# configure terminal
Router(config)# telephony-service
Router(config)# srst mode auto-provision all
Router(config-telephony)# call-park system ? recall Configure parameters for recall
Router(config-telephony)# call-park system recall ? force Force recall for busy call park
```

```
initiator
Router(config-telephony)# call-park system recall force
```

Park Monitor

In Cisco Unified CME 8.5 and later versions, the park monitor feature allows you to park a call and monitor the status of the parked call until the parked call is retrieved or abandoned. When a Cisco Unified SIP IP Phone 8961, 9951, or 9971 parks a call using the park soft key, the park monitoring feature monitors the status of the parked call. The park monitoring call bubble is not cleared until the parked call gets retrieved or is abandoned by the parkee. This parked call can be retrieved using the same call bubble on the parker's phone to monitor the status of the parked call.

Once a call is parked, Cisco Unified CME sends a SIP NOTIFY message to the parker phone indicating the "parked" event along with the park slot number so that the parker phone can display the park slot number as long as the call remains parked.

When a parked call is retrieved, Cisco Unified CME sends another SIP NOTIFY message to the parker phone indicating the "retrieved" event so that the phone can clear the call bubble. When a parked call is disconnected by the parkee, Cisco Unified CME sends a SIP NOTIFY message to the parker phone indicating the "abandoned" event and the parker phone clears the call bubble upon cancellation of the parked call.

When a parked call is recalled or transferred, Cisco Unified CME sends a SIP NOTIFY message to the parker phone indicating the "forwarded" event so that parker phone can clear the call bubble during park, recall, and transfer. You can also retrieve a parked call from the parker phone by directly selecting the call bubble or pressing the resume soft key on the phone.

Display Support for Name of Called Voice Hunt Groups

A voice hunt group is associated with a pilot number. But because there is no association with the name of the voice hunt group when calls are forwarded from the voice hunt group to the final number, the forwarding number is sent without the name of the forwarding party. The final number can be in the form of a voicemail, a Basic Automatic Call Distribution (BACD) script, or another extension.

In Cisco Unified SRST 9.5, the display of the name of the called voice-hunt-group pilot is supported by configuring the following command in **voice hunt-group** or **ephone-hunt** configuration mode:

[**no**] **name** *"primary pilot name"* [**secondary** *"secondary pilot name"*]

The secondary name is optional and when the secondary pilot name is not explicitly configured, the primary pilot name is applicable to both pilot numbers.

For configuration information, see the "Associating a Name with a Called Voice Hunt Group" section of Cisco Unified Communications Manager Administration Guide.

For configuration examples, see the "Example: Associating a Name with a Called Voice Hunt Group" section of Cisco Unified Communications Manager Administration Guide.

Restrictions

- Display support applies to Cisco Unified SCCP IP phones in voice hunt-group and ephone-hunt configuration modes but are not supported in Cisco Unified SIP IP phones.
- Called name and called number information displayed on the caller's phone follows existing behavior, where the called names and called numbers are updated so that a sequential hunt reflects the name and number of the ringing phone.

The following example configures the primary pilot name for both the primary and secondary pilot numbers:

```
name SALES
```

The following example configures different names for the primary and secondary pilot numbers:

```
name SALES secondary SALES-SECONDARY
```

Note Use quotes (") when input strings have spaces in between as shown in the next three examples.

The following example associates a two-word name for the primary pilot number and a one-word name for the secondary pilot number:

```
name "CUSTOMER SERVICE" secondary CS
```

The following example associates a one-word name for the primary pilot number and a two-word name for the secondary pilot number:

```
name FINANCE secondary "INTERNAL ACCOUNTING"
```

The following example associates two-word names for the primary and secondary pilot numbers:

```
name "INTERNAL LLER" secondary "EXTERNAL LLER"
```

Preventing Local-Call Forwarding to Final Agent in Voice Hunt Groups

Local or internal calls are calls originating from a Cisco Unified SIP or Cisco Unified SCCP IP phone in the same Cisco Unified CME system.

Before Cisco Unified CME 9.5, the **no forward local-calls** command was configured in ephone-hunt group to prevent a local call from being forwarded to the next agent.

In Cisco Unified CME 9.5, local calls are prevented from being forwarded to the final destination using the **no forward local-calls to-final** command in parallel or sequential voice hunt-group configuration mode.

When the **no forward local-calls to-final** command is configured in sequential voice hunt-group configuration mode, local calls to the hunt-group pilot number are sent sequentially only to the list of members of the group using the rotary-hunt technique. In case all the group members of the voice hunt group are busy, the caller hears a busy tone. If any of the group members are available but do not answer, the caller hears a ringback tone and is eventually disconnected after the specified timeout. The call is not forwarded to the final number.

When the **no forward local-calls to-final** command is configured in parallel voice hunt-group configuration mode, local calls to the hunt-group pilot number are sent simultaneously to the list of members of the group using the blast technique. In case all the group members of the voice hunt group are busy, the caller hears a busy tone. If any of the group members are available but do not answer, the caller hears a ringback tone and is eventually disconnected after the specified timeout.The call is not forwarded to the final number. or configuration examples, see the “Preventing Local-Call Forwarding to Final Agent in Voice Hunt Groups” section of” section of Cisco Unified Communications Manager Administration Guide.

Trunk-to-Trunk Transfer Blocking for Toll Fraud Prevention on Cisco Unified SIP IP Phones

In Cisco Unified Survivable Remote Site Telephony (SRST) 4.0, trunk-to-trunk transfer blocking for toll bypass fraud prevention is supported on Cisco Unified Skinny Client Control Protocol (SCCP) IP phones.

The following table lists the transfer-blocking commands and the appropriate configuration modes for Cisco Unified CME and Cisco Unified SRST.

Commands	Cisco Unified SRST
transfer-pattern	call-manager-fallback
transfer max-length	voice register pool
transfer-pattern blocked	voice register pool
conference transfer-pattern	call-manager-fallback
conference max-length	voice register pool or voice register template
conference-pattern blocked	voice register pool or voice register template

Note The call transfer and conference restrictions apply when transfers or conferences are initiated toward external parties, like a PSTN trunk, a SIP trunk, or an H.323 trunk. The restrictions do not apply to transfers and conferences to local extensions.

Transfer-Pattern

The **transfer-pattern** command for Cisco Unified SIP IP phones functions like the **transfer-pattern** command for Cisco Unified SCCP IP phones by allowing all, not just local, transfers to take place.

The **transfer-pattern** command specifies the directory numbers for Call Transfer. The command can be configured up to 32 times using the following command syntax: **transfer-pattern** *transfer-pattern* [**blind**].

Note The **blind** keyword in the **transfer-pattern** command applies only to Cisco Unified SCCP IP phones and does not apply to Cisco Unified SIP IP phones.

With the **transfer-pattern** command configured, only Call Transfers to numbers that match the configured transfer pattern are allowed to take place. With the transfer pattern configured, all or a subset of transfer numbers can be dialed and the transfer to a remote party can be initiated.

The following are examples of configurable transfer patterns:

- .T—This configuration allows Call Transfers to any destinations with one or more digits, like 123, 877656, or 76548765.
- 919........—This configuration only allows Call Transfers to remote numbers beginning with “919” and followed by eight digits, like 91912345678. However, Call Transfers to 9191234 or 919123456789 are not allowed.

Backward Compatibility

To maintain backward compatibility, all Call Transfers from Cisco Unified SIP IP phones to any number (local or over the trunk) are allowed when no transfer patterns are configured through the **transfer-pattern** , **transfer-pattern blocked**, or **transfer max-length** commands.

For Cisco Unified SCCP IP phones, if you do not configure transfer patterns, Call Transfers over the trunk are blocked.

Dial Plans

Whatever dial plan is used for external calls, the same numbers should be configured as specific numbers using the**transfer-pattern** command.

If a dial plan requires “9” to be dialed before making an external call, then prefix “9” to the transfer-pattern number. For example, if 12345678 is an external number that requires “9” to be dialed before making the external call, then the transfer-pattern number is 912345678.

Transfer Max-Length

The **transfer max-length** command is used to indicate the maximum length of the number being dialed for Call Transfer. When only a specific number of digits are allowed during a Call Transfer, value from 3 through 16 is configured. When the number dialed exceeds the maximum length, then the Call Transfer is blocked.

For example, if you configure 5 as the maximum length, Call Transfers from Cisco Unified SIP IP phones allows up to a five-digit directory number. All Call Transfers to directory numbers with more than five digits are blocked.

Note If only **transfer max length** is configured and **conference max-length** is not configured, then **transfer max length** takes effect for transfers and conferences.

Transfer-Pattern Blocked

When the **transfer-pattern blocked** command is configured for a specific phone, no Call Transfers are allowed from that phone over the trunk.

This feature forces unconditional blocking of all Call Transfers from the specific phone to any other nonlocal numbers (external calls from one trunk to another trunk). No Call Transfers from this specific phone are possible even when a transfer pattern matches the dialed digits for transfer.

The following table compares the behaviors of Cisco Unified SCCP and SIP IP phones for specific configurations.

Configuration	Cisco Unified SCCP IP Phones	Cisco Unified SIP IP Phones
No transfer patterns are configured.	Blocks all nonlocal Call Transfers.	Allows all nonlocal Call Transfers for backward compatibility.
Specific transfer patterns are configured.	Allows Call Transfers to specific external entities.	Allows Call Transfers to specific external entities.
The **transfer-pattern blocked** command is configured.	Blocks all nonlocal Call Transfers are blocked. **Note** The configuration reverts to the default, where no transfer patterns are configured.	All nonlocal Call Transfers are blocked. **Note** The configuration unconditionally blocks all nonlocal Call Transfers. It does not return to the default, where all nonlocal Call Transfers are allowed.

Conference-Pattern Blocked

The **conference-pattern blocked** command is used to prevent extensions on a voice register Pool from initiating conferences.

The following table summarizes the behavior of the **conference-pattern blocked** command in relation to **no conference-pattern blocked** ,**conference max-length** , **no conference max-length** , and **transfer max-length** commands.

	Conference max-length	**No conference max-length**
No conference-pattern blocked (default case)	Allowing/Blocking of conference call depends on configured conference max-length.	Allowing/Blocking of conference call depends on configured transfer max-length.
Conference-pattern blocked	Conference calls are not allowed on SIP and SCCP phones.	

	Max-length <= allowed max-length		**Max-length > allowed max-length**	
	Transfer	Conference	Transfer	Conference
Transfer max-length + No Conference max-length (use transfer max-length for conference cases too, as conference max-length not configured)	Y	Y	N	N
No transfer max-length + Conference max-length (conference max-length has precedence over transfer max-length for conference)	Y	Y	Y	N
No transfer max-length + Conference max-length (conference max-length has precedence over transfer max-length for conference)	Y	Y	N	N
No transfer max-length + No conference max-length	All transfer and conference calls are allowed.			

Configuring the Maximum Number of Digits for a Conference Call

Before you begin

Cisco Unified SRST 10.5 or a later version.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register pool** *pool-tag* OR**ephone***phone-tag*
4. **conference max-length** *value*
5. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router# enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice register pool** *pool-tag* OR**ephone***phone-tag* **Example:** `Router(config)# voice register pool 25`	Enters voice register Pool configuration mode and creates a Pool configuration for a Cisco Unified SIP IP phone in Cisco Unified Communications Manager Express or for a set of Cisco Unified SIP IP phones in Cisco Unified SIP SRST. • *pool-tag* : Unique number assigned to the Pool. Range is 1–100. OR Enters voice register template configuration mode and defines a template of common parameters for Cisco Unified SIP IP phones. • *template-tag* : Declares a template tag. Range is 1–10. OR Enters ephone configuration mode. • *phone-tag* : Unique sequence number that identifies this ephone during configuration tasks. The maximum number of ephones is version and platform-specific. Type? To display range.
Step 4	**conference max-length** *value* **Example:** `Router(config-telephony)# conference max-lenght 6`	Allows the conference of calls from Cisco IP phones to specified directory numbers of phones other than Cisco IP phones. *conference max-length* Allows conference call depending on the configured conference max-length. Range is 3–16.

	Command or Action	Purpose

Step 5	**end**
	Example:
	`Router(config-telephony)# end`
		Exits telephony-service configuration mode and enter privileged EXEC mode.

Configuring Conference Blocking Options for Phones

Before you begin

- Use Cisco Unified SRST 10.5 or a later version.
- Configure the transfer-pattern command.
- Configure the conference transfer-pattern command.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register pool** *pool-tag* OR**ephone***phone-tag*
4. **conference-pattern blocked**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router# enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice register pool** *pool-tag* OR**ephone***phone-tag* **Example:** `Router(config)# voice register pool 25`	Enters voice register Pool configuration mode and creates a Pool configuration for a Cisco Unified SIP IP phone in Cisco Unified Communications Manager Express or for a set of Cisco Unified SIP IP phones in Cisco Unified SIP SRST. • *pool-tag* : Unique number assigned to the Pool. Range is 1–100. OR Enters voice register template configuration mode and defines a template of common parameters for Cisco Unified SIP IP phones.

	Command or Action	Purpose
		- *template-tag* : Declares a template tag. Range is 1–10. OR Enters ephone configuration mode. - *phone-tag* : Unique sequence number that identifies this ephone during configuration tasks. The maximum number of ephones is version and platform-specific. Type? To display range.
Step 4	**conference-pattern blocked** **Example:** `Router(config-telephony)# conference-pattern blocked`	Allows the conference of calls from Cisco IP phones to specified directory numbers of phones other than Cisco IP phones. *conference-pattern blocked* No conference calls are allowed.
Step 5	**exit** **Example:** `Router(config-telephony)# exit`	Exits telephony-service configuration mode and enter global configuration mode.

Transfer-Pattern Blocked

When the **transfer-pattern blocked** command is configured for a specific phone, no Call Transfers are allowed from that phone over the trunk.

This feature forces unconditional blocking of all Call Transfers from the specific phone to any other nonlocal numbers (external calls from one trunk to another trunk). No Call Transfers from this specific phone are possible even when a transfer pattern matches the dialed digits for transfer.

The following table compares the behaviors of Cisco Unified SCCP and SIP IP phones for specific configurations.

Configuration	Cisco Unified SCCP IP Phones	Cisco Unified SIP IP Phones
No transfer patterns are configured.	Blocks all nonlocal Call Transfers.	Allows all nonlocal Call Transfers for backward compatibility.
Specific transfer patterns are configured.	Allows Call Transfers to specific external entities.	Allows Call Transfers to specific external entities.
The **transfer-pattern blocked** command is configured.	Blocks all nonlocal Call Transfers are blocked. **Note** The configuration reverts to the default, where no transfer patterns are configured.	All nonlocal Call Transfers are blocked. **Note** The configuration unconditionally blocks all nonlocal Call Transfers. It does not return to the default, where all nonlocal Call Transfers are allowed.

Conference Transfer-Pattern

When both the **transfer-pattern** and **conference transfer-pattern** commands are configured and dialed digits match the configured transfer pattern, conference calls are allowed. However, when the dialed digits do not match the configured transfer pattern, the conference call is blocked.

For information on provisioning Cisco Unified IP phones for secure access to web content using HTTPS, see the HTTPS Provisioning for Cisco Unified IP Phones section of Cisco Unified Communications Manager Express System Administrator Guide.

For configuration examples, see the Configuring HTTPS Support for Cisco Unified Communications Manager Express: Example section of Cisco Unified Communications Manager Administration Guide.

New Features in Cisco Unified SRST Version 9.1

Cisco Unified SRST 9.1 supports the following new features:

- Key Expansion Module Support for Cisco Unified SIP IP Phones
- Enhancement in Speed-Dial Support
- Voice Hunt Group Support

Note If you have older routers, such as the VG26nn and VG37nn platforms and Cisco Integrated Services Router (ISR) Generation 1 platforms (Cisco ISR 1861, 2800, and 3800 Series), you must upgrade to Cisco ISR 881, 886VA, 887VA, 888, 888E, 1861E, 2900, 3900, and 3900E Series platforms to utilize these new features.

Key Expansion Module Support for Cisco Unified SIP IP Phones

Cisco Unified IP Key Expansion Modules (KEMs) are supported on Cisco Unified 8851/51NR, 8861, 8961, 9951, and 9971 SIP IP phones from Cisco Unified SIP SRST 9.1.

For information on KEMs support for Cisco Unified 8851/51NR, 8861, 8961, 9951, and 971 SIP IP phones, see Phone Feature Support Guide for Cisco Unified Communications Manager Express, Cisco Unified SRST, Unified E-SRST, and Unified Secure SRST.

Restrictions

- Bulk registration is not supported for KEMs in Cisco Unified SRST. Phones do not send bulk Registration Requests but always use the UDP port for registration.
- KEMs is not supported for Cisco Unified SCCP IP Phones and Cisco Unified SIP IP Phones other than the Cisco Unified 8851/51NR, 8861, 8961, 9951, and 9971 SIP IP phones.
- Features configured on keys are disabled when supported Cisco Unified SIP IP phones are in Cisco Unified SIP SRST.
- All Cisco Unified 8851/51NR, 8861,8961, 9951, and 9971 SIP IP phone restrictions and limitations apply to KEMs.
- All Cisco Unified SIP SRST feature restrictions and limitations apply to KEMs.

For more information on how the **blf-speed-dial** , **number** , and **speed-dial** commands, in voice register Pool configuration mode, have been modified, see Cisco Unified Communications Manager Express Command Reference.

For information on installing KEMs on Cisco Unified IP Phone, see the Installing a Key Expansion Module on the Cisco Unified IP Phone section of Cisco Unified IP Phone 8961, 9951, and 9971 Administration Guide for Cisco Unified Communications Manager 7.1 (3) (SIP).

For information on installing KEMs on Cisco Unified 8811, 8841, 8851, 8851NR, and 8861 Phones, see the Cisco IP Phone Key Expansion Module section of Cisco IP Phone 8811, 8841, 8851, 8851NR, and 8861 Administration Guide for Cisco Unified Communications Manager.

Enhancement in Speed-Dial Support

Cisco Unified SRST 9.1 ignores the "," or comma (pause indicator) to avoid break-in speed-dial support.

Because the pause speed-dial feature (supported in Cisco Unified Communications Manager or Cisco Unified Communications Manager) is not supported in Cisco Unified SRST, Cisco Unified Communications Manager and phones (Cisco Unified SCCP IP phones and Cisco Unified SIP IP phones) registered in Cisco Unified SRST maintain backward compatibility in Cisco Unified SRST mode. When phones failover to the Cisco Unified SRST router during WAN outages and Cisco Unified Communications Manager fails, the phones only send the speed-dial numbers when the pause speed-dial buttons are pressed. The comma pause indicator is ignored and the preconfigured FAC, PIN, and DTMF are not sent.

For information on configuring speed-dial in Cisco Unified Communications Manager, see the "Device setup" chapter of Cisco Unified Communications Manager Administration Guide.

Voice Hunt Group Support

Cisco Unified SIP SRST 9.1 supports voice hunt groups. Voice hunt groups allow call placed to a single (pilot) number to contact multiple destinations.

There are three different types of voice hunt groups. Each type uses a different strategy to determine the first number that rings for successive calls to the pilot number until a number answers.

- Parallel Hunt Groups—Allows an incoming call to simultaneously ring all the numbers in the hunt group member list.
- Sequential Hunt Groups—Allows an incoming call to ring all the numbers in the left-to-right order in which they were listed while defining the hunt group. The first number in the list is always the first number tried when the pilot number is called. Maximum number of hops is not a configurable parameter for sequential hunt groups.
- Longest-idle Hunt Groups—Allows an incoming call to first go to the number that has been idle the longest for the number of hops specified when the hunt group was defined. The longest-idle time is determined from the last time that a phone registered, reregistered, or went on-hook.

Cisco Unified SCCP IP phones support only ephone hunt groups whereas a voice hunt group supports Cisco Unified SCCP IP phones, Cisco Unified SIP IP phones. In addition, it also supports a mixture of Cisco Unified SCCP IP phones and Cisco Unified SIP IP phones.

With the voice hunt group feature preconfigured in the Cisco Unified SIP SRST router, voice hunt groups continue to be supported after phones fallback from Cisco Unified Communications Manager to the Cisco Unified SIP SRST router.

Restrictions

- Hunt group statistics is not supported for voice hunt groups in Cisco Unified SRST.
- Hunt group nesting or setting the final number of one hunt groups as the pilot of another hunt group is not supported.

New Features in Cisco Unified SRST Version 9.0

Support for Cisco Unified 6901 and 6911 SIP IP Phones

For information on feature support for the Cisco Unified 6901 and 6911SIP IP Phones in Cisco Unified SRST, see Phone Feature Support Guide for Cisco Unified Communications Manager Express, Cisco Unified SRST, Unified E-SRST, and Unified Secure SRST.

Support for Cisco Unified 6921, 6941, 6945, and 6961 SIP IP Phones

For information on feature support for the Cisco Unified 6921, 6941, 6945, and 6961 SIP IP Phones in Cisco Unified SRST, see Phone Feature Support Guide for Cisco Unified Communications Manager Express, Cisco Unified SRST, Unified E-SRST, and Unified Secure SRST.

Support for Cisco Unified 8941 and 8945 SIP IP Phones

For information on feature support for the Cisco Unified 8941 and 8945 SIP IP Phones in Cisco Unified SRST, see Phone Feature Support Guide for Cisco Unified Communications Manager Express, Cisco Unified SRST, Unified E-SRST, and Unified Secure SRST.

Multiple Calls Per Line

Cisco Unified SRST 9.0 supports the Multiple Calls Per Line (MCPL) feature on Cisco Unified 6921, 6941, 6945, and 6961 SIP IP phones. In addition, it supports Cisco Unified 8941, 8945 SCCP, and SIP IP phones.

Before Cisco Unified SRST 9.0, supports only two calls for every directory number (DN) on Cisco Unified 8941 and 8945 SCCP IP phones.

With Cisco Unified SRST 9.0, the MCPL feature overcomes the limitation on the maximum number of calls per line.

Cisco Unified SRST 9.0 does not support the MCPL feature on Cisco Unified 6921, 6941, 6945, and 6961 SCCP IP phones. Allows only two calls on these phones whereas allows only one call on octo-line directory numbers on these phones before activating Call Forward Busy or busy tone.

Cisco Unified 8941 and 8945 SCCP IP Phones

Before Cisco Unified SRST 9.0, the values for the **max-dn** and **timeouts busy** commands were hardcoded for Cisco Unified 8941 and 8945 SCCP IP phones.

In Cisco Unified SRST 9.0, you can configure the **max-dn** and**timeouts busy** commands in call-manager-fallback configuration mode. Use the **max-dn** command to set the maximum number of DNs that can be supported by the router and enable dual-line mode, octo-line mode, or both modes. Use the **timeouts busy** command to set the timeout value for Call Transfers to busy destinations.

For configuration information, see Configuring the Maximum Number of Calls.

Cisco Unified 6921, 6941, 6945, 6961, 8941, and 8945 SIP IP Phones

In Cisco Unified SRST 9.0, the maximum number of calls for Cisco Unified 6921, 6941, 6945, 6961, 8941, and 8945 SIP IP phones is controlled by the phones.

Prerequisites

- Cisco Unified SRST 9.0 and later versions.
- Correct firmware is installed:
    - 9.2(1) or a later version for Cisco Unified 6921, 6941, 6945 and 6961 SIP IP phones.
    - 9.2(2) or a later version for Cisco Unified 8941 and 8945 SIP IP phones.

Voice and Fax Support on Cisco ATA-187

Cisco ATA-187 is a SIP-based analog phone adapter that turns traditional phone devices into IP devices. Cisco ATA-187 can connect with a regular analog FXS phone or fax machine on one end, while the other end is an IP side that uses SIP for signaling and registers as a Cisco Unified SIP IP phone.

Cisco ATA-187 functions as a Cisco Unified SIP IP phone that supports T.38 fax relay and fax pass-through, enabling the real-time transmission of fax over IP networks. The fax rate is from 7.2 to 14.4 kbps.

For information on feature support for the Cisco ATA-187 in Cisco Unified SRST, see Phone Feature Support Guide for Cisco Unified Communications Manager Express, Cisco Unified SRST, Unified E-SRST, and Unified Secure SRST.

For more information on Cisco ATA-187, see Cisco ATA 187 Analog Telephone Adaptor Administration Guide for SIP.

New Features in Cisco Unified SRST Version 8.8

Cisco Unified SRST 8.8 supports the following new Cisco Unified SCCP IP phones:

- Cisco Unified 6945 SCCP IP Phones
- Cisco Unified 8941 SCCP IP Phones
- Cisco Unified 8945 SCCP IP Phones

Support for Cisco Unified 6945, 8941, and 8945 SCCP IP Phones

For information on feature support for the Cisco Unified 6945, 8941, and 8945 SCCP IP Phones in Cisco Unified SRST, see Phone Feature Support Guide for Cisco Unified Communications Manager Express, Cisco Unified SRST, Unified E-SRST, and Unified Secure SRST.

For information on the Cisco Unified 6945 SCCP IP Phone, see Cisco Unified IP Phone 6945 User Guide for Cisco Unified Communications Manager Express Version 8.8 (SCCP).

For information on the Cisco Unified 8941 and 8945 SCCP IP Phones, see Cisco Unified IP Phone 8941 and 8945 User Guide for Cisco Unified Communications Manager Express Version 8.8 (SCCP).

New Features in Cisco Unified SRST Version 8.0

Beginning with Cisco IP Phone firmware 8.5(3) and Cisco IOS Release 15.1(1)T, Cisco Unified SRST supports SIP signaling over UDP, TCP, and TLS connections, providing both RTP and SRTP media connections based on the security settings of the IP phone.

New Features in Cisco Unified SRST Version 7.0/4.3

Cisco Unified SRST 7.0/4.3 supports the following new features:

- Configuring Eight Calls per Button (Octo-Line)
- Configuring Consultative Transfer

New Features in Cisco Unified SRST Version 4.2(1)

Cisco Unified SRST Version 4.2(1) introduces the new feature enhancements for Enhanced 911 Services.

New Features in Cisco Unified SRST Version 4.1

Cisco Unified SRST Version 4.1 introduces the following new feature:

- Enhanced 911 Services

New Features in Cisco Unified SRST Version 4.0

Additional Cisco Unified IP Phone Support

The following IP phones are supported with Cisco Unified SRST systems:

- Cisco Unified IP Phone 7911G
- Cisco Unified IP Phone 7941G and Cisco Unified IP Phone 7941G-GE
- Cisco Unified IP Phone 7960G
- Cisco Unified IP Phone 7961G and Cisco Unified IP Phone 7961G-GE

In addition, the Cisco Unified IP Phone 7914 Expansion Module can attach to the Cisco 7941G-GE and Cisco 7961G-GE. The Cisco 7914 Expansion Module adds additional features, such as adding 14 line appearances or speed-dial numbers to your phone. You can attach one or two expansion modules to your IP phone. When you use two expansion modules, you have 28 additional line appearances or speed-dial numbers, or a total of 34 line appearances or speed-dial numbers. For more information, see Cisco IP Phone 7914 Expansion Module Quick Start Guide.

No additional SRST configuration is required for these phones.

The **show ephone** command is enhanced to display the configuration and status of the new Cisco IP Phones added to SRST Version 4.0. For more information, see the **show ephone** command in Cisco Unified SRST and Cisco Unified SIP SRST Command Reference (All Versions).

To determine compatible firmware, platforms, memory, and additional voice products that are associated with Cisco Unified SRST 4.0, see Cisco Unified SRST 4.3 Supported Firmware, Platforms, Memory, and Voice Products.

Cisco IP Communicator Support

Cisco IP Communicator is a software-based application that delivers enhanced telephony support on personal computers. This SCCP-based application allows computers to function as IP phones, providing high-quality voice calls on the road, in the office, or from wherever users may have access to the corporate network. Cisco IP Communicator appears on a user's computer monitor as a graphical, display-based IP phone with a color screen, a key pad, feature buttons, and soft keys.

Fax Pass-through using SCCP and ATAs Support

Fax pass-through mode is now supported using Cisco VG 224 voice gateways, Analog Telephone Adaptors (ATA), and SCCP. ATAs ship with SIP firmware, so SCCP firmware must be loaded before this feature can be used.

Note For ATAs that are registered to a Cisco Unified SRST system to participate in FAX calls, they must have their ConnectMode parameter set to use the “standard payload type 0/8” as the RTP payload type in FAX pass-through mode. For ATAs used with Cisco Unified SRST 4.0 and higher versions, this is done by setting bit 2 of the ConnectMode parameter to 1 on the ATA. For more information, see the “Parameters and Defaults” chapter in Cisco ATA 186 and Cisco ATA 188 Analog Telephone Adaptor Administrator's Guide for SCCP.

H.323 VoIP Call Preservation Enhancements for WAN Link Failures for SCCP Phones

H.323 VoIP call preservation enhancements for WAN link failures sustains connectivity for H.323 topologies where signaling is handled by an entity, such as Cisco Unified Communications Manager, that is different from the other endpoint and brokers signaling between the two connected parties.

Call preservation is useful when a gateway and the other endpoint (typically a Cisco Unified IP phone) are collocated at the same site and the call agent is remote and therefore more likely to experience connectivity failures. H.323 VoIP call preservation enhancements does not support SIP Phones.

For configuration information see the “Configuring H.323 Gateways” chapter in Cisco IOS H.323 Configuration Guide.

Video Support

This feature allows you to set video parameters for the Cisco Unified SRST to maintain close feature parity with Cisco Unified CM. When the Cisco Unified SRST is enabled, Cisco Unified IP Phones do not have to be reconfigured for video capabilities because all ephones retain the same configuration used with Cisco Unified CM. However, you must enter call-manager-fallback configuration mode to set video parameters for Cisco Unified SRST. The feature set for video is the same as that for Cisco Unified SRST audio calls.

For more information, see Setting Video Parameters.

New Features in Cisco Unified SRST Version 3.4

Cisco SIP SRST 3.4

Cisco SIP SRST Version 3.4 describes SRST functionality for Session Initiation Protocol (SIP) networks. Cisco SIP SRST Version 3.4 provides backup to an external SIP call control (IP-PBX) by providing basic registrar and back-to-back user agent (B2BUA) services. These services are used by a SIP IP phone in the event of a WAN connection outage when the SIP phone is unable to communicate with its primary SIP proxy.

Cisco SIP SRST Version 3.4 can support SIP phones with standard RFC 3261 feature support locally and across SIP WAN networks. With Cisco SIP SRST Version 3.4, SIP phones can place calls across SIP networks in the same way as Skinny Client Control Protocol (SCCP) phones. For full information about SIP SRST, Version 3.4, see Cisco SIP SRST Version 3.4 System Administrator Guide.

New Features in Cisco SRST Version 3.3

Secure SRST

Secure Cisco IP phones that are located at remote sites and that are attached to gateway routers can communicate securely with Cisco Unified Communications Manager using the WAN. But if the WAN link or Cisco Unified Communications Manager goes down, all communication through the remote phones becomes nonsecure. To overcome this situation, gateway routers can now function in secure SRST mode, which activates when the WAN link or Cisco Unified Communications Manager goes down. When the WAN link or Cisco Unified Communications Manager is restored, Cisco Unified Communications Manager resumes secure call-handling capabilities.

Secure SRST provides new SRST security features such as authentication, integrity, and media encryption. Authentication provides assurance to one party that another party is whom it claims to be. Integrity provides assurance that the given data has not been altered between the entities. Encryption implies confidentiality; that is, that no one can read the data except the intended recipient. These security features allow privacy for SRST voice calls and protect against voice security violations and identity theft. For more information, see Configure Secure SRST for SCCP and SIP, on page 265.

Cisco Unified IP Phone 7970G and Cisco Unified 7971G-GE Support

The Cisco Unified IP Phones 7970G and 7971G-GE are full-featured telephones that provide voice communication over an IP network. They function much like a traditional analog telephones, allowing you to place and receive phone calls and to access features such as mute, hold, transfer, speed dial, call forward, and more. In addition, because the phones are connected to your data network, they offer enhanced IP telephony features, including access to network information and services, and customizable features and services. The phones also support security features that include file authentication, device authentication, signaling encryption, and media encryption.

The Cisco Unified IP Phones 7970G and 7971G-GE also provide a color touchscreen, support for up to eight line or speed-dial numbers, context-sensitive online help for buttons and feature, and a variety of other sophisticated functions. No configurations specific to SRST are necessary.

For more information, see the Cisco Unified IP Phone 7900 Series documentation index.

Note The Cisco Unified IP Phone 7914 Expansion Module can attach to your Cisco Unified IP Phones 7970G and 7971G-GE. See the Cisco Unified IP Phone Expansion Module 7914 Support section for more information.

Enhancement to the show ephone Command

The **show ephone** command is enhanced to display the configuration and status of the Cisco Unified IP Phone 7970G and Cisco Unified IP Phone 7971G-GE. For more information, see the **show ephone** command in Cisco Unified SRST and Cisco Unified SIP SRST Command Reference (All Versions).

New Features in Cisco SRST Version 3.2

Enhancement to the alias Command

The **alias** command is enhanced as follows:

- The **cfw** keyword was added, providing call forward no-answer/busy capabilities.
- The maximum number of **alias** commands used for creating calls to telephone numbers that are unavailable during Cisco Unified Communications Manager fallback was increased to 50.
- The *alternate-number* argument can be used in multiple **alias** commands.

For more information, see the **alias** command in Cisco Unified SRST and Cisco Unified SIP SRST Command Reference (All Versions).

Enhancement to the cor Command

The maximum number of **cor** lists has increased to 20.

For more information, see the **cor** command in Cisco Unified SRST and Cisco Unified SIP SRST Command Reference (All Versions).

Enhancement to the pickup Command

The **pickup** command was introduced to enable the PickUp soft key on all Cisco Unified IP Phones, allowing an external Direct Inward Dialing (DID) call coming into one extension to be picked up from another extension during SRST.

For more information, see the **pickup** command in Cisco Unified SRST and Cisco Unified SIP SRST Command Reference (All Versions).

Enhancement to the user-locale Command

The **user-locale** command is enhanced to display the Japanese Katakana country code. Japanese Katakana is available in Cisco Unified Communications Manager V4.0 or later versions.

For more information, see the **user-locale** command in the Cisco Unified SRST and Cisco Unified SIP SRST Command Reference (All Versions).

Increased the Number of Cisco Unified IP Phones Supported on the Cisco 3845

The Cisco 3845 now supports 720 phones and up to 960 ephone-dns or virtual voice ports.

MOH Live-Feed Support

Cisco Unified SRST is enhanced with the new **moh-live** command. The **moh-live** command provides live-feed MOH streams from an audio device connected to an E&M or FXO port to Cisco IP phones in SRST mode. If an FXO port is used for a live feed, the port must be supplied with an external third-party adaptor to provide a battery feed. Music from a live feed is obtained from a fixed source and is continuously fed into the MOH playout buffer instead of being read from a flash file. Live-feed MOH can also be multicast to Cisco IP phones. See the Appendix B: Integrating Cisco Unified Communications Manager and Cisco Unified SRST to Use Cisco Unified SRST as a Multicast MOH Resource section for configuration instructions.

No Timeout for Call Preservation

To preserve existing H.323 calls on the branch in the event of an outage, disable the H.225 keepalive timer by entering the **no h225 timeout keepalive** command. This feature is supported in Cisco IOS Releases 12.3(7)T1 and higher versions. See the Cisco Unified SRST Feature Overview section for more information.

H.323 is not supported with SIP phones.

RFC 2833 DTMF Relay Support

Cisco Skinny Client Control Protocol (SCCP) phones, such as those used with Cisco SRST systems, provide only out-of-band DTMF digit indications. To enable SCCP phones to send digit information to remote SIP-based IVR and voice-mail applications, Cisco SRST 3.2 and later versions provide conversion from the out-of-band SCCP digit indication to the SIP standard for DTMF relay, which is RFC 2833. You select this method in the SIP VoIP dial peer using the dtmf-relay rtp-nte command. See the How to Configure DTMF Relay for SIP Applications and Voicemail section for configuration instructions.

To use voicemail on a SIP network that connects to a Cisco Unity Express system, use a nonstandard SIP Notify format. To configure the Notify format, use the **sip-notify** keyword with the **dtmf-relay** command. Using the **sip-notify** keyword may be required for backward compatibility with Cisco SRST Versions 3.0 and 3.1.

Translation Profile Support

Cisco SRST 3.2 and later versions support translation profiles. Translation profiles allow you to group translation rules together and to associate translation rules with the following:

- Called numbers
- Calling numbers
- Redirected called numbers

See the Enable Translation Profiles section for more configuration information. For more information on the **translation-profile** command, see Cisco Unified SRST and Cisco Unified SIP SRST Command Reference (All Versions).

New Features in Cisco Unified SRST Version 3.1

Cisco Unified SRST V3.1 introduced the new features described in the following sections:

- Cisco Unified IP Phone 7920 Support
- Cisco Unified IP Phone 7936 Support

Note For information about Cisco Unified IP phones, see the Cisco Unified IP Phone 7900 Series documentation.

Cisco Unified IP Phone 7920 Support

The Cisco Unified Wireless IP Phone 7920 is an easy-to-use IEEE 802.11b wireless IP phone that provides comprehensive voice communications in conjunction with Cisco Unified CM and Cisco Aironet 1200, 1100, 350, and 340 Series of Wi-Fi (IEEE 802.11b) access points. As a key part of the Cisco AVVID Wireless Solution, the Cisco Unified Wireless IP Phone 7920 delivers seamless intelligent services, such as security, mobility, quality of service (QoS), and management, across an end-to-end Cisco network.

No configuration is necessary.

Cisco Unified IP Phone 7936 Support

The Cisco Unified IP Conference Station 7936 is an IP-based, hands-free conference room station that uses VoIP technology. The IP Conference Station replaces a traditional analog conferencing unit by providing business conferencing features—such as call hold, call resume, call transfer, call release, redial, mute, and conference—over an IP network.

No configuration is necessary.

New Features in Cisco SRST Version 3.0

Additional Language Options for IP Phone Display

Displays for the Cisco Unified IP Phone 7940G and Cisco Unified IP Phone 7960G can be configured with extra ISO-3166 codes for German, Danish, Spanish, French, Italian, Japanese, Dutch, Norwegian, Portuguese, Russian, Swedish, United States.

Note This feature is available only for Cisco Unified SRST running under Cisco Unified Communications Manager V3.2.

Consultative Call Transfer and Forward Using H.450.2 and H.450.3 for SCCP Phones

Cisco Unified SRST V1.0, Cisco Unified SRST V2.0, and Cisco Unified SRST V2.1 allow blind Call Transfers and blind call forwarding. Blind calls do not give transferring and forwarding parties the ability to announce or consult with destination parties. These three versions of Cisco Unified SRST use a Cisco Unified SRST proprietary mechanism to perform blind transfers. Cisco Unified SRST V3.0 adds the ability to perform Call Transfers with consultation using the ITU-T H.450.2 (H.450.2) standard and call forwarding using the ITU-T H.450.3 (H.450.3) standard for H.323 calls.

Cisco Unified SRST V3.0 provides support for IP phones to initiate Call Transfer and forwarding with H.450.2 and H.450.3 by using the default session application. The built-in H.450.2 and H.450.3 support that is provided by the default session application applies to Call Transfers and call forwarding initiated by IP phones, regardless of the PSTN interface type.

Note All voice gateway routers in the VoIP network must support H.450. For H.450 support, routers with Cisco Unified SRST must run either Cisco Unified SRST V3.0 and higher versions or Cisco IOS Release 12.2(15)ZJ and later releases. Routers without Cisco Unified SRST must run either Cisco Unified SRST V2.1 and higher versions or Cisco IOS Release 12.2(11)YT and later releases. SIP phones do not support this feature.

For more information about the default session application, see the Default Session Application Enhancements Guide.

For configuration information, see the Enabling Consultative Call Transfer and Forward Using H.450.2 and H.450.3 with Cisco Unified SRST 3.0 section.

Customized System Message for Cisco Unified IP Phones

The display message that appears on Cisco Unified IP Phone 7905G, Cisco Unified IP Phone 7940G, Cisco Unified IP Phone 7960G, and Cisco Unified IP Phone 7910 units when they are in fallback mode can be customized. The new system message command allows you to edit these display messages on a per-router basis. The custom system message feature supports English only.

For further information, see the Configuring Customized System Messages for Cisco Unified IP Phones section.

Dual-Line Mode

A new keyword that was added to the **max-dn** command allows you to set IP phones to dual-line mode. Each dual-line IP phone must have one voice port and two channels to handle two independent calls. This mode enables call waiting, Call Transfer, and conference functions on a single ephone-dn (ephone directory number). There is a maximum number of DNs available during Cisco Unified SRST fallback. The **max-dn** command affects all IP phones on a Cisco Unified SRST router.

For configuration information, see the Configuring Dual-Line Phones section.

E1 R2 Signaling Support

Cisco Unified SRST V3.0 supports E1 R2 signaling. R2 signaling is an international signaling standard that is common to channelized E1 networks; however, there is no single signaling standard for R2. The ITU-T Q.400-Q.490 recommendation defines R2, but several countries and geographic regions implement R2 in entirely different ways. Cisco addresses this challenge by supporting many localized implementations of R2 signaling in its Cisco IOS Software.

The Cisco E1 R2 signaling default is ITU, which supports the following countries: Denmark, Finland, Germany, Russia (ITU variant), Hong Kong (ITU variant), and South Africa (ITU variant). The expression “ITU variant” means that there are multiple R2 signaling types in the specified country, but Cisco supports the ITU variant.

Cisco also supports specific local variants of E1 R2 signaling in the following regions, countries, and corporations:

- Argentina
- Australia
- Bolivia

- Brazil
- Bulgaria
- China
- Colombia
- Costa Rica
- East Europe (includes Croatia, Russia, and Slovak Republic)
- Ecuador (ITU)
- Ecuador (LME)
- Greece
- Guatemala
- Hong Kong (uses the China variant)
- Indonesia
- Israel
- Korea
- Laos
- Malaysia
- Malta
- New Zealand
- Paraguay
- Peru
- Philippines
- Saudi Arabia
- Singapore
- South Africa (Panaftel variant)
- Telmex Corporation (Mexico)
- Telnor Corporation (Mexico)
- Thailand
- Uruguay
- Venezuela
- Vietnam

European Date Formats

The date format on a Cisco IP phone display can be configured with the following two extra formats:

- yy-mm-dd (year-month-day)
- yy-dd-mm (year-day-month)

For configuration information, see the Configuring IP Phone Clock, Date, and Time Formats section.

Huntstop for Dual-Line Mode

A new keyword was added to the huntstop command. The **channel** keyword causes hunting to skip the secondary channel in dual-line configuration if the primary line is busy or does not answer.

For configuration information, see the Configuring Dial-Peer and Channel Hunting section.

Music On Hold for Multicast from Flash Files

You can configure Cisco Unified SRST to support continuous multicast output of MOH from a flash MOH file in flash memory.

For more information, see the Defining XML API Schema section.

Ringing Timeout Default

A ringing timeout default can be configured for extensions on which no-answer call forwarding has not been enabled. Expiration of the timeout causes incoming calls to return a disconnect code to the caller. This mechanism provides protection against hung calls for inbound calls received over interfaces such as Foreign Exchange Office (FXO) that do not have forward-disconnect supervision. For more information, see the Configuring the Ringing Timeout Default section.

Secondary Dial Tone

Secondary dial tone is available for Cisco Unified IP Phones running Cisco Unified SRST. The secondary dial tone is generated when you dial a predefined PSTN access prefix. For example, you would hear different dial tone when a designated number is pressed to reach an outside line.

The secondary dial tone is created through the secondary dial tone command. For more information, see the Configuring a Secondary Dial Tone section.

Enhancement to the Show ephone Command

The **show ephone** command is enhanced to display the following:

- Configuration and status of additional phones (new keywords: **7905, 7914, 7935, ATA**)
- Status of all phones with the call-forwarding all (CFA) feature enabled on at least one of their DNs (new keyword: **cfa**)

For more information, see the **show ephone** command in Cisco Unified SRST and Cisco Unified SIP SRST Command Reference (All Versions).

System Log Messages for Phone Registrations

Diagnostic messages are added to the system log whenever a phone registers or unregisters from Cisco Unified SRST.

Three-Party G.711 Ad Hoc Conferencing

Cisco Unified SRST supports three-party instant meeting conferencing using the G.711 coding technique. For conferencing to be available, connect two lines to one or more buttons of an IP phone.

For more information, see the Enabling Three-Party G.711 Ad Hoc Conferencing section.

Support for Cisco VG248 Analog Phone Gateway 1.2(1) and Higher Versions

The Cisco VG248 Analog Phone Gateway is a mixed-environment solution, enabled by Cisco Unified Communications system. It allows organizations to support their legacy analog devices while taking advantage of the new opportunities afforded by using IP telephony. The Cisco VG248 is a high-density gateway for using analog phones, fax machines, modems, voicemail systems, and speakerphones within an enterprise voice system based on Cisco Unified Communications Manager.

During Cisco Unified Communications Manager fallback, Cisco Unified SRST considers the Cisco VG248 to be a group of Cisco Unified IP Phones. Cisco Unified SRST counts each of the 48 ports on the Cisco VG248 as a separate Cisco Unified IP Phone. Support for Cisco VG248 Version 1.2(1) and higher versions is also available in Cisco Unified SRST Version 2.1.

For more information, see Cisco VG248 Analog Phone Gateway Data Sheet and Cisco VG248 Analog Phone Gateway Version 1.2(1) Release Notes.

New Features in Cisco SRST Version 2.1

Cisco SRST V2.1 introduced the new features described in the following sections:

- Additional Language Options for IP Phone Display
- Cisco Unified SRST Aggregation
- Cisco ATA 186 and ATA 188 Support
- Cisco Unified IP Phone 7902G Support
- Cisco Unified IP Phone 7905G Support
- Cisco Unified IP Phone 7912G Support
- Cisco Unified IP Phone Expansion Module 7914 Support
- Enhancement to the Dial Plan-Pattern Command

Note For information about Cisco Unified IP phones, see the Cisco Unified IP Phone 7900 Series documentation.

Additional Language Options for IP Phone Display

Displays for the Cisco Unified IP Phone 7940G and Cisco Unified IP Phone 7960G can be configured with ISO-3166 codes for the following countries:

- France
- Germany
- Italy
- Portugal
- Spain
- United States

Note This feature is available only in Cisco Unified SRST running under Cisco Unified Communications Manager V3.2.

For configuration information, see the Configuring IP Phone Language Display section.

Cisco Unified SRST Aggregation

For systems running Cisco Unified Communications Manager 3.3(2) and later versions, the restriction of running Cisco Unified SRST on a default gateway was removed. Multiple SRST routers can be used to support more phones. Carefully plan and configure the dial peers and dial plans for Call Transfer and forwarding to work properly.

Cisco ATA 186 and ATA 188 Support

The Cisco ATA analog phone adapters are handset-to-Ethernet adapters that allow regular analog phones to operate on IP-based telephony networks. Cisco ATAs support two voice ports, each with an independent phone number. The Cisco ATA 188 also has an RJ-45 10/100BASE-T data port. Cisco Unified SRST supports Cisco ATA 186 and Cisco ATA 188 using Skinny Client Control Protocol (SCCP) for the voice calls only.

Cisco Unified IP Phone 7902G Support

The Cisco Unified IP Phone 7902G is an entry-level IP phone that addresses the voice communications needs of a lobby, laboratory, manufacturing floor, hallway, or other area where only basic calling capability is required.

The Cisco Unified IP Phone 7902G is a single-line IP phone with fixed feature keys that provide one-touch access to the redial, transfer, conference, and voicemail access features. Consistent with other Cisco IP phones, the Cisco Unified IP Phone 7902G supports inline power, which allows the phone to receive power over the LAN. This capability gives the network administrator centralized power control and thus greater network availability.

Cisco Unified IP Phone 7905G Support

The Cisco Unified IP Phone 7905G is a basic IP phone that provides a core set of business features. It provides single-line access and four interactive softkeys that guide a user through call features and functions via the pixel-based LCD. The graphic capability of the display presents calling information, intuitive access to features, and language localization in future firmware releases. The Cisco Unified IP Phone 7905G supports inline power, which allows the phone to receive power over the LAN.

No configuration is necessary.

Cisco Unified IP Phone 7912G Support

The Cisco Unified IP Phone 7912G provides core business features and addresses the communication needs of a cubicle worker who conducts low to medium phone traffic. Four dynamic softkeys provide access to call features and functions. The graphic display shows calling information and allows access to features.

The Cisco Unified IP Phone 7912G supports an integrated Ethernet switch, providing LAN connectivity to a local PC. In addition, the Cisco Unified IP Phone 7912G supports inline power, which allows the phone to receive power over the LAN. This capability gives the network administrator centralized power control and thus greater network availability. The combination of inline power and Ethernet switch support reduces cabling needs from a single wire to the desktop.

Cisco Unified IP Phone Expansion Module 7914 Support

The Cisco Unified IP Phone 7914 Expansion Module attaches to your Cisco Unified IP Phone 7960G, adding 14 line appearances or speed-dial numbers to your phone. You can attach one or two expansion modules to your IP phone. When you use two expansion modules, you have 28 additional line appearances or speed-dial numbers or a total of 34 line appearances or speed-dial numbers.

Enhancement to the Dial Plan-Pattern Command

A new keyword was added to the **dialplan-pattern** command. The extension-pattern keyword sets an extension number’s leading digit pattern when it is different from the E.164 phone number’s leading digits defined in the *pattern* variable. This enhancement allows manipulation of IP phone abbreviated extension number prefix digits. See the **dialplan-pattern** command in Cisco Unified SRST and Cisco Unified SIP SRST Command Reference (All Versions).

New Features in Cisco SRST Version 2.02

Cisco Unified IP Phone Conference Station 7935 Support

The Cisco IP Conference Station 7935 is an IP-based, full-duplex hands-free conference station for use on desktops and offices and in small-to-medium-sized conference rooms. This device attaches a Cisco Catalyst 10/100 Ethernet switch port with a simple RJ-45 connection and dynamically configures itself to the IP network via the DHCP. Other than connecting the Cisco 7935 to an Ethernet switch port, no further administration is necessary. The Cisco 7935 dynamically registers to Cisco Unified CM for connection services and receives the appropriate endpoint phone number and any software enhancements or personalized settings, which are preloaded within Cisco Unified CM.

The Cisco Unified IP Phone 7935 provides three soft keys and menu navigation keys that guide a user through call features and functions. The Cisco Unified IP Phone 7935 also features a pixel-based LCD display. The display provides features such as date and time, calling party name, calling party number, digits dialed, and feature and line status. No configuration is necessary.

Increase in Directory Numbers

The following table shows the increases in directory numbers.

Cisco Router	Maximum Phones	Increase in Maximum Directory Number	
		From	To
Cisco 1751	24	96	120
Cisco 1760	24	96	120
Cisco 2600XM	24	96	120
Cisco 2691	72	216	288
Cisco 3640	72	216	288
Cisco 3660	240	720	960
Cisco 3725	144	432	576
Cisco 3745	240	720	960

Cisco Unity Voicemail Integration Using In-Band DTMF Signaling Across the PSTN and BRI/PRI

Cisco Unity voicemail and other voicemail systems can be integrated with Cisco Unified SRST. Voicemail integration introduces six new commands:

- Pattern direct
- Pattern ext-to-ext busy
- Pattern ext-to-ext no-answer
- Pattern trunk-to-ext busy
- Pattern trunk-to-ext no-answer
- Vm-integration

CHAPTER 2

Cisco Unified SRST Feature Overview

This chapter describes Cisco Unified Survivable Remote Site Telephony (Cisco Unified SRST) and what it does. It also includes information about support for Cisco Unified IP Phones and Platforms, specifications, features, prerequisites, restrictions and where to find additional reference documents.

SRST Overview

Cisco Unified SRST provides Cisco Unified CM with fallback support for Cisco Unified IP phones that are attached to a Cisco router on your local network. Cisco Unified SRST enables routers to provide call-handling support for Cisco Unified IP phones when they lose connection to remote primary, secondary, or tertiary Cisco Unified CM installations or when the WAN connection is down.

Cisco Unified CM supports Cisco Unified IP phones at remote sites attached to Cisco multiservice routers across the WAN. Before Cisco Unified SRST, when the WAN connection between a router and the Cisco Unified CM failed or when connectivity with Cisco Unified CM was lost for some reason, Cisco Unified IP phones on the network became unusable for the duration of the failure. Cisco Unified SRST overcomes this problem and ensures that the Cisco Unified IP phones offer continuous (although minimal) service by providing call-handling support for Cisco Unified IP phones directly from the Cisco Unified SRST router. The system automatically detects a failure and uses Simple Network Auto Provisioning (SNAP) technology to autoconfigure the branch office router to provide call processing for Cisco Unified IP phones that are registered with the router. When the WAN link or connection to the primary Cisco Unified CM is restored, call handling reverts to the primary Cisco Unified CM.

When Cisco Unified IP phones lose contact with primary, secondary, and tertiary Cisco Unified CM, they must establish a connection to a local Cisco Unified SRST router to sustain the call-processing capability necessary to place and receive calls. The Cisco Unified IP phone retains the IP address of the local Cisco Unified SRST router as a default router in the Network Configuration area of the Settings menu. The Settings menu supports a maximum of five default router entries; however, Cisco Unified CM accommodates a maximum of three entries. When a secondary Cisco Unified CM is not available on the network, the local

Cisco Unified SRST Router's IP address is retained as the standby connection for Cisco Unified CM during normal operation.

Note Cisco Unified CM fallback mode telephone service is available only to those Cisco Unified IP phones that are supported by a Cisco Unified SRST router. Other Cisco Unified IP phones on the network remain out of service until they re-establish a connection with their primary, secondary, or tertiary Cisco Unified CM.

How Fallback Occurs

Typically, it takes three times the keepalive period for a phone to discover that its connection to Cisco Unified CM has failed. The default keepalive period is 30 seconds. If the phone has an active standby connection established with a Cisco Unified SRST router, the fallback process takes 10 to 20 seconds after connection with Cisco Unified CM is lost. An active standby connection to a Cisco Unified SRST router exists only if the phone has the location of a single Cisco Unified CM in its Unified Communications Manager list. Otherwise, the phone activates a standby connection to its secondary Cisco Unified CM.

If a Cisco Unified IP phone has multiple Cisco Unified CM in its Cisco Unified CM list, it progresses through its list of secondary and tertiary Cisco Unified CM before attempting to connect with its local Cisco Unified SRST router. Therefore, the time that passes before the Cisco Unified IP phone eventually establishes a connection with the Cisco Unified SRST router increases with each attempt to contact to a Cisco Unified CM. If each attempt to connect to a Cisco Unified CM takes about 1 minute, the Cisco Unified IP phone in question could remain offline for 3 minutes or more following a WAN link failure.

Note

- The time it takes for a Cisco Unified IP Phone to fall back to the SRST router can vary depending on the phone type. Phones such as the Cisco 7902, Cisco 7905, and Cisco 7912 can take approximately 2.5 minutes to fall back to the SRST mode.
- During a WAN connection failure, when Cisco Unified SRST is enabled, Cisco Unified IP phones display a message informing you that they are operating in Cisco Unified CM fallback mode. For example, the Cisco Unified IP Phone 7960G and Cisco Unified IP Phone 7940G display a "CM Fallback Service Operating" message, and the Cisco Unified IP Phone 7910 displays a "CM Fallback Service" message when operating in Cisco Unified CM fallback mode. When the Cisco Unified CM is restored, the message goes away and full Cisco Unified IP phone functionality is restored.

Resumption of Primary Call Control

While in Cisco Unified CM fallback mode, Cisco Unified IP phones periodically attempt to re-establish a connection with Cisco Unified CM at the central office. Generally, the default time that Cisco Unified IP phones wait before attempting to re-establish a connection to a remote Cisco Unified CM is 120 seconds. The time can be changed in Cisco Unified CM by editing the Connection Monitor Duration parameter. See the "Configure SRST” chapter of the *System Configuration Guide for Cisco Unified Communications Manager*. A manual reboot can immediately reconnect Cisco Unified IP phones to Cisco Unified CM.

When a connection is re-established with Cisco Unified CM, Cisco Unified IP phones automatically cancel their registration with the Cisco Unified SRST Router. However, if a WAN link is unstable, Cisco Unified IP phones can bounce between Cisco Unified CM and Cisco Unified SRST. A Cisco Unified IP phone cannot re-establish a connection with the primary Cisco Unified CM at the central office if it is currently engaged in an active call.

Supported Call Combinations

Cisco Unified SRST supports the following call combinations:

- SIP phone to SIP phone
- SIP phone to SCCP phone
- SIP phone to PSTN/router voice-port
- SIP phone to WAN VoIP using SIP
- SCCP phone to SIP phone
- SCCP phone to SCCP phone
- SCCP phone to PSTN/router voice-port
- SCCP phone to WAN VoIP using SIP or H.323

The following figure shows a remote site that connects to primary call control over a WAN IP connection. In this example, the WAN is down, making primary call control impossible to reach via IP networks. The SRST router acts as a fallback server, providing backup call control for IP Phones at the remote site, which can still use the PSTN for external calls, and for calls to phones that still register to the primary site.

Figure 1: Branch Office Cisco Unifed IP Phones Connected to a Remote Central Cisco Unified Communications Manage Operating in SRST Mode

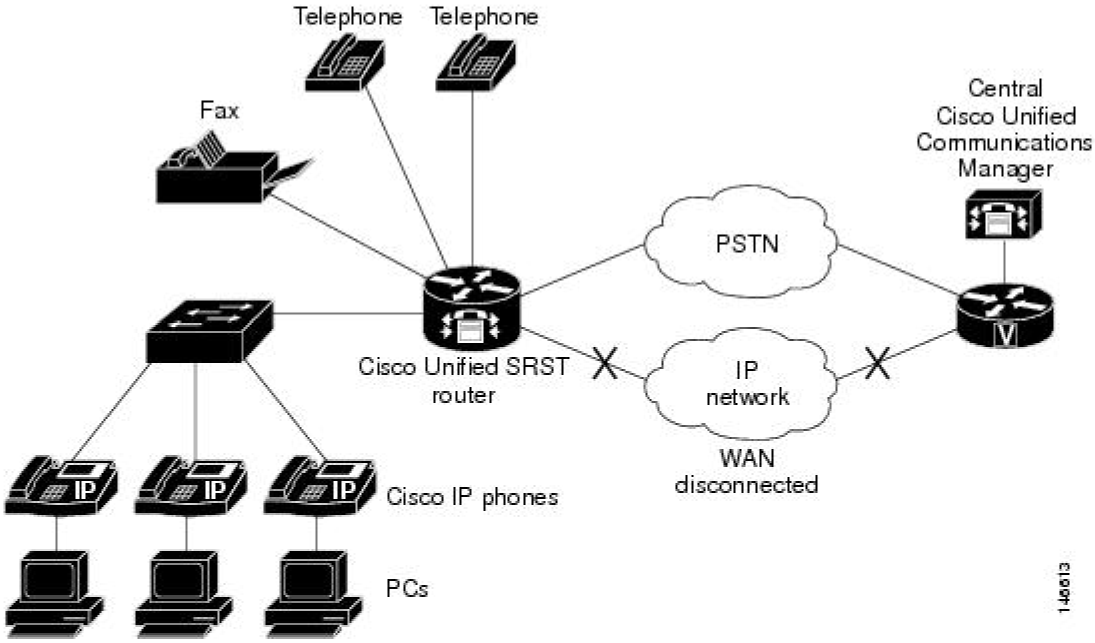


Figure 1: Branch Office Cisco Unified IP Phones Connected to a Remote Central Cisco Unified Communications Manage Operating in SRST Mode

SRST Operating Modes

SRST can be deployed in any of the following operating modes.

- Unified SRST mode (the default operating mode)
- Enhanced SRST mode
- Webex Survivability Gateway mode

Note that you cannot run SRST in more than one of these modes at the same time. However, the basic feature set of Unified SRST support is included with both Enhanced SRST mode or Webex Survivability Gateway mode. As a result, if you enable either of these modes, your router also supports Unified SRST features.

Unified SRST Mode

By default, SRST is running in Unified SRST mode, unless one of the other modes has been enabled. Unified SRST supports basic call failover service for SIP or SCCP endpoints. Support is for audio-only calls with base features such as Call Transfer, Conference and Music on Hold.

No specific configuration is required to enabled Unified SRST mode. Once calling services are enabled on the router, Unified SRST is configured by default. If you have one of the other operating modes enabled (Enhanced SRST or Webex Survivability Gateway mode), you can revert to Unified SRST mode by running the **default mode** command while in voice register global configuration mode.

Note If you enable either Enhanced SRST mode or Webex Survivability Gateway mode, your router supports the same features as Unified SRST even though the router is not running in Unified SRST mode.

Enhanced SRST Mode

Enhanced SRST mode can be enabled by running the **mode esrst** command while in voice register global configuration mode or in telephony-service configuration mode.

Enhanced SRST mode provides the same support as Unified SRST mode, but adds advanced calling features such as:

- Video calls (local calling only)
- Shared Line
- BLF
- B-ACD
- cBarge
- Privacy on Hold
- Voice hunt group support is enhanced to include:
  - Shared Lines
  - Mixed Shared Lines (SIP and SCCP)
  - Hunt Statistics Collection
  - Mixed Deployment (SIP and SCCP)

For more information, including configuration info, see the Enhanced SRST, on page 121 chapter.

Webex Survivability Gateway Mode

Webex Survivability Gateway mode provides Site Survivability for Webex Calling endpoints. If you're depoying Webex Calling, configure this mode on a gateway in the local network. This operating mode is enabled by running the **mode Webex-sgw** command while in voice register global configuration mode.

To configure Webex Survivability Gateway mode on a gateway, see the Webex article Site Survivability for Webex Calling.

Survivability Gateway Colocation with Unified SRST

As of Cisco IOS XE 17.9.3 and Cisco IOS XE Dublin 17.11.1a onwards, you can colocate a Survivability Gateway configuration and a Unified SRST mode configuration on the same gateway. This feature lets your gateway support survivability for Webex Calling endpoints and on-premises endpoints that register to Unified Communications Manager.

To configure colocation, the gateway must be in Webex Survivability Gateway mode. Do the following:

- Complete the procedures in the preceding Webex article link to configure Webex Survivability Gateway mode on a gateway.
- Complete the procedures in the subsequent chapters of this document to configure the same gateway with Unified SRST survivability for on-premises endpoints.

Call routing considerations for colocation

Consider the following when configuring call routing for colocation scenarios:

- The Survivability Gateway routes internal calls automatically provided that both endpoints in the call are registered to the Survivability Gateway. Internal calls are automatically routed between any registered clients (SRST or Webex Calling).
- It's possible to have a situation where the connection to one call control system goes down while the connection to the other call control system remains up. As a result, one set of endpoints registers to the Survivability Gateway while another set of endpoints at the same site registers to primary call control. In this case, you may need to route calls between the two sets of endpoints to a SIP trunk or PSTN circuit.
- External calls and E911 calls can be routed to a SIP trunk or PSTN circuit.

Secure SRST

Secure SRST refers to security features that can be enabled for any of the SRST operating modes: Unified SRST mode, Enhanced SRST mode, or Webex Survivability Gateway mode. Secure SRST provides security features such as TLS 1.2 signaling and SRTP media using secure encryption ciphers. For SIP registrations, you can enable SIP OAuth authentication or apply a security policy that blocks nonsecure registrations, adding more security to your deployment.

Note TLS version 1.3 security feature is not supported for Webex Survivability Gateway operating mode.

Starting from Cisco Unified SRST 14.4 Release (Cisco IOS XE 17.14.1a), SRST security feature is enhanced to support TLS version 1.3 in addition to TLS versions 1.0, 1.1 and 1.2 and associated ciphers. It is recommended that TLS version 1.2 or 1.3 is used wherever possible to ensure security or compliance. The following functionalities are supported with secure SRST:

- The TLS exclusivity functionality enables only the configured version of TLS (1.0 or 1.1 or 1.2 or 1.3).
- In the default form, all the TLS versions 1.3, 1.2, and 1.1 are supported. However, to configure TLS v1.0, you must explicitly specify the TLS version.
- In sip-ua configuration mode, SIP SRST supports minimum TLS version functionality. You can configure the minimum TLS version only with TLS v1.2, which supports both TLS v1.2 and v1.3 cipher negotiations.

Secure SRST resolves a situation that can occur for secure Cisco IP phones during network failure situations. Secure Cisco IP phones that are located at remote sites and that are attached to gateway routers can communicate securely with Unified Communications Manager using the WAN. However, if the network connection breaks, either because of a WAN link failure, or because of a Unified Communications Manager server failure, all communication through the remote phones becomes nonsecure by default. Secure SRST overcomes this situation by providing security features that are active while the endpoints are registered to the SRST router.

Secure SRST provides authentication, integrity, and media encryption.

- Authentication provides assurance to one party that another party is whom it claims to be.
- Integrity provides assurance that the given data has not been altered between the entities.
- Encryption implies confidentiality, that is, that no one can read the data except the intended recipient.

These security features allow privacy for SRST voice calls and protect against voice security violations and identity theft.

For more information on how to configure security for SRST, see Configure Secure SRST for SCCP and SIP, on page 265.

SRST for SIP Networks

This guide describes Cisco Unified SRST functionality for SIP networks. Cisco Unified SIP SRST provides backup to an external SIP call control (IP-PBX) by providing basic registrar and redirect server or back-to-back user agent (B2BUA) services. These services are used by a SIP IP phone in the event of a WAN connection outage when the SIP phone is unable to communicate with its primary SIP proxy.

Cisco Unified SIP SRST can support SIP phones with standard RFC 3261 feature support locally and across SIP WAN networks. With Cisco Unified SIP SRST, SIP phones can place calls across SIP networks in the same way as SCCP phones.

SIP proxy, registrar, and B2BUA servers are key components of a SIP VoIP network. These servers are usually located in the core of a VoIP network. If SIP phones located at remote sites at the edge of the VoIP network lose connectivity to the network core (because of a WAN outage), they may be unable to make or receive calls. Cisco Unified SIP SRST functionality on a SIP PSTN gateway provides service reliability for SIP-based IP phones in the event of a WAN outage. Cisco Unified SIP SRST enables the SIP IP phones to continue to make and receive calls to and from the PSTN and also to make and receive calls to and from other SIP IP phones.

To see a branch office Cisco Unifed IP Phones connected to a remote central Cisco Unified CM Operating in SRST mode, see Figure Branch Office Cisco Unifed IP Phones Connected to a Remote Central Cisco Unified Communications Manage Operating in SRST Mode.

Prerequisites for Configuring Cisco Unified SIP SRST

Before configuring Cisco Unified SIP SRST, you must do the following:

An SRST feature license is required to enable the Cisco Unified SIP SRST feature. Contact your account representative if you have further questions. For more information about Licensing on Unified SRST, refer to Licensing section in Cisco Unified SIP SRST on Cisco 4000 Series Integrated Services Router chapter.

Restrictions for Configuring Cisco Unified SIP SRST

The following table provides a history of restrictions from Cisco SIP SRST Version 3.0 to the present version of Cisco Unified SIP SRST.

Cisco Unified SRST Version	Cisco IOS Release	Restrictions
Version 8.0	15.1(1)T	SIP phones may be configured on the Cisco Unified CM with an Authenticated device security mode. The Cisco Unified CM ensures integrity and authentication for the phone using a TLS connection with NULL-SHA cipher for signaling. If such an Authenticated SIP phone fails over to the Cisco Unified SRST device, and if the Cisco Unified CM and SRST device are configured to support secure SIP SRST, it will register using TCP instead of TLS/TCP, thus disabling the Authenticated mode until the phone fails back to the Cisco Unified CM.

Cisco Unified SRST Version	Cisco IOS Release	Restrictions
Version 4.1	12.4.(15)T	

Cisco Unified SRST Version	Cisco IOS Release	Restrictions
		• Cisco Unified SRST does not support BLF speed-dial notification, call forward all synchronization, dial plans, directory services, or music-on-hold (MOH). • Prior to SIP phone load 8.0, SIP phones maintained dual registration with both Cisco Unified Communications Manager and Cisco Unified SRST simultaneously. In SIP phone load 8.0 and later versions, SIP phones use keepalive to maintain a connection with Cisco Unified SRST during active registration with Cisco Unified Communications Manager. Every two minutes, a SIP phone sends a keepalive message to Cisco Unified SRST. Cisco Unified SRST responds to this keepalive with a 404 message. This process repeats until fallback to Cisco Unified SRST occurs. After fallback, SIP phones send a keepalive message every two minutes to Cisco Unified Communications Manager while the phones are registered with Cisco Unified SRST. Cisco Unified SRST continues to support dual registration for SIP phone loads older than 8.0. • Enhanced 911 Services for Cisco Unified SRST does not interface with the Cisco Emergency Responder. • The information about the most recent phone that called 911 is not preserved after a reboot of Cisco Unified SRST. • Cisco Emergency Responder does not have access to any updates made to the emergency call history table when remote IP Phones are in Cisco Unified SRST fallback mode. Therefore, if the PSAP calls back after the Cisco Unified IP Phones register back to Cisco Unified Communications Manager, Cisco Emergency Responder will not have any history of those calls. As a result, those calls will not get routed to the original 911 caller. Instead, the calls are routed to the default destination that is configured on Cisco Emergency Responder for the corresponding ELIN. • For Cisco Unified Wireless 7920 and 7921 IP Phones, a caller's location can only be determined by the static information configured by the system administrator. For more information, see Precautions for Mobile Phones in Configuring Enhanced 911 Services. • The extension numbers of 911 callers can be translated to only two emergency location identification numbers (ELINs) for each emergency response location (ERL). • Using ELINs for multiple purposes can result in unexpected interactions with existing Cisco Unified SRST features. These multiple uses of an ELIN can include configuring an ELIN for use as an actual phone number (ephone-dn, voice register dn, or FXS destination-pattern), a Call Pickup number, or an alias rerouting number. For more information, see Multiple Usages of an ELIN in Configuring Enhanced 911 Services. • There are a number of other ways that your configuration of Enhanced 911 Services can interact with existing Cisco Unified SRST features and cause

Cisco Unified SRST Version	Cisco IOS Release	Restrictions
		unexpected behavior. For a complete description of interactions between Enhanced 911 Services and existing Cisco Unified SRST features, see the Interactions with Existing Cisco Unified CME Features in Configuring Enhanced 911 Services.

Cisco Unified SRST Version	Cisco IOS Release	Restrictions
Version 4.0 Version 3.4 Version 3.2 Version 3.1 Version 3.0	12.4(4)XC 12.4(4)T 12.3(11)T 12.3(7)T 12.2(15)ZJ 12.3(4)T	Not Supported • MOH is not supported for a call hold invoked from a SIP phone. A caller hears only silence when placed on hold by a SIP phone. • As of Cisco IOS Release 12.4(4)T, bridged call appearance, find-me, incoming call screening, paging, SIP presence, call park, call pickup, and SIP location are not supported. • SIP-NAT is not supported. • Cisco Unity Express is not supported. • Transcoding is not supported. Phone Features • For call waiting to work on the Cisco ATA and Cisco IP Phone 7912 and Cisco Unified IP Phone 7905G with a 1.0(2) build, the incoming call leg should be configured with the G.711 codec. **Note** Cisco Unified IP Phone 7905G, Cisco Unified IP Phone 7912G, and Cisco Analog Telephone Adaptor (ATA) 186 are not capable of dual registration; thus they are not supported and have limited functionality with Cisco Unified SIP SRST. General • Call detail records (CDRs) are only supported by standard IOS RADIUS support; CDRs are not supported otherwise. • All calls must use the same codec, either G.729r8 or G.711. • Calls that have been transferred cannot be transferred a second time. • URL dialing is not supported. Only number dialing is supported. • The SIP registrar functionality provided by Cisco Unified SIP SRST provides no security or authentication services. • SIP IP phones that do not support dual concurrent registration with both their primary and their backup SIP proxy or registrar may be unable to receive incoming calls from the Cisco Unified SIP SRST gateway during a WAN outage. These phones may take a significant amount of time to discover that their primary SIP proxy or registrar is unreachable before they initiate a fallback registration to their backup proxy or registrar (the SIP SRST gateway). • SIP-phone-to-SIP-trunk support requires Refer and 302/300 Redirection to be supported by the SIP trunk (Version 3.0).

SRST for SCCP Devices

Cisco Unified SRST provides Cisco Unified CM with fallback support for SCCP-based Cisco IP phones that are attached to a Cisco router on your local network. You can deploy SRST for SCCP phones in either of the following modes:

- Unified SRST mode
- Enhanced SRST mode

Prerequisites for Configuring Cisco Unified SCCP SRST

Before configuring Cisco Unified SRST, you must do the following:

- An SRST feature license is required to enable the Cisco Unified SCCP SRST feature. Contact your account representative if you have further questions. For more information about Licensing on Unified SRST, refer Licensing.
- You have an account on Cisco.com to download software.

  To obtain an account on Cisco.com, go to http://www.cisco.com and click**Register** at the top of the screen.

Restrictions for Configuring Cisco Unified SCCP SRST

The following table provides a history of restrictions from Cisco SCCP SRST Version 1.0 to the present version of Cisco Unified SCCP SRST.

Cisco Unified SRST Version	Cisco IOS Release	Restrictions
Version 4.1	12.4.(15)T	• Enhanced 911 Services for Cisco Unified SRST does not interface with the Cisco Emergency Responder. • The information about the most recent phone that called 911 is not preserved after a reboot of Cisco Unified SRST. • Cisco Emergency Responder does not have access to any updates made to the emergency call history table when remote IP phones are in Cisco Unified SRST fallback mode. Therefore, if the PSAP calls back after the Cisco Unified IP phones register back to Cisco Unified Communications Manager, Cisco Emergency Responder will not have any history of those calls. As a result, those calls will not get routed to the original 911 caller. Instead, the calls are routed to the default destination that is configured on Cisco Emergency Responder for the corresponding ELIN. • For Cisco Unified Wireless IP Phone 7920 and 7921, a caller's location can only be determined by the static information configured by the system administrator. For more information, see the Precautions for Mobile Phones in Configuring Enhanced 911 Services. • The extension numbers of 911 callers can be translated to only two emergency location identification numbers (ELINs) for each emergency response location (ERL). • Using ELINs for multiple purposes can result in unexpected interactions with existing Cisco Unified SRST features. These multiple uses of an ELIN can include configuring an ELIN for use as an actual phone number (ephone-dn, voice register dn, or FXS destination-pattern), a Call Pickup number, or an alias rerouting number. For more information, see the Multiple Usages of an ELIN in Configuring Enhanced 911 Services . • There are a number of other ways that your configuration of Enhanced 911 Services can interact with existing Cisco Unified SRST features and cause unexpected behavior. For a complete description of interactions between Enhanced 911 Services and existing Cisco Unified SRST features, see the Interactions with Existing Cisco Unified CME Features in Configuring Enhanced 911 Services.

Cisco Unified SRST Version	Cisco IOS Release	Restrictions
Version 4.0 Version 3.4 Version 3.2 Version 3.1 Version 3.0 Version 2.1 Version 2.02 Version 2.01 Version 2.0	12.4(4)XC 12.4(4)T 12.3(11)T 12.3(7)T 12.2(15)ZJ 12.3(4)T 12.2(15)T 12.2(13)T 12.2(11)T 12.2(8)T1 12.2(8)T 12.2(2)XT	• All of the restrictions in Cisco SRST Version 1.0. • Caller-id display on supported Cisco Unified IP phones: SIP phones in fallback mode displays the name and number of the caller. SCCP phones in fallback mode display only the caller-id number assigned to the line; the caller-ID name configuration for SCCP phones is not preserved during SRST fallback. Call transfer is supported only on the following: • VoIP H.323, VoFR, and VoATM between Cisco gateways running Cisco IOS Release 12.2(11)T and using the H.323 nonstandard information element • FXO and FXS loop-start (analog) • FXO and FXS ground-start (analog) • Ear and mouth (E&M) (analog) and DID (analog) • T1 channel-associated signaling (CAS) with FXO and FXS ground-start signaling • T1 CAS with E&M signaling • All PRI and BRI switch types The following Cisco Unified IP Phone function keys are dimmed because they are not supported during SRST operation: • MeetMe • GPickUp (group pickup) • Park • Confrn (conference) • Although the Cisco IAD2420 series integrated access devices (IADs) support the Cisco Unified SRST feature, this feature is not recommended as a solution for enterprise branch offices.

Cisco Unified SRST Version	Cisco IOS Release	Restrictions
Version 1.0	12.2(2)XB 12.2(2)XG 12.1(5)YD	• Does not support first generation Cisco Unified IP phones, such as Cisco IP Phone 30 VIP and Cisco IP Phone 12 SP+. • Does not support other Cisco Unified Communications Manager applications or services: Cisco IP SoftPhone, Cisco One: Voice and Unified Messaging Application, or Cisco IP Contact Center. • Does not support Centralized Automatic Message Accounting (CAMA) trunks on the Cisco 3660 routers. **Note** If you are in one of the states in the United States of America where there is a regulatory requirement for CAMA trunks to interface to 911 emergency services, and you would like to connect more than 48 Cisco Unified IP phones to the Cisco 3660 multiservice routers in your network, contact your local Cisco account team for help in understanding and meeting the CAMA regulatory requirements.

Note Voice VRF is not supported for SCCP SRST on Cisco Integrated Services Router Generation 2 (ISR G2).

Supported Devices, Platform and Components

Supported Devices, Router Platforms and Memory Specifications

Refer to *Unified SRST/E-SRST Supported Firmware, Platforms, Memory, and Voice Products* for information on:

- Supported Cisco IP Phones
- Supported router platforms
- Maximum number of IP phones, directory numbers or virtual voice ports per router
- Memory specifications per router

For support information for your release, see Compatibility Information for Unified SRST/E-SRST 14.3 Supported Firmware, Platforms, Memory, and Voice Products.

Supported Cisco IOS Releases

For a list of Cisco IOS releases that support SRST, see Cisco Unified CME and Cisco IOS Software Version Compatibility Matrix.

Note Platform support for particular Cisco IOS software releases is dependent on the availability of the software images for those platforms. Software images for some platforms may be deferred, delayed, or changed without prior notice. For updated information about platform support and availability of software images for each Cisco IOS software release, see the online release notes or, if supported, Cisco Feature Navigator.

Cisco Feature Navigator

Cisco IOS software is packaged in feature sets that are supported on specific platforms. To get updated information regarding platform support for this feature, access Cisco Feature Navigator. Cisco Feature Navigator dynamically updates the list of supported platforms as new platform support is added for the feature.

Cisco Unified Communications Manager

For compatibility information for Cisco Unified Communications Manager, see Compatibility Matrix for Cisco Unified Communications Manager.

Language Support

For information on supported languages and locale files, see Cisco Unified Communications Manager Express Localization Matrix.

Interface Support with Cisco Unified Communications Manager Express and SRST

Cisco Unified Communications Manager Express and Cisco Unified SRST routers have multiple interfaces and is used for signaling and data packet transfers. The two types of interfaces available on a Cisco router include the physical interface and the virtual interface. The types of physical interfaces available on a router depend on its interface processors or port adapters. Virtual interfaces are software-based interfaces that you create in the memory of the networking device using Cisco IOS commands. To configure a virtual interface for connectivity, use the Loopback Interface for Cisco Unified Communications Manager Express and Cisco Unified SRST.

Cisco Unified Communications Manager Express and Cisco Unified SRST supports the following interfaces:

- Gigabit Ethernet Interface (IEEE 802.3z) (**interface gigabitethernet**)
- Loopback Interface (interface loopback)
- Fast Ethernet Interface (interface fastethernet)

Signal Support

Cisco Unified SRST supports FXS, FXO, T1, E1, and E1 R2 signals.

Switch Support

Cisco SRST 3.2 and later versions support all PRI and BRI switches including the following:

- basic-1tr6
- basic-5ess
- basic-dms100

- basic-net3
- basic-ni
- basic-ntt NTT switch type for Japan
- basic-ts013
- primary-4ess Lucent 4ESS switch type for the United States
- primary-5ess Lucent 5ESS switch type for the United States
- primary-dms100 Northern Telecom DMS-100 switch type for the United States
- primary-net5 NET5 switch type for the United Kingdom, Europe, Asia, and Australia
- primary-ni National ISDN switch type for the United States
- primary-ntt NTT switch type for Japan
- primary-qsig QSIG switch type
- primary-ts014 TS014 switch type for Australia (obsolete)

Where to Go Next

The next chapters of this book describe how to configure Cisco Unified SIP SRST. As shown in the following table, each chapter takes you through tasks in the order in which they need to be performed. The first task for configuring Cisco Unified SRST is to ensure that the basic software and hardware in your system are configured correctly for Cisco Unified SRST.

Task	Where Task Is Described
7. Setting up a Cisco Unified SRST system to communicate with your network	Setting Up the Network
8. Configuring Version 4.1 features	Cisco Unified SIP SRST 4.1
9. Setting up the basic Cisco Unified SRST phone configuration using SCCP	Setting Up Cisco Unified IP Phones using SCCP
10. Providing a backup to an external SIP call control (IP-PBX) by supplying basic registrar services	Setting Up Cisco Unified IP Phones using SIP
11. Configuring incoming and outgoing calls	Configuring Call Handling
12. Configuring optional security for SRST	Configuring Secure SRST for SCCP and SIP
13. Setting up voicemail	Integrating Voicemail with Cisco Unified SRST
14. Setting up video parameters	Setting Video Parameters
15. Monitoring and maintaining Cisco Unified Survivable Remote Site Telephony (SRST)	Monitoring and Maintaining Cisco Unified SRST

Related Documents and References

Related Documents

Related Topic	Documents
Cisco IOS voice product configuration	• Cisco IOS Voice Configuration Library • Cisco IOS Voice Command Reference • Cisco IOS Debug Command Reference • Cisco IOS Tcl IVR and VoiceXML Application Guide • Cisco IOS Survivable Remote Site Telephony Version 3.2 System Administrator Guide
Configuring SRST and MGCP Fallback	• Configuring MGCP Gateway Support for Cisco Unified Communications Manager • MGCP Gateway Fallback Transition to Default H.323 Session Application • Configuring SRS Telephony and MGCP Fallback
Cisco Unified Communications Manager user documentation	• Cisco Unified Communications Manager • Cisco Unified Communications Manager Security Guide • Cisco Unified Communications Operating System Administration Guide
Cisco Unified IP Phones	• Cisco 7900 Series Unified IP Phones End-User Guides • Cisco IP Phone Authentication and Encryption for Cisco Communications Manager • Cisco Unified IP Phone 7970 Series Administration Guide for Cisco Unified CallManager, Release 5.0 (for models 7970G and 7971G-GE) (SCCP), “Understanding Security Features for Cisco IP Phones” section.
Cisco Unified SRST commands and specifications	• Cisco Unified SRST and Cisco Unified SIP SRST Command Reference (All Versions) • Cisco Unified SRST 8.0 Supported Firmware, Platforms, Memory, and Voice Products • Cisco Unified SRST 4.3 Supported Firmware, Platforms, Memory, and Voice Products

Related Topic	Documents
Cisco Security Documentation	• Media and Signaling Authentication and Encryption Feature for Cisco IOS MGCP Gateways • Cisco IOS Certificate Server • Manual Certificate Enrollment (TFTP and Cut-and-Paste) • Certification Authority Interoperability Commands • Certificate Enrollment Enhancements
Cisco SIP SRST V3.4: Cisco IOS SIP Survivable Remote Site Telephony Feature Roadmap	• Cisco IOS SIP SRST Feature Roadmap
Cisco SIP functionality	• Cisco IOS SIP Configuration Guide
Cisco SRST command reference	• Cisco IOS Survivable Remote Site Telephony Version 3.2 Command Reference
Command reference information for voice and telephony commands	• Cisco IOS Voice Command Reference • Cisco IOS Debug Command Reference
DHCP	• Cisco IOS DHCP Server
Media Inactive Call Detection	• Media Inactive Call Detection
Phone documentation for Cisco Unified SRST	• Cisco Unified IP Phones 7900 Series • Survivable Remote Site Telephony
Standard Glossary	• Cisco IOS Voice Configuration Library Glossary
Standard Preface	• Cisco IOS Voice Configuration Library Preface

Standards

Standard	Title
ITU X. 509 Version 3	Public-Key and Attribute Certificate Frameworks

MIBs

MIB	MIBs Link
No new or modified MIBs are supported by this feature, and support for existing MIBs has not been modified by this feature.	To locate and download MIBs for selected platforms, Cisco IOS releases, and feature sets, use Cisco MIB Locator found at the following URL: http://www.cisco.com/go/mibs

RFCs

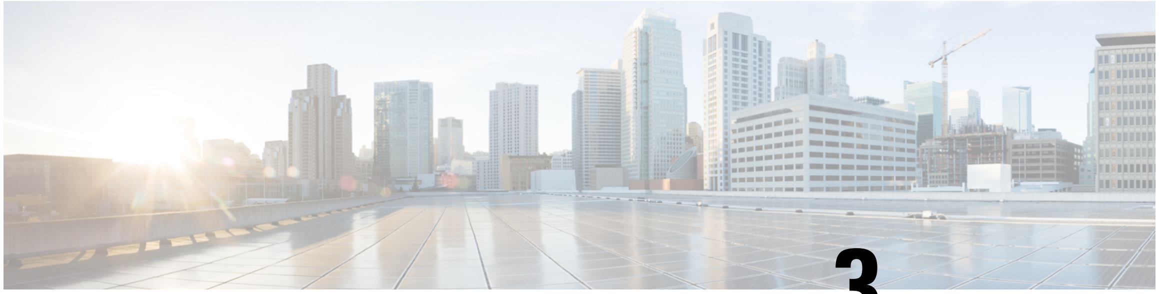
RFC	Title
RFC2246	The Transport Layer Security (TLS) Protocol Version 1.0
RFC 2543	SIP: Session Initiation Protocol
RFC 3261	SIP: Session Initiation Protocol
RFC3711	The Secure Real-Time Transport Protocol (SRTP)

Technical Assistance

Description	Link
The Cisco Technical Support & Documentation website contains thousands of pages of searchable technical content, including links to products, technologies, solutions, technical tips, and tools. Registered Cisco.com users can log in from this page to access even more content.	http://www.cisco.com/techsupport

Obtaining Documentation, Obtaining Support, and Security Guidelines

For information on obtaining documentation, obtaining support, providing documentation feedback, security guidelines, and also recommended aliases and general Cisco documents, see the monthly *What's New in Cisco Product Documentation*, which also lists all new and revised Cisco technical documentation, at http://www.cisco.com/en/US/docs/general/whatsnew/whatsnew.html.



CHAPTER 3

Cisco Unified SIP SRST on Cisco 4000 Series Integrated Services Router

This chapter describes the support for Unified SIP SRST on the Cisco 4000 Series Integrated Services platform.

Note Unified SRST 12.6 on Cisco IOS XE Gibraltar 16.11.1a Release is not a recommended release version for call flows that include Multicast Music On Hold.

Overview

This chapter describes Unified SRST functionality on Cisco 4000 Series Integrated Services Routers for SIP phones. Unified SIP SRST provides backup to Unified Communications Manager when the IP connectivity to Unified Communications Manager is down.

Cisco Unified SIP SRST supports the following during a WAN outage:

- Basic Registration of SIP phones.
- Basic call support on SIP phones.
- Basic supplementary services such as Call Transfer, MOH, and Conference
- SIP phone to SIP phone
- SIP phone to PSTN / router voice-port
- SIP phone to Skinny Client Control Protocol (SCCP) phone
- SIP phone to WAN VoIP using SIP

Platform and Memory Support

From Unified SRST Release 10.0 (Cisco IOS XE Release 3.10S), Unified SIP SRST is supported on the Cisco 4000 Series Integrated Services platform. As part of the Cisco IOS XE Release 3.10S Release, support was introduced on the Cisco 4451-X Integrated Services Router. From Unified SRST Release 10.5 (Cisco IOS XE Release 3.13S), SIP SRST is supported on all Cisco 4000 Series Integrated Services Routers.

The following Cisco 4000 Series Integrated Services Router platforms are supported:

- Cisco ISR 4321 Integrated Services Routers
- Cisco ISR 4331 Integrated Services Routers
- Cisco ISR 4351 Integrated Services Routers
- Cisco ISR 4431 Integrated Services Routers
- Cisco ISR 4451 Integrated Services Routers

For more information on Platform and Memory Support, see Compatibility Information.

Cisco IOS Software Releases that Support Unified SRST

For information on the Unified SRST Release and the corresponding IOS Software, see Unified CME, Unified SRST, and Cisco IOS Software Version Compatibility Matrix for related compatibility information.

To access Cisco Feature Navigator, go to http://www.cisco.com/go/cfn. An account on Cisco.com is not required.

Install Cisco IOS XE Software

To verify that the recommended software is installed on the Cisco router and if necessary, download and install a Cisco IOS XE image, perform the following steps.

Before you begin

The Cisco router is installed including sufficient memory, all Cisco voice services hardware, and other optional hardware.

SUMMARY STEPS

1. Identify which Cisco IOS XE software release is installed on router. Log in to the router and use the **show version EXEC** command.
2. Compare the Cisco IOS XE release installed on the Cisco router to the information in the Cisco Unified CME, Unified SRST, and Cisco IOS Software Version Compatibility Matrix to determine whether the Cisco IOS release supports the recommended Unified SRST.
3. If necessary, download and extract the recommended Cisco IOS XE image to flash memory in the router.
4. To reload the Unified SRST router with the new software after replacing or upgrading the Cisco IOS XE release, use the **reload** privileged EXEC command.

DETAILED STEPS

	Command or Action	Purpose

Step 1	Identify which Cisco IOS XE software release is installed on router. Log in to the router and use the **show version EXEC** command.
	Example:
	`Router> show version`
	`Cisco IOS XE Software, Version`
	`BLD_POLARIS_DEV_LATEST_20200621_053200`
	`Cisco IOS Software [Amsterdam], ISR Software`
	`(X86_64_LINUX_IOSD-UNIVERSALK9-M),`
	`Version 17.3.1`
	`[S2C-build-polaris_dev-116144-/nobackup/mcpre/BLD-BLD_POLARIS_DEV_LATEST_20200621_0532`
	`00 259]`
	`Copyright (c) 1986-2020 by Cisco Systems, Inc.`
	`Compiled Sun 21-Jun-20 07:03 by mcpre`	

Step 2	Compare the Cisco IOS XE release installed on the Cisco router to the information in the Cisco Unified CME, Unified SRST, and Cisco IOS Software Version Compatibility Matrix to determine whether the Cisco IOS release supports the recommended Unified SRST.	
Step 3	If necessary, download and extract the recommended Cisco IOS XE image to flash memory in the router.	To find software installation information, access information located at www.cisco.com > Support > Products & Downloads > Networking Software > {Choose release} > Configuration Guides / System Management / Configuration fundamentals.

Step 4	To reload the Unified SRST router with the new software after replacing or upgrading the Cisco IOS XE release, use the **reload** privileged EXEC command.
	Example:
	`Router# reload`
	`System configuration has been modified. Save?`
	`[yes/no]: yes`	

	Command or Action	Purpose
	Building configuration... [OK] Proceed with reload? [confirm] Jun 24 00:45:13.827: %PMAN-5-EXITACTION: R0/0: pvp: Process manager is exiting: process exit with reload chassis code Initializing Hardware ... Checking for PCIe device presence...done System integrity status: 0x610 Rom image verified correctly System Bootstrap, Version 16.12(2r), RELEASE SOFTWARE Copyright (c) 1994-2019 by cisco Systems, Inc. Current image running: Boot ROM0 Last reset cause: LocalSoft ISR4331/K9 platform with 4194304 Kbytes of main memory .... Located isr4300-universalk9.BLD_POLARIS_DEV_LATEST_20200621_053200.SSA.bin Router>	

Feature Support

The following features are supported for Unified SIP SRST on Cisco 4000 Series Integrated Services Platform:

- Auto-answer (If enabled on Unified Communications Manager)
- Alert/Semi-Consult/Attended/Consult Transfer
- Ad-hoc Software Conference
- Hold or Resume
- Headset Answer
- Caller ID Display
- Call Forward to Voice Hunt Group
- Call Transfer to a Voice Hunt Group
- Voicemail
- Message Waiting Indicator (MWI)
- Do Not Disturb (DND)
- DTMF
- Feature Button or Programmable Line Key (PLK) - If enabled on Unified Communications Manager
- Key Expansion Module (KEM - Supported only on the 8851/8851NR/8861 phones)
- Bulk Registration Support
- Enabling or Disabling KPML

- Alias Feature
- Call Forward (All, Busy, No Answer, Mailbox)
- Call Forward All Softkey on Phone
- Unicast MOH
- Audio codecs (G.722, G.711, G.729, iLBC)
- Translation Profile
- Conference Blocking
- Transfer Blocking
- COR
- Voice Class Codec
- SNMP/MIB (Supported only to get mode and number of registered phones)
- Speed Dial (If enabled on Unified Communications Manager)
- Call Waiting (If enabled on Unified Communications Manager)
- Forced Authorization Code
- Redial
- Speakerphone (Dialing, Answering)
- System Message
- After Hours
- SSH to Phone
- Span to PC (except Cisco IP Phone 8831)
- Web Access to Phone
- Voice Hunt Group (Support for Parallel, Sequential, Peer, and Longest-idle hunt groups). Basic features such as Call, Hold or Resume are only supported.)

Restrictions of Unified SRST on Cisco 4000 Series Integrated Services Routers

- Multicast MOH for SIP is not supported on the Cisco 4000 Series Integrated Services Routers.
- Transcoding is not supported on the Unified SRST.
- Voice VRF is not supported for SCCP SRST on Cisco Integrated Services Router Generation 2 (ISR G2).
- Shared lines and Mixed shared lines are not supported on the Unified SRST (supported on the Unified E-SRST).
- Privacy (on hold) is not supported on the Unified SRST (supported on the Unified E-SRST).
- SNMP/MIB support is restricted to fetching information on mode and number of registered phones.

- The CLI command **max-redirect** is not supported for SIP on Unified SRST.
- Unified SRST supports only the basic voice hunt group features. To configure advanced voice hunt group features, you must deploy the Cisco Unified Enhanced Survivable Remote Site Telephony.
- Video Calling is not supported on Unified SIP SRST.

Unified IP Phone Support

Unified SIP SRST on Cisco 4000 Series Integrated Services Platform is supported on all the SIP phones, including Cisco IP Phone 7800 Series and Cisco IP Phone 8800 Series.

Cisco Jabber with Unified SRST

Unified SRST 12.8 (Cisco IOS XE Amsterdam 17.2.1r) and later releases support the following Cisco Jabber clients:

- Cisco Jabber for Windows,12.9
- Cisco Jabber for Mac, 12.9

Cisco Unified Communications Manager Compatibility

For more information on Unified Communications Manager compatibility, see Cisco Unified Communications Manager Compatibility Matrix.

Installing Cisco Unified Communications Manager

When installing Cisco Unified Communications Manager, consider the following:

- See the installation instructions for your version in the Cisco Unified Communications Manager Install and Upgrade Guides.
- Integrate Cisco Unified SRST with Cisco Unified Communications Manager. Integration is performed from Cisco Unified Communications Manager. See the Integrating Cisco Unified SIP SRST with Cisco Unified Communications Manager section.

Integrating Cisco Unified SIP SRST with Cisco Unified Communications Manager

The procedure for integrating Unified SRST with Cisco Unified Communications Manager is as follows:

For Cisco Communications Manager integration with Unified SIP SRST, you must create an SRST reference and apply it to a device pool. An SRST reference is the IP address of the Cisco Unified SRST Router.

SUMMARY STEPS

1. Create an SRST reference.

2. Apply the SRST reference or the default gateway to one or more device pools.

DETAILED STEPS

	Command or Action	Purpose
Step 1	Create an SRST reference.	
Step 2	Apply the SRST reference or the default gateway to one or more device pools.	

Supported PSTN Trunk Connectivity

Unified SRST is supported with SIP trunks. Also, Unified SIP SRST supports the following trunk types:

- FXO/FXS
- Basic Rate ISDN
- Primary Rate ISDN (T1 or E1)

Language Support

For information on language support, see Localization Matrix.

Switch Support

Unified SRST supports all PRI and BRI switches including the following:

- basic-1tr6
- basic-5ess
- basic-dms100
- basic-net3
- basic-ni
- basic-ntt NTT switch type for Japan
- basic-ts013
- primary-4ess Lucent 4ESS switch type for the United States
- primary-5ess Lucent 5ESS switch type for the United States
- primary-dms100 Northern Telecom DMS-100 switch type for the United States
- primary-net5 NET5 switch type for the United Kingdom, Europe, Asia, and Australia
- primary-ni National ISDN switch type for the United States
- primary-ntt NTT switch type for Japan
- primary-qsig QSIG switch type

primary-ts014 TS014 switch type for Australia (obsolete)

Interface Support for Unified SRST

Unified SRST routers have multiple interfaces that are used for signaling and data packet transfers. The two types of interfaces available on a Cisco router include the physical interface and the virtual interface. The type of physical interfaces available on a router depends on its interface processors or port adapters. Virtual interfaces are software-based interfaces that you create in the memory of the networking device using Cisco IOS commands. To configure a virtual interface for connectivity, you can use the Loopback Interface for Unified SRST.

The following interfaces are supported on Unified SRST:

- Gigabit Ethernet Interface (IEEE 802.3z) (**interface gigabitethernet**)
- Loopback Interface (**interface loopback**)
- Fast Ethernet Interface (**interface fastethernet**)

Simple Network Management Protocol (SNMP) Support for Unified SRST

Unified SRST supports Simple Network Management Protocol (SNMP) Management Information Base (MIBs) for monitoring the product status. Unified SRST Release 12.6 and later versions is SNMP Version 3 (SNMPv3) compliant. The following is the main SNMP MIB supported by Unified SRST:

- CISCO-SRST-MIB

For information on configuration of SNMP version 3 on Unified SRST router, see SNMP Configuration Guide.

Licensing

This section provides information on licensing of Cisco Unified Survivable Remote Site Telephony (Unified SRST).

Cisco Smart Licensing for Unified SRST

Cisco Smart Licensing is a software licensing model that provides visibility of ownership and usage through the Cisco Smart Software Manager (CSSM) portal. CSSM is a central license repository that manages licenses across all Cisco products that you own, including Unified SRST. Devices send license usage to CSSM either directly or use an on-premises satellite. Your Smart Account Administrator controls your access to CSSM. Use your Cisco credentials to access the CSSM portal using http://software.cisco.com.

Smart Licensing applies to all platform technology (UCK9, Security) and Unified SRST feature licenses that the router uses. Unified SRST requires one license entitlement (SRST_EP) for each configured SIP or SCCP phone.

CSSM shows license usage across all devices that are registered to a virtual account. A Virtual Account License Inventory displays the quantity of licenses that are purchased, those licenses in use, and a balance. An **Insufficient Licenses** alert is displayed if the license balance is below 0.

For example, consider a smart account in CSSM with 50 SRST_EP licenses. If you have a single registered Unified SRST router with 20 phones configured, the CSSM licenses page shows **Purchased** as 50, **In Use** as 20 and **Balance** as 30.

For more information on Smart Software Manager, see the Cisco Smart Software Manager User Guide.

Note The SRST_EP license count reflects the total phone count for both the ephones and voice register pools that are configured in the Unified SRST irrespective of whether the phones are registered or not. To avoid unnecessary reporting while Unified SRST is being configured, license usage is reported three minutes after the last configuration change.

Note Unified SRST Smart Licenses also provide RTU entitlement for routers that are not configured for Smart Licensing.

Smart License Operation

Cisco IOS XE Everest 16.5.1 Release to Cisco IOS XE Fuji 16.9.1 Release

Cisco 4000 Series Integrated Services Routers support Smart Licensing as an alternative to Cisco Software RTU Licensing. Use the **license smart enable** command to enable Smart Licensing. To disable Smart Licensing, use the **no** form of the command and re-accept the EULA using the **license accept end user agreement** command.

Cisco IOS XE Gibraltar 16.10.1 Release Onwards

The Cisco RTU Licensing and the CLI **license smart enable** command are deprecated. Smart Licensing is mandatory from this release.

Cisco IOS XE Everest 16.5.1 Release to Cisco IOS XE Amsterdam 17.3.1a Release

Routers configured to use Smart Licensing offer a 90-day evaluation period, during which you can use all the features without registering to CSSM. A Unified SRST device is associated with CSSM using a registration token. You can obtain the registration token from the virtual CSSM account or from an on-premises satellite. Once registered, the evaluation period pauses and you can use the balance license later. You cannot renew the evaluation period on its expiry.

Warning Unified SRST shuts down when the router is unregistered and allowed to pass in to the Evaluation Expired state.

To register the Unified SRST router with CSSM, use **license smart register idtoken** command. For information on registering the device with CSSM, see Software Activation Configuration Guide.

Upon successful registration, the device sends an authorization request to CSSM for the licenses in use. For each license type requested, if the Smart Account has sufficient licenses, CSSM responds with **Authorized** . If the Smart Account does not have sufficient licenses, CSSM responds with **Out of Compliance** .

Post successful authorization of the request, licenses are bound to the requesting device until the next authorization request submission. An authorization request is sent every 30 days or when there is any change in license consumption, to maintain the registration with CSSM. The authorization expires if you do not update the license request for the router within 90 days. The certificate issued to identify the router at the time of registration is valid for one year and renewed every six months. The router displays the License authorization as follows:

```
Router# show license summary
Smart Licensing is ENABLED
Registration:
Status: REGISTERED
Smart Account: ABC
Virtual Account: XYZ
Export-Controlled Functionality: Not Allowed
Last Renewal Attempt: None
Next Renewal Attempt: Jun 07 12:08:10 2017 UTC
License Authorization:
Status: AUTHORIZED
Last Communication Attempt: SUCCESS
Next Communication Attempt: Apr 13 07:11:48 2017 UTC
License Usage:
License                  Entitlement tag            Count Status
-----------------------------------------------------------------------------
ISR_4351_UnifiedCommun.. (ISR_4351_UnifiedCommun..) 1     AUTHORIZED
SRST v12 Endpoint Li...  (SRST_EP)                  4     AUTHORIZED
```

Cisco IOS XE Gibraltar 16.12.1 Release to Cisco IOS XE Amsterdam 17.3.1a Release

Specific License Reservation (SLR) is supported on Cisco 4000 Series Integrated Services Routers. SLR allows reservation and utilization of Cisco Smart Licenses without communicating the license information to CSSM. To reserve specific licenses for a device, generate request code from the device. Enter the request code in CSSM along with the required licenses and their quantity, and generate authorization code. Enter the authorization code on the device to map the license to the Unique Device identifier (UDI).

Note If upgrading to IOS XE Amsterdam 17.3.1a with a license reservation in place, update the reservation to include version 14, rather than version 12 SRST licenses. The reservation may be updated before or after the software upgrade.

Cisco IOS XE Amsterdam 17.3.2 and Cisco IOS XE Bengaluru 17.4.1a Release Onwards

This release introduces a new paradigm for tracking license usage across your business. In earlier releases, license authorization was forward looking, binding licenses to a device until the next authorization request. Actual license usage during the proceeding reporting period is now sent to CSSM, allowing you to plan ongoing license requirements based on historical usage data. Initial device registration is no longer required to use most platform functionality and the evaluation period is deprecated.

License usage reports are submitted periodically according to a minimum reporting policy set for your account. Typically, this period could be once per year. However, you can generate reports more frequently if the use of licensed features varies over time. CSSM acknowledges each Resource Utilization Monitoring (RUM) report to ensure that the usage is recorded reliably. If the router does not receive an acknowledgment within the minimum reporting period, call processing is disabled. Call processing is resumed when a valid acknowledgment is received.

Reports can be submitted to CSSM directly or through a satellite. Cisco Smart Licensing Utility (CSLU) applications can also receive usage reports, providing you with more flexibility in managing your license usage. Also, when a device is not able to communicate directly with a licensing server, a signed usage report can be generated and manually uploaded to CSSM. The acknowledgment that is generated by CSSM must be uploaded to the device within the license reporting policy period to ensure continued use.

As license reporting is now based on historical usage, the registration process that is used previously has been replaced with a trust association that also defines the reporting policy set in your account. Establishing trust with CSSM or Cisco Smart Software Manager Satellite uses an identity token similar to earlier registrations. Use the **license smart trust idtoken** *token* command to establish the trust relationship within the initial reporting period set for the device. The CLI **license smart register** command is deprecated from this release.

Current license usage for Cisco Unified SRST is displayed using the **show license summary** command:

Warning When using any of the following releases, Unified SRST shuts down if the router does not receive a report acknowledgment from CSSM before the acknowledgment deadline set by the account policy: 17.3.2, 17.3.3, 17.3.4a, 17.6.1a, or any 17.4 or 17.5 release. Unified SRST does not shut down in this way with later releases.

Note Smart License Reservation (SLR) for SRST licenses is not compatible with IOS XE Amsterdam 17.3.2 and later releases. Even if a reservation is in place when upgrading to one of these releases, license use reporting will still be required in accordance with the device policy.

```
Router#show license summary
License Usage:
License                  Entitlement tag               Count   Status
-----------------------------------------------------------------------------
appxk9...................(ISR_4400_Application) ......1...... IN USE
uck9.................... (ISR_4400_UnifiedCommun..)...1.......IN USE
securityk9.............. (ISR_4400_Security)......... 1.......IN USE
SRST_E_EP............... (SRST_E_EP)..................2.......IN USE
SRST_EP..................(SRST_EP)...................18...... IN USE
```

Configure SIP Registrar Functionality for SIP Phones on Unified SRST

Session Initiation Protocol (SIP) registrar functionality in Cisco IOS software is an essential part of Cisco Unified SIP Survivable Remote Site Telephony (SRST). According to RFC 3261, a SIP registrar is a server that accepts Register requests.

Unified SIP SRST provides backup to Cisco Unified Communications Manager. The registrar functionality is configured on the Unified SRST gateway so as to assist fallback of endpoints to Unified SRST from Unified Communications Manager.

These services are used by a SIP IP phone if there is a WAN connection outage, and the SIP phone is unable to communicate with its primary SIP call control (IP-PBX). The Unified SIP SRST device also provides PSTN gateway access for placing and receiving PSTN calls.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **allow-connections sip to sip**
5. **sip**
6. **registrar server** [**expires** [**max** *sec*] [**min** *sec*]]
7. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice service voip** **Example:** `Router(config)# voice service voip`	Enters voice service configuration mode.
Step 4	**allow-connections sip to sip** **Example:** `Router(config-voi-srv)# allow-connections sip to sip`	Allows connections from SIP to SIP endpoints.
Step 5	**sip** **Example:** `Router(config-voi-srv)# sip`	Enters SIP configuration mode.
Step 6	**registrar server** [**expires** [**max** *sec*] [**min** *sec*]] **Example:** `Router(config)# call-manager-fallback`	Enables SIP registrar functionality. The keywords and arguments are defined as follows: • **expires** : (Optional) Sets the active time for an incoming registration. • **max** *sec* : (Optional) Maximum expiration time for a registration, in seconds. The range is from 600 to 86400. The default is 3600. **Note** Ensure that the registration expiration timeout is set to a value smaller than the TCP connection aging timeout to avoid disconnection from the TCP.

	Command or Action	Purpose
		• **min** *sec* : (Optional) Minimum expiration time for a registration, in seconds. The range is from 60 to 3600. The default is 60.

Step 7	**end**
	Example:
	`Router(conf-serv-sip)# end`
		Returns to privileged EXEC mode.

Configure Backup Registrar Service to SIP Phones

Backup registrar service to SIP IP phones can be provided by configuring a voice register pool on SIP gateways. The voice register pool configuration provides registration permission control and can be used to configure some dial-peer attributes that are applied to the dynamically created VoIP dial peers when SIP phone registrations match the pool. The following call types are supported:

- SIP IP phone to or from:
- Local PSTN
- Local analog FXS phones
- Local SIP IP phone

The commands in the configuration provide registration permission control and set up a basic voice register pool. The pool gives users control over which registrations are accepted by a Cisco Unified SIP SRST device and which can be rejected. Registrations that match this pool create VoIP SIP dial peers with the dial-peer attributes set to these configurations. Although only the **id** command is mandatory, this configuration example shows basic functionality.

Restrictions

- The **id** command identifies the individual SIP IP phone or sets of SIP IP phones that are to be configured. Thus, the**id**command configured in Step 5 is required and must be configured before any other voice register pool commands. For Unified SRST, It is recommended to configure **id ip/nework/device-id-name** and avoid using **id mac**.

Note To monitor SIP proxies, the **call fallback active** command must be configured, as described in Step 3.

Note The command **proxy** described in Step 7 is an optional configuration.

Note It is recommended that **id mac** command is not configured for Unified SRST, as the phones falling back from Unified Communications Manager to Unified SRST do not mostly fall back on the same network.

Before you begin

The SIP registrar must be configured before a voice register pool is set up.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **call fallback active**
4. **voice register pool** *tag*
5. **id** [{**network** *address* **mask** *mask* |**ip** *address* **mask** *mask* |**mac** *address* }] [**device-id-name** *devicename*]
6. **preference** *preference-order*
7. **proxy** *ip-address* [**preference** *value*] [**monitor probe** {**icmp-ping** | **rtr**} [*alternate-ip-address*]]
8. **voice-class codec** *tag*
9. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**call fallback active** **Example:** `Router(config)# call fallback active`	(Optional) Enables a call request to fall back to alternate dial peers if there is network congestion. • This command is used if you want to monitor the proxy dial peer and fallback to the next preferred dial peer. For full information on the **call fallback active** command, see PSTN Fallback Feature.
Step 4	**voice register pool** *tag* **Example:** `Router(config)# voice register pool 12`	Enters voice register pool configuration mode for SIP phones. Use this command to control which registrations are accepted or rejected by a Cisco Unified SIP SRST device.
Step 5	**id** [{**network** *address* **mask** *mask* |**ip** *address* **mask** *mask* |**mac** *address* }] [**device-id-name** *devicename*] **Example:** `Router(config-register-pool)# id network 172.16.0.0 mask 255.255.0.0`	Explicitly identifies a locally available individual or set of SIP IP phones. The keywords and arguments are defined as follows: • **network** *address* **mask** *mask* : The **network** *address* mask *mask* keyword/argument combination is used to accept SIP Register messages for the indicated phone numbers from any IP phone within the indicated IP subnet.

	Command or Action	Purpose
		• **ip** *address* **mask** *mask*: The **ip** *address* mask *mask* keyword/argument combination is used to identify an individual phone. • **mac** *address*: MAC address of a particular Cisco Unified IP Phone. • **device-id-name** *devicename*: Defines the device name to be used to download the phone's configuration file.
Step 6	**preference** *preference-order* **Example:** `Router(config-register-pool)# preference 2`	Sets the preference order for the VoIP dial peers to be created. Range is from 0 to 10. Default is 0, which is the highest preference. The preference must be greater (lower priority) than the preference configured with the **preference** keyword in the **proxy** command.
Step 7	**proxy** *ip-address* [**preference** *value*] [**monitor probe** {**icmp-ping** | **rtr**} [*alternate-ip-address*]] **Example:** `Router(config-register-pool)# proxy 10.2.161.187 preference 1`	(Optional) Autogenerates additional VoIP dial peers to reach the main SIP proxy whenever a Cisco Unified SIP IP Phone registers with a Cisco Unified SIP SRST gateway. The keywords and arguments are defined as follows: • *ip-address* : IP address of the SIP proxy. • **preference** *value* : (Optional) Defines the preference of the proxy dial peers that are created. The preference must be less (higher priority) than the preference configured with the **preference** *value* command. Range is from 0 to 10. The highest preference is 0. There is no default. • **monitor probe** : (Optional) Enables monitoring of proxy dial peers. • **icmp-ping**: Enables monitoring of proxy dial peers using ICMP ping. **Note** The dial peer on which the probe is configured will be excluded from call routing only for outbound calls. Inbound calls can arrive through this dial peer. • **rtr**: Enables monitoring of proxy dial peers using RTR probes. • *alternate-ip-address* : (Optional) Enables monitoring of alternate IP addresses other than the proxy address. For example, to monitor a gateway front end to a SIP proxy.

	Command or Action	Purpose
Step 8	**voice-class codec** *tag* **Example:** `Router(config-register-pool)# voice-class codec 15`	Sets the voice class codec parameters. The *tag* argument is a codec group number between 1 and 10000.

Step 9	**end**
	Example:
	`Router(config-register-pool)# end`
		Returns to privileged EXEC mode.

Configure Backup Registrar Service to SIP Phones (Using Optional Commands)

The prior configurations set up a basic voice register pool. The configuration in this procedure adds optional attributes to increase functionality. As part of this configuration, you can support:

- Translation Profile—Applies the translation profile to a specific directory number or to all directory numbers on a SIP phone.
- Alias—Allows Cisco Unified SIP IP Phones to handle inbound PSTN calls to phone numbers that are unavailable when the main SIP call control (IP-PBX) is not available.
- Class of restriction (COR)—COR specifies which incoming dial peers can use which outgoing dial peers to make a call. Each dial peer can be provisioned with an incoming and outgoing COR list.

Before you begin

Before configuring the **alias** command, translation rules must be set using the translation-profile outgoing (**voice register pool**) command.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register pool***tag*
4. **translation-profile outgoing** *profile-tag*
5. **alias** *tag pattern* **to** *target* [**preference** *value*]
6. **cor** {**incoming** | **outgoing**} *cor-list-name* {*cor-list-number starting-number* [- *ending-number*] | **default** }
7. **incoming called-number** *[number]*
8. **number** *tag number-pattern* {**preference***value*} [**huntstop**]
9. **dtmf-relay** [**cisco-rtp**] [**rtp-nte**] [**sip-notify**]
10. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable**	Enables privileged EXEC mode.

	Command or Action	Purpose
	Example: `Router> enable`	• Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice register pool***tag* **Example:** `Router(config)# voice register pool 12`	Enters voice register pool configuration mode. Use this command to control which registrations are accepted or rejected by a Cisco Unified SIP SRST device.

Step 4	**translation-profile outgoing** *profile-tag*
	Example:
	`Router(config-register-pool)#`
	`voice translation-rule 1`
	`rule 1 /1000/ /1006/`
	`!`
	`!`
	`voice translation-profile 1`
	`translate called 1`
	`!`
	`voice register pool xxx`
	`translation-profile outgoing 1`
		Use this command to apply the translation profile to a specific directory number or to all directory numbers on a SIP phone.
		• *Profile-tag* : Translation profile name to handle translation to outgoing calls.

Step 5	**alias** *tag pattern* **to** *target* [**preference** *value*] **Example:** `Router(config-register-pool)# alias 1 94... to 91011 preference 8`	Allows Cisco Unified SIP IP Phones to handle inbound PSTN calls to phone numbers that are unavailable when the main proxy is not available. The keywords and arguments are defined as follows: • *tag* : Number from 1 to 5 and the distinguishing factor when there are multiple alias commands. • *pattern*: The prefix number; matches the incoming phone number and may include wildcards. • **to** : Connects the tag number pattern to the alternate number. • *target*: The target number; an alternate phone number to route incoming calls to match the number pattern. • **preference** *value* : (Optional) Assigns a dial-peer preference value to the alias. The *value* argument is the value of the associated dial peer, and the range is from 1 to 10. There is no default.

Step 6	**cor** {**incoming** | **outgoing**} *cor-list-name* {*cor-list-number starting-number* [- *ending-number*] | **default** }
	Example:
	`Router(config-register-pool)# cor incoming call91 1 91011`
		Configures a class of restriction (COR) on the VoIP dial peers associated with directory numbers. COR specifies which incoming dial peers can use which outgoing dial peers to make a call. Each dial peer can be provisioned with an incoming and outgoing COR list. The keywords and arguments are defined as follows:

	Command or Action	Purpose
		• **incoming** : COR list to be used by incoming dial peers. • **outgoing** : COR list to be used by outgoing dial peers. • *cor-list-name*: COR list name. • *cor-list-number*: COR list identifier. The maximum number of COR lists that can be created is four, comprised of incoming or outgoing dial peers. • *starting-n*umber: Start of a directory number range, if an ending number is included. Can also be a standalone number. • (Optional) Indicator that a full range is configured. • *ending-number*: (Optional) End of a directory number range. • **default** : Instructs the router to use an existing default COR list.
Step 7	**incoming called-number** *[number]* **Example:** `Router(config-register-pool)# incoming called-number 308`	Applies incoming called parameters to dynamically created dial peers. The number argument is optional and indicates a sequence of digits that represent a phone number prefix.
Step 8	**number** *tag number-pattern* {**preference***value*} [**huntstop**] **Example:** `Router(config-register-pool)# number 1 50.. preference 2`	Indicates the E.164 phone numbers that the registrar permits to handle the Register message from the Cisco Unified SIP IP Phone. The keywords and arguments are defined as follows: • *tag* : Number from 1 to 10 and the distinguishing factor when there are multiple number commands. • *number-pattern*: Phone numbers (including wildcards and patterns) that are permitted by the registrar to handle the Register message from the SIP IP phone. • **preference** *value* : (Optional) Defines the number list preference order. • **huntstop**: (Optional) Stops hunting if the dial peer is busy.
Step 9	**dtmf-relay** [**cisco-rtp**] [**rtp-nte**] [**sip-notify**] **Example:** `Router(config-register-pool)# dtmf-relay rtp-nte`	Specifies how a SIP gateway relays dual tone multifrequency (DTMF) tones between telephony interfaces and an IP network. The keywords are defined as follows: • **cisco-rtp** : (Optional) Forwards DTMF tones by using Real-Time Transport Protocol (RTP) with a Cisco proprietary payload type.

	Command or Action	Purpose
		• **rtp-nte** : (Optional) Forwards DTMF tones by using RTP with the Named Telephone Event (NTE) payload type. • **sip-notify** : (Optional) Forwards DTMF tones using SIP NOTIFY messages.

Step 10	**end**
	Example:
	`Router(config-register-pool)# end`
		Returns to privileged EXEC mode.

Verify SIP Registrar Configuration

To help you troubleshoot a SIP registrar and voice register pool, perform the following steps.

SUMMARY STEPS

1. **debug voice register errors**
2. **debug voice register events**
3. **show sip-ua status registrar**

DETAILED STEPS

	Command or Action	Purpose

Step 1	**debug voice register errors**
	Example:
	`Router# debug voice register errors`
	`*Apr 22 11:52:54.523 PDT: VOICE_REG_POOL: Contact`
	`doesn't match any pools`
	`*Apr 22 11:52:54.539 PDT: VOICE_REG_POOL: Register`
	`request for (33015) from (10.2.152.39)`
	`*Apr 22 11:52:54.539 PDT: VOICE_REG_POOL: Contact`
	`doesn't match any pools.`
	`*Apr 22 11:52:54.559 PDT: VOICE_REG_POOL: Register`
	`request for (33017) from (10.2.152.39)`
	`*Apr 22 11:53:04.559 PDT: VOICE_REG_POOL: Maximum`
	`registration threshold for pool(3) hit`
		Use this command to debug errors that happen during registration.
		If there are no voice register pools configured for a particular registration request, the message “Contact doesn’t match any pools” is displayed.

Step 2	**debug voice register events**
	Example:
	`Router# debug voice register events`
	`Apr 22 10:50:21.731 PDT: VOICE_REG_POOL: Contact`
	`matches pool 1`
	`Apr 22 10:50:21.731 PDT: VOICE_REG_POOL: key(91011)`
	`contact(192.168.0.2) add to contact table`
	`Apr 22 10:50:21.731 PDT: VOICE_REG_POOL: key(91011)`
	`exists in contact table`
	`Apr 22 10:50:21.731 PDT: VOICE_REG_POOL:`
	`contact(192.168.0.2) exists in contact table, ref`
	`updated`
	`Apr 22 10:50:21.731 PDT: VOICE_REG_POOL: Created`
		Using the **debug voice register events** command should suffice to display registration activity. Registration activity includes matching of pools, registration creation, and automatic creation of dial peers. For more details and error conditions, you can use the **debug voice register errors** command.
		The phone number 91011 registered successfully, and *type 1* is reported, which means there is a pre-existing VoIP dial peer.

	Command or Action	Purpose

	`dial-peer entry of type 1`
	`Apr 22 10:50:21.731 PDT: VOICE_REG_POOL:`
	`Registration successful for 91011, registration id`
	`is 257`	

Step 3	**show sip-ua status registrar**
	Example:
	`Router# show sip-ua status registrar`
	`Line    destination expires(sec) contact`
	`======= =========== ============ =======`
	`91021   192.168.0.3 227           192.168.0.3`
	`91011   192.168.0.2 176           192.168.0.2`
	`95021   10.2.161.50 419           10.2.161.50`
	`95012   10.2.161.50 419           10.2.161.50`
	`95011   10.2.161.50 420           10.2.161.50`
	`95500   10.2.161.50 420           10.2.161.50`
	`94011   10.2.161.40 128           10.2.161.40`
	`94500   10.2.161.40 129           10.2.161.40`
		Use this command to display all the SIP endpoints currently registered with the contact address.

Verify Proxy Dial-Peer Configuration

To use the **icmp-ping** keyword with the **proxy** command to assist in troubleshooting proxy dial peers, perform the following steps.

SUMMARY STEPS

1. **configure terminal**
2. **voice register pool** *tag*
3. **proxy** *ip-address* [**preference** *value*] [**monitor probe** {**icmp-ping** | **rtr**} [*alternate-ip-address*]]
4. **end**
5. **show voice register dial-peers**
6. **show dial-peer voice**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**configure terminal** **Example:** `Router# configure terminal`	Use this command to enter global configuration mode.
Step 2	**voice register pool** *tag* **Example:** `Router(config)# voice register pool 1`	Use this command to enter voice register pool configuration mode.

Step 3	**proxy** *ip-address* [**preference** *value*] [**monitor probe** {**icmp-ping** | **rtr**} [*alternate-ip-address*]]
	Example:
	`Router(config-register-pool)# proxy 10.2.161.187`
	`preference 1 monitor probe icmp-ping`
		Set the **proxy** command to monitor with **icmp-ping**.

	Command or Action	Purpose
Step 4	**end** **Example:** Router(config-register-pool)# end	Returns to privileged EXEC mode.
Step 5	**show voice register dial-peers** **Example:** Router# show voice register dial-peers dial-peer voice 40035 voip preference 5 destination-pattern 91011 session target ipv4:192.168.0.2 session protocol sipv2 voice-class codec 1 dial-peer voice 40036 voip preference 1 destination-pattern 91011 session target ipv4:10.2.161.187 session protocol sipv2 voice-class codec 1 monitor probe icmp-ping 10.2.161.187	Use this command to verify dial-peer configurations, and notice that icmp-ping monitoring is set.
Step 6	**show dial-peer voice** **Example:** Router# show dial-peer voice VoiceOverIpPeer40036 peer type = voice, information type = voice, description = `', tag = 40036, destination-pattern = `91011', answer-address = `', preference=1, CLID Restriction = None CLID Network Number = `' CLID Second Number sent source carrier-id = `', target carrier-id = `', source trunk-group-label = `', target trunk-group-label = `', numbering Type = `unknown' group = 40036, Admin state is up, Operation state  is up, incoming called-number = `', connections/maximum = 0/unlimited, ! Default output for incoming called-number command DTMF Relay = disabled, modem transport = system, huntstop = disabled, in bound application associated: 'DEFAULT' out bound application associated: '' dnis-map = permission :both incoming COR list:maximum capability ! Default output for cor command outgoing COR list:minimum requirement ! Default output for cor command Translation profile (Incoming): Translation profile (Outgoing): incoming call blocking:	Use the **show dial-peer voice** command on dial peer 40036, and notice the monitor probe status. **Note** Also highlighted is the output of the **cor** and **incoming called-number** commands.

	Command or Action	Purpose
	translation-profile = `' disconnect-cause = `no-service' advertise 0x40 capacity_update_timer 25 addrFamily 4 oldAddrFamily 4 type = voip, session-target = `ipv4:10.2.161.187', technology prefix: settle-call = disabled ip media DSCP = ef, ip signaling DSCP = af31, ip video rsvp-none DSCP = af41,ip video rsvp-pass DSCP = af41 ip video rsvp-fail DSCP = af41, UDP checksum = disabled, session-protocol = sipv2, session-transport = system, req-qos = best-effort, acc-qos = best-effort, req-qos video = best-effort, acc-qos video = best-effort, req-qos audio def bandwidth = 64, req-qos audio max bandwidth = 0, req-qos video def bandwidth = 384, req-qos video max bandwidth = 0, RTP dynamic payload type values: NTE = 101 Cisco: NSE=100, fax=96, fax-ack=97, dtmf=121, fax-relay=122 S=123, ClearChan=125, PCM switch over u-law=0,A-law=8 RTP comfort noise payload type = 19 fax rate = voice, payload size = 20 bytes fax protocol = system fax-relay ecm enable fax NSF = 0xAD0051 (default) codec = g729r8, payload size = 20 bytes, Media Setting = flow-through (global) Expect factor = 0, Icpif = 20, Playout Mode is set to adaptive, Initial 60 ms, Max 300 ms Playout-delay Minimum mode is set to default, value 40 ms Fax nominal 300 ms Max Redirects = 1, signaling-type = cas, VAD = enabled, Poor QOV Trap = disabled, Source Interface = NONE voice class sip url = system, voice class sip rel1xx = system, monitor probe method: icmp-ping ip address: 10.2.161.187, Monitored destination reachable voice class perm tag = `' Time elapsed since last clearing of voice call statistics never Connect Time = 0, Charged Units = 0, Successful Calls = 0, Failed Calls = 0, Incomplete Calls = 0 Accepted Calls = 0, Refused Calls = 0, Last Disconnect Cause is "", Last Disconnect Text is "", Last Setup Time = 0.	

Unified SRST, Unified E-SRST, and Unified Secure SRST Password Policy

From Unified SRST 12.6 Release (Cisco IOS XE Gibraltar 16.11.1a) onwards, all configurations on Unified SRST, Unified E-SRST, and Unified Secure SRST must meet the password policy.

General Password Policy Guidelines:

- Passwords must have a minimum of 6 alphanumeric characters, and a maximum of 15 alphanumeric characters.
- Passwords must not contain symbols or special characters.
- Passwords must contain at least one numeral, one uppercase alphabet, and one lowercase alphabet.

If the password is not configured as per the policy, the Unified SRST router displays an error message:

```
Error: The password you have entered is incorrect.
Your password must contain:
1. A minimum of 6 and a maximum of 15 alphanumeric characters, excluding symbols and
special characters.
2. A minimum of one numeral, one uppercase alphabet, and one lowercase alphabet.
```

The Unified CME password policy is applicable for Unified SRST configurations on Cisco IOS XE 16.11.1a and later. Unified SRST password policy is not applicable in the following scenarios:

- Upgrade from an older IOS version to Cisco IOS XE 16.11.1a
- Downgrade from Cisco IOS XE 16.11.1a to an older version

Guidelines for Password Configuration and Encryption

Configure the passwords relevant to Unified SRST, Unified E-SRST, and Unified Secure SRST using the CLI commands as follows:

- **call-manager-fallback** configuration mode
- **xml user** *username* **password [0|6]***password privilege-level*

Note The 0 in the parameter **[0|6]** mentioned in the CLI command represents plain, unencrypted text and 6 represents level 6 password encryption.

- Apart from the parameter configurations ([0|6]) at the command level, configure the Unified SRST router to support encryption.
- Configure the CLI command **encrypt password** under **call-manager-fallback** configuration mode to support type 6 encryption on the Unified SRST router.
- Also, it is mandatory to configure **key config-key password-encrypt***[key]***password encryption** *aes* to support encryption on the Unified SRST router.

- If the key used to encrypt the password is replaced with a new key (replace key or re-key), then the password is re-encrypted with the new key.
- You must adhere to SRST Password Policy for both type 0 and type 6 parameters that you configure on Unified SRST.
- Configure **no encrypt password** for type 0 password on the Unified SRST router. A type 0 password is displayed as unencrypted plain text.
- If you are performing a downgrade from Unified SRST 12.6 to an earlier version, then you must execute the CLI command **no encrypt password**. If the CLI command **no encrypt password** is configured, the password is presented as plain text.

The following is a sample configuration on Unified SRST router to support password encryption:

```
Router(config)#key config-key password-encrypt <cisco123>
Router(config)#password encryption aes
Router(config)#call-manager-fallback
Router(config-cm-fallback)encrypt password
```

Deprecation of CLI commands

From Unified SRST Release 12.6 onwards, the following CLI commands that are configured under **call-manager-fallback** configuration mode are deprecated to enhance product security:

- **log password***password-string*
- **xmltest**
- **xmlschema***schema-url*
- **xmlthread** *number*

Removal of Passwords and Keys from Logs

From Unified SRST Release 12.6 onwards, passwords and sRTP keys are not printed to logs to enhance security of Unified SRST. The information about keys is available only in the show commands from Unified SRST 12.6 release onwards. The CLI command **show ephone offhook** for SCCP and **show sip-ua calls** for SIP are enhanced to display the keys that are in use per media stream, along with the sRTP Ciphers.

The following is a sample output for the show command, **show sip-ua calls**. The lines that are added to the show command output as part of the Unified SRST 12.6 enhancement are the local crypto key and the remote crypto key:

```
SIP UAC CALL INFO
Number of SIP User Agent Client(UAC) calls: 0
SIP UAS CALL INFO
Call 1
SIP Call ID : 007278df-12e00376-6ed02377-6ffbaca9@8.55.0.195
State of the call : STATE_ACTIVE (7)
Substate of the call : SUBSTATE_NONE (0)
Calling Number : 1001
Called Number : 6901%23
Called URI : sip:6901%23@8.39.25.11;user=phone
Bit Flags : 0x10C0401C 0x10000100 0x4
CC Call ID : 196
```

```
Local UUID : 61488a9100105000a000007278df12e0
Remote UUID : c4b7f9475629538096ef61699b96746f
Source IP Address (Sig ): 8.39.25.11
Destn SIP Req Addr:Port : [8.55.0.195]:52704
Destn SIP Resp Addr:Port: [8.55.0.195]:52704
Destination Name : 8.55.0.195
Number of Media Streams : 1
Number of Active Streams: 1
RTP Fork Object : 0x0
Media Mode : flow-through
Media Stream 1
State of the stream : STREAM_ACTIVE
Stream Call ID : 196
Stream Type : voice+dtmf (1)
Stream Media Addr Type : 1
Negotiated Codec : g711ulaw (160 bytes)
Codec Payload Type : 0
Negotiated Dtmf-relay : rtp-nte
Dtmf-relay Payload Type : 101
QoS ID : -1
Local QoS Strength : BestEffort
Negotiated QoS Strength : BestEffort
Negotiated QoS Direction : None
Local QoS Status : None
Media Source IP Addr:Port: [8.39.25.11]:8080
Media Dest IP Addr:Port : [8.55.0.195]:23022
Local Crypto Suite : AEAD_AES_256_GCM
Remote Crypto Suite : AEAD_AES_256_GCM (
AEAD_AES_256_GCM
AEAD_AES_128_GCM
AES_CM_128_HMAC_SHA1_80
AES_CM_128_HMAC_SHA1_32 )
Local Crypto Key : 3taqc13ClF6BBpvd65WTMPrad/i0uyQ6iNouh+jYHxbf48d4TFmsOGyh4Vs=
Remote Crypto Key : 2/TNTV+Rc1Nh/wbGj0MGwIsLrJ4l+N2jKWGczolEnf7sgsA0Q9AEIz0a4eg=
Mid-Call Re-Assocation Count: 0
SRTP-RTP Re-Assocation DSP Query Count: 0
```

The following is a sample output for the show command, **show ephone offhook** . The lines that are added to the show command output as part of the Unified SRST 12.6 enhancement are local key and remote key.

```
ephone-1[0] Mac:549A.EBB5.8000 TCP socket:[1] activeLine:1 whisperLine:0 REGISTERED in
SCCP
ver 21/17 max_streams=1 + Authentication + Encryption with TLS connection
mediaActive:1 whisper_mediaActive:0 startMedia:1 offhook:1 ringing:0 reset:0 reset_sent:0
paging 0 debug:0 caps:8
IP:8.44.22.63 * 17872 SCCP Gateway (AN) keepalive 28 max_line 1 available_line 1
port 0/0/0
button 1: cw:1 ccw:(0 0)
dn 1 number 6901 CM Fallback CH1 CONNECTED CH2 IDLE
Preferred Codec: g711ulaw
Lpcor Type: none Active Secure Call on DN 1 chan 1 :6901 8.44.22.63 18116
to 8.39.25.11 8066 via 8.39.0.1
G711Ulaw64k 160 bytes no vad
SRTP cipher: AES_CM_128_HMAC_SHA1_32
local key: 0OPV0yxvcnRLPMzHfmYbwgHfdxcuS1uPbp5j/Tjk
remote key: e8DQl3Kvk7LjZlipaCoMg9TMreBmiPsFmNiVHwIA
Tx Pkts 0 bytes 0 Rx Pkts 0 bytes 0 Lost 0
Jitter 0 Latency 0 callingDn -1 calledDn -1
```

Toll Fraud Prevention for SIP Line Side on Unified SRST

Unified SRST Release 12.6 enhances the existing Toll Fraud Prevention feature by enforcing security on the SIP line side of Unified SRST. The feature enhancement secures the Unified SRST system against potential toll fraud exploitation by unauthorized users from the SIP line side.

Note Unified SRST 8.1 to 12.5 Releases restricts toll fraud prevention only to securing calls over the SIP trunk. For more information about Toll Fraud Prevention over a SIP trunk, see Configuring a Trusted IP Address List for Toll-Fraud Prevention.

Some of the key features of Toll Fraud Prevention on Unified SRST for secure calls over SIP lines are:

- Authenticates all the SIP line messages that are triggered from the endpoints to Unified SRST.
- If the IP address of the endpoint is not part of the IP address trusted list, the call is rejected by Unified SRST.
- Unified SRST authenticates both IPv4 an IPv6 addresses as part of the toll fraud prevention mechanism.

Prerequisites for Configuring Toll Fraud Prevention for SIP Line Side

- Unified SRST 12.6 or a later version.
- Cisco IOS XE Gibraltar Release 16.11.1a or later.

Configuration Recommendations for Toll Fraud Prevention on Unified SRST

Unified SRST 12.6 enforces security and toll fraud prevention for SIP line side on Unified SRST. The **ip address trusted authentication** configuration blocks unauthorized calls from the line side. Hence, the toll fraud prevention feature secures Unified SRST 12.6 and later from unauthorized users on the line side.

The IP addresses of SRST endpoints are available before registration with Unified SRST, as they are configured (under **voice register pool**) for fallback from Unified CM. Hence, it is not mandatory that the endpoints are registered to Unified SRST for configuring toll fraud prevention.

The IP trust list for Unified SRST is populated based on the IP address information available under **voice register pool** configuration mode. You can find the IP address of the SIP endpoints on Unified SRST under the following commands in voice register pool configuration mode:

- **id ip** (For example, **id ip***192.168.0.0*)
- **id network** (For example, **id network***192.168.25.0***mask***255.255.255.0*)

Sometimes, IP addresses of endpoints are not available to Unified SRST before registration. Consider a scenario where **id device-id** is the CLI command configured under voice register pool configuration mode to define the device name. Then, the IP address of the device or endpoint is available to Unified SRST only during registration.

The following are the configurations of Toll Fraud Prevention in Unified SRST, 12.6:

- The CLI command **ip address trusted authentication** is enabled by default in Unified SRST. The command **ip address trusted authentication** ensures that security is enabled on the Unified SRST system.
- You can manually configure your Unified SRST endpoints as trusted by entering the IP address or subnet of the trusted phone under the**iptrust-list** configuration mode, as follows:

```
Router#config t
Router(config)#voice service voip
Router(conf-voi-serv)#ip address trusted list
Router(cfg-iptrust-list)#ipv4 192.168.10.0 /16
OR
Router(cfg-iptrust-list)#ipv4 192.168.12.0 255.255.255.0
```

- You can verify the manually added IP address of the Unified SRST endpoint, as follows:

```
Router#show running-config | section voice service voip
voice service voip
ip address trusted list
ipv4 192.168.10.1
ipv4 192.168.10.2 255.255.0.0
ipv4 192.168.10.3 255.255.0.0
ipv4 192.168.10.4 255.255.255.0
```

- The CLI command **ip address trusted list** under **voice service voip** configuration mode supports manual configuration of trusted IP addresses.
- The CLI command **show ip address trusted check** provides information on whether a particular IP address is trusted or not.
- The CLI command **silent-discard untrustedsip** in configuration mode silently discards SIP requests from untrusted sources. This command is enabled by default on Unified SRST.
- The **show ip address trusted list** CLI command displays a list of trusted IP addresses. The trusted IP addresses are displayed under the following lists:
- Dial Peer (only applicable for trunk side): Provides details on the IP address of the trunk that is configured under the dial-peer configuration mode.
- Configured IP Address Trusted List: Provides details on the manually configured IP addresses that are trusted.
- Dynamic IP Address Trusted List: Provides details on the IP address of all the phones that are configured for fallback from Unified CM. This list is introduced in Unified CME 12.6 Release.
- Server Group: Provides details on the IP address of the phones that are configured under server-groups configuration mode.

```
Router>enable
Router#show ip address trusted list
IP Address Trusted Authentication
Administration State: UP
Operation State: UP
IP Address Trusted Call Block Cause: call-reject (21)
VoIP Dial-peer IPv4 and IPv6 Session Targets:
Peer Tag Oper State Session Target
-------- ---------- --------------
4        UP        ipv4:10.65.125.155
Configured IP Address Trusted List:
ipv4 192.168.20.1
```

```
ipv4 192.168.20.2 255.255.0.0
ipv4 192.168.20.3 255.255.0.0
ipv4 192.168.20.4 255.255.255.0
Dynamic IP Address Trusted List:
IP Address                                   Subnet Mask     Count Reason
-------------------------------------------- --------------- ----- ----------------
ipv4:8.55.0.0                                255.255.0.0         1 Pool Configured
ipv4:192.168.0.1                             255.255.0.0         2 Pool Configured
ipv6:2001:420:54FF:13::312:0                 119                 1 Pool Configured
ipv4:8.55.22.15                                                  1 Phone Registered
```

Note The column Count in Dynamic IP Address Trusted List displays the number of directory numbers (DNs) sharing the same IP address. For example, ipv4 192.168.0.1 with count 2 represents two DNs sharing the IP address 192.168.0.1.

Note The output of **show ip address trusted list** command displays the entry in column **Type** as ‘Phone Registered’ if **id device-id** is configured.

Upgrade Considerations

When you upgrade to Unified SRST 12.6 version, you need not perform extra configurations for supporting toll fraud prevention. All the endpoints that are manually configured or auto-registered on Unified SRST are added to the Unified SRST IP Address Trust List. You can view the list of trusted IP addresses under the output of the CLI command **show ip address trusted list**.

Configure Toll Fraud Prevention

Configure IP Address Trusted Authentication for Incoming VoIP Calls

Before you begin

- Unified SRST 8.1 or a later version for secure trunk calls.
- Unified SRST 12.6 or a later version for secure line and trunk calls.
- The CLI command **silent-discard untrusted** needs to be configured for the feature to work

Restrictions

For an incoming VoIP call, IP trusted authentication must be invoked when the IP address trusted authentication is in “UP” operational state.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**

4. **ip address trusted authenticate**
5. **ip-address trusted call-block cause**
6. **end**
7. **show ip address trusted list**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice service voip** **Example:** `Router(config)# voice service voip`	Enters voice service voip configuration mode.
Step 4	**ip address trusted authenticate** **Example:** `Router(conf-voi-serv)# ip address trusted authenticate`	Enables IP address authentication on incoming H.323 or SIP trunk calls for toll fraud prevention support. IP address trusted list authenticate is enabled by default. Use the **no ip address trusted list authenticate** command to disable the IP address trusted list authentication.
Step 5	**ip-address trusted call-block cause** **Example:** `Router(conf-voi-serv)#ip address trusted call-block cause call-reject`	Issues a cause-code when the incoming call is rejected to the IP address trusted authentication. This command is enabled by default. **Note** If the IP address trusted authentication fails, a call-reject (21) cause-code is issued to disconnect the incoming VoIP call.

Step 6	**end**
	Example:
	`Router()# end`
		Returns to privileged EXEC mode.

Step 7	**show ip address trusted list**
	Example:
	`Router# #show ip address trusted list`
	`IP Address Trusted Authentication`
	`Administration State: UP`
	`Operation State: UP`
	`IP Address Trusted Call Block Cause:`
	`call-reject (21)`
		Verifies a list of valid IP addresses.

Example

```
Router>enable
Router#show ip address trusted list
IP Address Trusted Authentication
Administration State: UP
Operation State: UP
IP Address Trusted Call Block Cause: call-reject (21)
VoIP Dial-peer IPv4 and IPv6 Session Targets:
Peer Tag Oper State Session Target
-------- ---------- -------------
Configured IP Address Trusted List:
ipv4 192.168.20.1
ipv4 192.168.20.2 255.255.0.0
ipv4 192.168.20.3 255.255.0.0
ipv4 192.168.20.4 255.255.255.0
Dynamic IP Address Trusted List:
IP Address                                  Subnet Mask     Count Type
------------------------------------------- --------------- ----- ----------------
ipv4:8.55.0.0                               255.255.0.0         1 Pool Configured
ipv4:192.168.0.1                            255.255.0.0         1 Pool Configured
```

Add Valid IP Addresses For Incoming VoIP Calls

Before you begin

Cisco Unified CME 8.1 or a later version.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **ip address trusted list**
5. **ipv4 ipv4 address network mask** { <*ipv4 address*>[<*network mask*>] }
6. **end**
7. **show ip address trusted list**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.

	Command or Action	Purpose
Step 3	**voice service voip** **Example:** `Router(config)# voice service voip`	Enters voice service voip configuration mode.

Step 4	**ip address trusted list**
	Example:
	`Router(conf-voi-serv)# ip address trusted list`
	`Router(cfg-iptrust-list)#`
		Enters ip address trusted list mode and allows to manually add additional valid IP addresses.

Step 5	**ipv4 ipv4 address network mask** { <*ipv4 address*>[<*network mask*>] }
	Example:
	`Router(cfg-iptrust-list)#ipv4 172.19.245.1`
	`Router(cfg-iptrust-list)#ipv4 172.19.243.1`
		Allows you to add up to 100 IPv4 addresses in ip address trusted list. Duplicate IP addresses are not allowed in the ip address trusted list.
		• *network mask* — allows to define a subnet IP address.

Step 6	**end**
	Example:
	`Router(config-register-pool)# end`
		Returns to privileged EXEC mode.

Step 7	**show ip address trusted list** **Example:** `Router# show shared-line`	Displays a list of valid IP addresses for incoming H.323 or SIP trunk calls.

Example

The following example shows three IP addresses configured as trusted IP addresses:

```
Router#show ip address trusted list
IP Address Trusted Authentication
Administration State: UP
Operation State: UP
IP Address Trusted Call Block Cause: call-reject (21)
Configured IP Address Trusted List:
ipv4 192.168.20.1
ipv4 192.168.20.2 255.255.0.0
ipv4 192.168.20.3 255.255.0.0
ipv4 192.168.20.4 255.255.255.0
```

Troubleshooting Tips for Toll Fraud Prevention

For troubleshooting toll fraud mechanism supported on Unified SRST, you can enable the CLI commands **debug voip iptrust debug** and **debug voip iptrust detail**, as follows:

```
Router#debug voip iptrust
voip iptrust debugging is on
Router#debug voip iptrust detail
voip iptrust detail debugging is on
```

VRF Support for Unified SRST

Virtual Routing and Forwarding (VRF) for Unified SRST divides a physical router into multiple logical routers. Each of these logical routers has its own set of interfaces and routing and forwarding tables. VRF support allows you to bind the Unified SRST feature to a specific VRF. Previously with the Cisco 4000 Series Integrated Services Routers, Unified SRST was always associated with the global or default routing instance.

From Unified SRST Release 12.8 (Cisco IOS XE 17.2.1r), support is introduced for VRF functionality on Cisco 4000 Series Integrated Services Router. Before Unified SRST Release 12.8 (Cisco IOS XE 17.2.1r), support for VRF was available only on Cisco Integrated Services Router Generation 2 platform.

From Unified SRST Release 12.8, the following support is available for VRF:

- VRF for line side on Cisco 4000 Series Integrated Services Routers– Introduced in Unified SRST 12.8
- VRF support for Unified SRST 12.8 and later releases is compatible with SIP trunks that are configured to use a VRF. However, you can configure different VRFs for the trunk and Unified SRST.

Information About VRF Support

Typically, service providers use a VRF between Provider Edge (PE) and Customer Edge (CE) routers to provide VPN support for customers. VRF is also used to segment data and voice traffic for improved traffic management. VRF can be configured on an interface to process incoming packets according to the assigned VRF.

By configuring VRF-awareness on voice gateways, you can specify a VRF for the voice traffic that is generated from within the gateway. Voice VRF is added to the VoIP service provider interface (SPI) of the gateway to send and receive signaling and media packets in the configured VRF. The SPI can send and receive signaling and media packets only in the configured VRF.

Note We recommend that you configure **voice vrf** for Unified SRST. For more information, see Design Recommendations for VRF.

Design Recommendations for VRF

- SIP endpoints supported by Unified SRST, including Cisco IP Phone 7800 Series, Cisco IP Phone 8800 Series, and Cisco Jabber support VRF for Unified SRST.
- VRF support is offered for both secure and nonsecure deployments of Unified SRST.
- Configuring SRST to use a VRF is compatible with both SIP and TDM trunk configurations.
- If Global Bind and **voice vrf** are configured on the Unified SRST, then preference is given to the Global Bind.
- We recommend that
    - For SRST line side, configure VRF using **voice vrf** command.

  - For SIP trunk side, configure VRF using **bind** command configured under **voice class tenant** configuration mode and attach the tenant to the required SIP trunk dial-peer.

- VRF Preference Order—The following is the binding preference order for call processing on the trunk side and line side for SRST:

Preference Order	Bind	Configuration
1	Dial-peer Bind	**bind** command is configured under **dial-peer** configuration mode **Note** This configuration is only for trunk side.
2	Tenant Bind	**bind** command is configured under **voice class tenant** configuration mode **Note** This configuration is only for trunk side.
3	Global Bind	**bind** command is configured under **sip** in **voice service voip** configuration mode. **Note** This configuration is both for trunk side and Unified SRST line side.
4	Voice VRF	**voice vrf** command configuration **Note** This configuration is both for trunk side and Unified SRST line side.

Configuration Examples for VRF

The following is a sample configuration for **voice vrf** in Unified SRST line side:

```
vrf definition vrf1
rd 100:101
!
address-family ipv4
exit-address-family

voice vrf vrf1
interface GigabitEthernet0/0/0
    vrf forwarding vrf1
    ip address 8.44.22.77 255.255.0.0
ip route vrf vrf1 8.0.0.0 255.0.0.0 8.44.0.1
```

The following is a sample configuration of Global bind (**voice service voip**). In this case, both Unified SRST line side and SIP trunks without an explicit binding use the same VRF configuration.

```
voice service voip
 no ip address trusted authenticate
 media statistics
 media bulk-stats
 media disable-detailed-stats
 allow-connections sip to sip
 no supplementary-service sip moved-temporarily
 no supplementary-service sip refer
 supplementary-service media-renegotiate
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 sip
  bind all source-interface GigabitEthernet 0/0/0
  session transport tcp
  min-se 90
  session refresh
  registrar server expires max 120 min 60
!
```

Configure Virtual Routing and Forwarding (VRF) for Unified SRST

Before you begin

- Unified SRST 12.8 or a later version.
- For design recommendations, see Design Recommendations for VRF.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **vrf definition vrf-name**
4. **rd route-distinguisher**
5. **address-family ipv4**
6. **exit-address-family**
7. **voice vrf vrf-name**
8. **interface interface-name**
9. **vrf forwarding customer-vrf-name**
10. **ip address <ip address> <network mask>**
11. **ip route vrf vrf-name <ip address> <networkmask> <ip address>**
12. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**vrf definition vrf-name** **Example:** `Router(config)# vrf definition vrf1`	Creates a VRF with the specified name. In the example, VRF name is vrf1. **Note** Space is not allowed in VRF name.

	Command or Action	Purpose
Step 4	**rd route-distinguisher** **Example:** `Router (config)# rd 100:101`	Creates a VRF table by specifying a route distinguisher.Enter either an AS number and an arbitrary number (xxx:y) or an IP address and arbitrary number (A.B.C.D:y).
Step 5	**address-family ipv4** **Example:** `Router(config)# address-family ipv4`	Configures IPv4 or IPv6 address-family sessions for a VRF configuration in Unified SRST.
Step 6	**exit-address-family** **Example:** `Router(config)# exit-address-family`	Leaves address-family configuration mode without removing the address family configuration.
Step 7	**voice vrf vrf-name** **Example:** `Router(config)# voice vrf vrf1`	Configures a voice VRF in global configuration mode.
Step 8	**interface interface-name** **Example:** `Router(config)# interface GigabitEthernet0/0/0`	Enters the interface configuration mode.
Step 9	**vrf forwarding customer-vrf-name** **Example:** `Router(config-if)# vrf forwarding vrf1`	Associates the customer VRF instance with the tunnel. Packets exiting the tunnel are forwarded to this VRF (inner IP packet routing).
Step 10	**ip address <ip address> <network mask>** **Example:** `Router(config-if)# ip address 8.44.22.77 255.255.0.0`	IP address is assigned to the interface.
Step 11	**ip route vrf vrf-name <ip address> <networkmask> <ip address>** **Example:** `Router(config-if)# ip route vrf vrf1 8.0.0.0 255.0.0.0 8.44.0.1`	(Optional) Generates IP routing information associated with a VRF. **Note** Required only if you need to add static routes.

Step 12	**end**
	Example:
	`Router(config-if)# end`
		Exits to privileged EXEC mode.

IPv6 Support for Unified SRST SIP IP Phones

Internet Protocol version 6 (IPv6) is the latest version of the Internet Protocol (IP). IPv6 uses packets to exchange data, voice, and video traffic over digital networks. Also, IPv6 increases the number of network address bits from 32 bits in IPv4 to 128 bits. From Unified SRST Release 12.0 onwards, Unified SRST supports IPv6 protocols for SIP IP phones.

IPv6 support in Unified SRST allows the network to behave transparently in a dual-stack (IPv4 and IPv6) environment and provides additional IP address space to SIP IP phones that are connected to the network. If you do not have a dual-stack configuration, configure the CLI command **call service stop** under **voice service voip** configuration mode before changing to dual-stack mode. For an example of switching to dual-stack mode, see Examples for Configuring IPv6 Pools for SIP IP Phones.

The Cisco IP Phone 7800 Series and 8800 Series are supported on IPv6 for Unified SRST.

For more information on configuring SIP IP phones for IPv6 source address, see Configure IPv6 Pools for SIP IP Phones.

For an example of configuring IPv6 Support on Unified SRST, see Examples for Configuring IPv6 Pools for SIP IP Phones.

For more details about IPv6 deployment, see IPv6 Deployment Guide for Cisco Collaboration Systems Release 12.0.

Feature Support for IPv6 in Unified SRST SIP IP Phones

The following basic features are supported for a IPv6 WAN down scenario:

- Basic SIP Line (IPv4 or IPv6) to SIP Line calls (IPv4 or IPv6) when Unified SRST is in dual-stack **no anat** mode.

The following supplementary services are supported as part of IPv6 in Unified SRST IP Phones:

- Hold/Resume
- Call Forward
- Call Transfer
- Three-way Conference (with BIB conferencing only)
- Line to T1/E1 Trunk and Trunk to Line with Supplementary Service Features
- Fax to and from PSTN (IPv4 ATA to ISDN T1/E1) for both T.38 Fax Relay and Fax Passthrough

Restrictions

The following are the known restrictions for IPv6 support on Unified SRST:

- SIP Trunks are not supported on Unified SRST for IPv6 deployment. PSTN calls are supported only through T1/E1 trunks.
- SCCP IP Phones are not supported in a deployment of IPv6 for Unified SRST.
- SIP Phones can be either in IPv4 only or IPv6 only mode (**no anat**).
- Trancoding and Transrating are not supported.
- H.323 trunks are not supported.
- Secure SIP lines or trunks are not supported.

- IPv6 on Unified SRST is not supported on the Cisco IOS platform. The support is restricted to Cisco IOS XE platform with Cisco IOS Release 16.6.1 or later versions.

Configure IPv6 Pools for SIP IP Phones

Before you begin

- Unified SRST 12.0 or a later version.
- IPv6 option only appears if protocol mode is dual-stack configured under sip-ua configuration mode or IPv6.
- Cisco Unified SRST License must be configured for the gateway to function as a Unified SRST gateway to support IPv6 functionality. For more information on licenses, see Licensing.
- Cisco Unified Communications Manager (Unified Communications Manager) is provisioned with the IPv6 address of Unified SRST. For information on configuration of Unified SRST on Unified Communications Manager, see the section Survivable Remote Site Telephony Configuration in Cisco Unified Communications Manager Administration Guide.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **ipv6 unicast-routing**
4. **voice service voip**
5. **sip**
6. **no ant**
7. **call service stop**
8. **exit**
9. **exit**
10. **sip-ua**
11. **protocol mode {ipv4 | ipv6 | dual-stack [preference {ipv4 | ipv6}]}**
12. **exit**
13. **voice service {voip}**
14. **sip**
15. **no call service stop**
16. **exit**
17. **voice register global**
18. **default mode**
19. **max-dn** *max-directory-numbers*
20. **max-pool** *max-voice-register-pools*
21. **exit**
22. **voice register pool***pool-tag*
23. **id { network** *address* **mask** *mask* | **ip address mask** *mask* | **mac** *address* }
24. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. - Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**ipv6 unicast-routing** **Example:** `Router(config)# ipv6 unicast-routing`	Enables the forwarding of IPv6 unicast datagrams.
Step 4	**voice service voip** **Example:** `Router (config)# voice service voip`	Enters voice-service configuration mode to specify a voice encapsulation type. **voip** —Specifies Voice over IP (VoIP) parameters.
Step 5	**sip** **Example:** `Router(config-voi-serv)# sip`	Enters SIP configuration mode.
Step 6	**no ant** **Example:** `Router(config-serv-sip)# no anat`	Disables Alternative Network Address Types (ANAT) on a SIP trunk.
Step 7	**call service stop** **Example:** `Router(config-serv-sip)# call service stop`	Shuts down SIP call service.
Step 8	**exit** **Example:** `Router(config-serv-sip)# exit`	Exits SIP configuration mode.
Step 9	**exit** **Example:** `Router(config-voi-serv)# exit`	Exits voice service voip configuration mode.
Step 10	**sip-ua** **Example:** `Router(config)# sip-ua`	Enters SIP user-agent configuration mode.
Step 11	**protocol mode {ipv4 | ipv6 | dual-stack [preference {ipv4 | ipv6}]}** **Example:**	Allows phones to interact with phones on IPv6 voice gateways. You can configure phones for IPv4 addresses, IPv6 addresses, or for a dual-stack mode.

	Command or Action	Purpose
	`Router(config-sip-ua)# protocol mode dual-stack preference ipv6`	• ipv4—Allows you to set the protocol mode as an IPv4 address. • ipv6—Allows you to set the protocol mode as an IPv6 address. • dual-stack—Allows you to set the protocol mode for both IPv4 and IPv6 addresses. • preference—Allows you to choose a preferred IP address family if protocol mode is dual-stack.
Step 12	**exit** **Example:** `Router(config-sip-ua)# exit`	Exits SIP configuration mode.

Step 13	**voice service {voip}**
	Example:
	`Router (config)# voice service voip`
		Enters voice-service configuration mode to specify a voice encapsulation type.
		voip—Specifies Voice over IP (VoIP) parameters.

Step 14	**sip** **Example:** `Router(config-voi-serv)# sip`	Enters SIP configuration mode.
Step 15	**no call service stop** **Example:** `Router(config-serv-sip)# call service stop`	Activates SIP call service.
Step 16	**exit** **Example:** `Router(config-serv-sip)# exit`	Exits SIP configuration mode.
Step 17	**voice register global** **Example:** `Router(config)# voice register global`	Enters voice register global configuration mode to set parameters for all supported SIP phones in Unified SRST.
Step 18	**default mode** **Example:** `Router(config-register-global)# default mode`	Enables mode for provisioning SIP phones in Unified SRST. The default mode is Unified SRST itself.
Step 19	**max-dn** *max-directory-numbers* **Example:** `Router(config-register-global)# max-dn 50`	Limits number of directory numbers to be supported by this router. Maximum number is platform and version-specific. Type ? for value.
Step 20	**max-pool** *max-voice-register-pools* **Example:**	Sets maximum number of SIP phones to be supported by the Unified SRST router.

	Command or Action	Purpose
	`Router(config-register-global)# max-pool 40`	
Step 21	**exit** **Example:** `Router(config-register-global)# exit`	Exits voice register global configuration mode.
Step 22	**voice register pool***pool-tag* **Example:** `Router(config)# voice register pool 1`	Enters voice register pool configuration mode to set phone-specific parameters for a SIP phone.

Step 23	**id {** **network** *address* **mask** *mask* | **ip address mask** *mask* | **mac** *address* }
	Example:
	`Router(config-register-pool)# id network 2001:420:54FF:13::901:0/117`
	`Router(config-register-pool)# id network 10.64.88.0 mask 255.255.255.0`
		Explicitly identifies a locally available individual SIP phone to support a degree of authentication.

Step 24	**end**
	Example:
	`Router(config)# end`
		Exits to privileged EXEC mode.

Configure Unified SRST on Cisco 4000 Series Integrated Services Platform

For Unified SRST Release 10.5 and later, Unified SRST is supported on Cisco 4000 Series Integrated Services Routers. A Unified SRST system supports SIP phones with standard-based RFC 3261 feature support locally and across SIP WAN networks. With Cisco Unified SIP SRST, SIP phones can place calls across SIP networks with similar features, as SCCP phones do. For example, most SCCP phone features such as caller ID, speed dial, and redial are supported on SIP networks, that give users the opportunity to choose SCCP or SIP.

Before you begin

- Cisco IOS XE Denali 16.3.1 or a later release.
- Cisco IP Phones 7800 Series or 8800 Series.
- An appropriate feature license to support Unified SIP SRST on the router.
- You need to configure **voice register global** in your router.
- You need to ensure that your router is in **default mode** (for Unified SRST).

SUMMARY STEPS

1. **enable**

2. **configure terminal**
3. **voice service voip**
4. **allow-connections** *from-type to to-type*
5. **no supplementary-service sip moved-temporarily**
6. **no supplementary-service sip refer**
7. **supplementary-service media-renegotiate**
8. **sip**
9. **registrar server [expires[max** *sec* **][min** *sec* **]]**
10. **exit**
11. **exit**
12. **voice register global**
13. **default mode**
14. **max-dn** *max-directory-numbers*
15. **max-pool** *max-voice-register-pools*
16. **exit**
17. **voice register pool** *pool-tag*
18. **id [network** *address* **mask** *mask* | **ip** *address* **mask** *mask*]
19. **dtmf-relay rtp-nte**
20. **no vad**
21. **codec** *codec-type [bytes]*
22. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice service voip** **Example:** `Router(config)# voice service voip`	Enters voice-service configuration mode and specifies voice-over-IP encapsulation. Enters voice register global configuration mode to set global parameters for all supported Cisco SIP IP phones in a Cisco Unified SIP SRST environment.
Step 4	**allow-connections** *from-type to to-type* **Example:** `Router(config-voi-serv)# allow-connections sip to sip`	Allows connections between specific types of endpoints in a VoIP network.
Step 5	**no supplementary-service sip moved-temporarily** **Example:**	Disables supplementary service for call forwarding.

	Command or Action	Purpose
	`Router(config-voi-serv)# no supplementary-service sip moved-temporarily`	
Step 6	**no supplementary-service sip refer** **Example:** `Router(config-voi-serv)# no supplementary-service sip refer`	Prevents the router from forwarding a REFER message to the destination for call transfers.
Step 7	**supplementary-service media-renegotiate** **Example:** `Router(config-voi-serv)# supplementary-service media-renegotiate`	Enables mid-call media renegotiation for supplementary services.
Step 8	**sip** **Example:** `Router(config-voi-serv)# sip`	Enters SIP configuration mode. Required only if you perform the following step for enabling the SIP registrar function.
Step 9	**registrar server [expires[max** *sec* **][min** *sec* **]]** **Example:** `Router(config-serv-sip)# registrar server expires max 120 min 60`	Enables SIP registrar functionality in Unified SRST. • **expires** : (Optional) Sets the active time for an incoming registration. • **max** *sec* : (Optional) Maximum time for a registration to expire, in seconds. Range: 600 to 86400. Default: 3600. Recommended value: 600. • **min** *sec* : (Optional) Minimum expiration time for a registration, in seconds. The range is from 60 to 3600. The default is 60.
Step 10	**exit** **Example:** `Router(config-serv-sip)# exit`	Exits SIP configuration mode.
Step 11	**exit** **Example:** `Router(config-voi-serv)# exit`	Exits voice-service configuration mode.
Step 12	**voice register global** **Example:** `Router(config)# voice register global`	Enters voice register global configuration mode to set parameters for all supported SIP phones in Unified SRST.
Step 13	**default mode** **Example:** `Router(config-register-global)# default mode`	Enables mode for provisioning SIP phones in Unified SRST. The default mode is Unified SRST itself.
Step 14	**max-dn** *max-directory-numbers* **Example:**	Limits number of directory numbers to be supported by this router.

	Command or Action	Purpose
	`Router(config-register-global)# max-dn 50`	Maximum number is platform and version-specific. Type ? for value.
Step 15	**max-pool** *max-voice-register-pools* **Example:** `Router(config-register-global)# max-pool 40`	Sets maximum number of SIP phones to be supported by the Unified SRST router. Maximum number is platform and version-specific. Type ? for value.
Step 16	**exit** **Example:** `Router(config-register-global)# exit`	Exits voice register global configuration mode.
Step 17	**voice register pool** *pool-tag* **Example:** `Router(config)# voice register pool 1`	Enters voice register pool configuration mode to set phone-specific parameters for a SIP phone.
Step 18	**id [network** *address* **mask** *mask* | **ip** *address* **mask** *mask*] **Example:** `Router(config)# voice service voip`	Enters voice service voip configuration mode.
Step 19	**dtmf-relay rtp-nte** **Example:** `Router(config-register-pool)# dtmf-relay rtp-nte`	Forwards DTMF tones by using Real-Time Transport Protocol (RTP) with the Named Telephone Event (NTE) payload type and enables DTMF relay using the RFC 2833 standard method.
Step 20	**no vad** **Example:** `Router(config-register-pool)# no vad`	Disables voice activity detection (VAD) on the VoIP dial peer. VAD is enabled by default. Because there is no comfort noise during periods of silence, the call may seem to be disconnected. You may prefer to set no vad on the SIP phone pool.
Step 21	**codec** *codec-type [bytes]* **Example:** `Router(config-register-pool)# codec g729r8`	Specifies the codec supported by a single SIP phone or a VoIP dial peer in a Cisco Unified SIP SRST environment. The *codec - type* argument specifies the preferred codec and can be one of the following: • **g711alaw**: G.711 a–law 64,000 bps. • **g711ulaw**: G.711 mu–law 64,000 bps. • **g729r8**: G.729 8000 bps (default). The *bytes* argument is optional and specifies the number of bytes in the voice payload of each frame
Step 22	**end** **Example:**	Returns to privileged EXEC mode.

	Command or Action	Purpose
	`Router()# end`	

Configure Voice Hunt Groups on Unified SRST

To redirect calls for a specific number (pilot number) to a defined group of directory numbers on Cisco Unified SCCP and SIP IP phones, perform the following steps.

Voice Hunt Group on Unified SRST is supported for Parallel, Sequential, Peer, and Longest-idle hunt groups. Only the basic call features such as Call, Hold or Resume are supported for Unified SRST on Cisco 4000 Series Integrated Services Routers. For support of advanced features such as Auto Logout, Members Logout, and supplementary call features, you need to configure Unified E-SRST. For more information on Voice Hunt Group support on Unified E-SRST, see Unified E-SRST with Support for Voice Hunt Group.

For a list of restrictions of Unified SRST on Cisco 4000 Series Integrated Services Routers, see Restrictions of Unified SRST on Cisco 4000 Series Integrated Services Routers, page 33

Before you begin

- Cisco IOS XE Denali 16.3.1 or later versions.
- Shared Lines are not supported on Unified SRST.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice hunt-group** *hunt-tag* [**longest-idle** | **parallel** | **peer** | **sequential**]
4. **pilot** *number* [**secondary** *number*]
5. **list** *number*
6. **final** *number*
7. **preference** *preference-order* [**secondary***secondary-order*]
8. **hops** *number*
9. **timeout** *seconds*
10. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.

	Command or Action	Purpose
Step 3	**voice hunt-group** *hunt-tag* [**longest-idle** | **parallel** | **peer** | **sequential**] **Example:** `Router(config)# voice hunt-group 1 longest-idle`	Enters voice hunt-group configuration mode to define a hunt group. • *hung-tag* —Unique sequence number of the hunt group to be configured. Range is 1 to100. • **longest idle** —Hunt group in which calls go to the directory number that has been idle for the longest time. • **parallel** —Hunt group in which calls simultaneously ring multiple phones. • **peer** —Hunt group in which the first directory number is selected round-robin from the list. • **sequential** —Hunt group in which directory numbers ring in the order in which they are listed, left to right. • To change the hunt-group type, remove the existing hunt group first by using the **no** form of the command; then, recreate the group.
Step 4	**pilot** *number* [**secondary** *number*] **Example:** `Router(config-voice-hunt-group)# pilot number 8100`	Defines the phone number that callers dial to reach a voice hunt group. • *number*—String of up to 16 characters that represents an E.164 phone number. • Number string may contain alphabetic characters when the number is to be dialed only by the Unified SRST router, as with an intercom number, and not from phone keypads. • **secondary** *number*—(Optional) Keyword and argument combination defines the number that follows as an additional pilot number for the voice hunt group. • Secondary numbers can contain wildcards. A wildcard is a period (.), which matches any entered digit.
Step 5	**list** *number* **Example:** `Router(config-voice-hunt-group)# list 8000, 8010, 8020, 8030`	Creates a list of extensions that are members of a voice hunt group. To remove a list from a router configuration, use the **no** form of this command. • *number*—List of extensions to be added as members to the voice hunt group. Separate the extensions with commas. • Add or delete all extensions in a hunt-group list at one time. You cannot add or delete a single number in an existing list.

	Command or Action	Purpose
		• There must be from 2 to 10 extensions in the hunt-group list, and each number must be a primary or secondary number. • Any number in the list cannot be a pilot number of a parallel hunt group.
Step 6	**final** *number* **Example:** `Router(config-voice-hunt-group)# final 8888`	Defines the last extension in a voice hunt group. • If a final number in one hunt group is configured as a pilot number of another hunt group, the pilot number of the first hunt group cannot be configured as a final number in any other hunt group.
Step 7	**preference** *preference-order* [**secondary***secondary-order*] **Example:** `Router(config-voice-hunt-group)# preference 6`	Sets the preference order for the directory number associated with a voice hunt-group pilot number. **Note** We recommend that the parallel hunt-group pilot number be unique in the system. Parallel hunt groups may not work if there are more than one partial or exact dial-peer match. For example, if the pilot number is “8000” and there is another dial peer that matches “8…”. If multiple matches cannot be avoided, give parallel hunt groups the highest priority to run by assigning a lower preference to the other dial peers. Note that 8 is the lowest preference value. By default, dial peers created by parallel hunt groups have a preference of 0. • *preference-order*—Range is 0 to 8, where 0 is the highest preference and 8 is the lowest preference. Default is 0. • **secondary** *secondary-order*—(Optional) Keyword and argument combination is used to set the preference order for the secondary pilot number. Range is 1 to 8, where 0 is the highest preference and 8 is the lowest preference. Default is 7.
Step 8	**hops** *number* **Example:** `Router(config-voice-hunt-group)# hops 2`	For configuring a peer or longest-idle voice hunt group only. Defines the number of times that a call can hop to the next number in a peer or longest-idle voice hunt group before the call proceeds to the final number. • *number*—Number of hops. Range is 2 to 10, and the value must be less than or equal to the number of extensions specified by the **list** command. • Default is the same number as there are destinations defined under the **list** command.

	Command or Action	Purpose
Step 9	**timeout** *seconds* **Example:** `Router(config-voice-hunt-group)# timeout 100`	Defines the number of seconds after which a call that is not answered is redirected to the next directory number in a voice hunt-group list. Default is 180 seconds.

Step 10	**end**
	Example:
	`Router(config-voice-hunt-group)# end`
		Exits to privileged EXEC mode.

Configure Feature Support on Unified SIP SRST

This section provides configuration information for some of the features supported on Unified SIP SRST.

Configure SIP-to-SIP Call Forwarding

SIP-to-SIP call forwarding (call routing) is available. Call forwarding is provided either by the phone or by using a back-to-back user agent (B2BUA), which allows call forwarding on any dial peer. Calls into a SIP device may be forwarded to other SIP or SCCP devices (including Cisco Unity, third-party voice-mail systems, or an auto attendant or IVR system such as IPCC and IPCC Express). In addition, SCCP IP phones may be forwarded to SIP phones.

Cisco Unity or other voice messaging systems connected by a SIP trunk or SIP user agent are able to pass a message-waiting indicator (MWI) when a message is left. The SIP phone then displays the MWI when indicated by the voice messaging system.

Note SIP-to-H.323 call forwarding is not supported.

To configure SIP-to-SIP call forwarding, you must first allow connections between specific types of endpoints in a Cisco IP-to-IP gateway. The **allow-connections** command grants this capability. Once the SIP-to-SIP connections are allowed, you can configure call forwarding under an individual SIP phone pool. Any of the following commands can be used to configure call forwarding, according to your needs:

Under the **voice register pool**

- **call-forward b2bua all** *directory-number*
- **call-forward b2bua busy** *directory-number*
- **call-forward b2bua mailbox** *directory-number*
- **call-forward b2bua noan** *directory-number* [**timeout** *seconds*]

In a typical Cisco Unified SIP SRST setup, the **call-forward b2bua mailbox** command is not used; however, it is likely to be used in a Cisco Unified SIP Communications Manager Express (CME) environment. Detailed procedures for configuring the **call-forward b2bua mailbox** command are found in the Cisco Unified Communications Manager (CallManager) documentation on Cisco.com.

The command **call-forward b2bua all** needs to point towards the trunk.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register pool** *tag*
4. **call-forward b2bua all** *directory- number*
5. **call-forward b2bua busy** *directory- number*
6. **call-forward b2bua mailbox** *directory- number*
7. **call-forward b2bua noan** *directory- number* **timeout** *seconds*
8. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice register pool** *tag* **Example:** `Router(config)# voice register pool 15`	Enters voice register pool configuration mode. • Use this command to control which phone registrations are accepted or rejected by a Cisco Unified SIP SRST device.
Step 4	**call-forward b2bua all** *directory- number* **Example:** `Router(config-register-pool)# call-forward b2bua all 5005`	Enables call forwarding for a SIP back-to-back user agent (B2BUA) so that all incoming calls are forwarded to another non-SIP station extension (that is, SIP trunk, H.323 trunk, SCCP device or analog/digital trunk). • *directory-number* : Phone number to which calls are forwarded. Represents a fully qualified E.164 number. Maximum length of the phone number is 32.
Step 5	**call-forward b2bua busy** *directory- number* **Example:** `Router(config-register-pool)# call-forward b2bua busy 5006`	Enables call forwarding for a SIP B2BUA so that incoming calls to a busy extension are forwarded to another extension. • *directory-number* : Phone number to which calls are forwarded. Represents a fully qualified E.164 number. Maximum length of the phone number is 32.

Step 6	**call-forward b2bua mailbox** *directory- number*
	Example:
	`Example:`
	`Router(config-register-pool)# call-forward b2bua mailbox 5007`
		Controls the specific voice-mail box selected in a voice-mail system at the end of a call forwarding exchange.
		• *directory-number* : Phone number to which calls are forwarded when the forwarded destination is busy or

	Command or Action	Purpose
		does not answer. Represents a fully qualified E.164 number. Maximum length of the phone number is 32.
Step 7	**call-forward b2bua noan** *directory- number* **timeout** *seconds* **Example:** `Router(config-register-pool)# call-forward b2bua noan 5010 timeout 10`	Enables call forwarding for a SIP B2BUA so that incoming calls to an extension that does not answer after a configured amount of time are forwarded to another extension. This command is used if a phone is registered with a Cisco Unified SIP SRST router, but the phone is not reachable because there is no IP connectivity (there is no response to Invite requests). • *directory-number* : Phone number to which calls are forwarded. Represents a fully qualified E.164 number. Maximum length of the phone number is 32. • **timeout** *seconds*: Duration, in seconds, that a call can ring with no answer before the call is forwarded to another extension. Range is 3 to 60000. The default value is 20.

Step 8	**end**
	Example:
	`Router(config-register-pool)# end`
		Returns to privileged EXEC mode.

Configure Call Blocking Based on Time of Day, Day of Week, or Date

This section applies to both SCCP and SIP SRST. Call blocking prevents the unauthorized use of phones and is implemented by matching a pattern of up to 32 digits during a specified time of day, day of week, or date. Cisco Unified SIP SRST provides SIP endpoints the same time-based call blocking mechanism that is currently provided for SCCP phones. The call blocking feature supports all incoming calls, including incoming SIP and analog FXS calls.

Note Pin-based exemptions and the “Login” toll-bar override are not supported in Cisco Unified SIP SRST.

The commands used for SIP phone call blocking are the same commands that are used for SCCP phones on your Cisco Unified SRST system. The Cisco SRST session application accesses the current after-hours configuration under call-manager-fallback mode and applies it to calls originated by Cisco SIP phones that are registered to the Cisco SRST router. The commands used in call-manager-fallback mode that set block criteria (time/date/block pattern) are the following:

- **after-hours block pattern** *pattern-tag pattern* [**7-24**]
- **after-hours day** *day start-time stop-time*
- **after-hours date** *month date start-time stop-time*

When a user attempts to place a call to digits that match a pattern that has been specified for call blocking during a time period that has been defined for call blocking, the call is immediately terminated and the caller hears a fast busy.

In SRST (call-manager-fallback configuration mode), there is no phone- or pin-based exemption to after-hours call blocking. However, in Cisco Unified SIP SRST (voice register pool mode), individual IP phones can be exempted from all call blocking using the **after-hours exempt** command.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **call-manager-fallback**
4. **after-hours block pattern** *tag pattern* [**7-24**]
5. **after-hours day** *day start-time stop-time*
6. **after-hours date** *month date start-time stop-time*
7. **exit**
8. **voice register pool** *tag*
9. **after-hour exempt**
10. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**call-manager-fallback** **Example:** `Router(config)# call-manager-fallback`	Enters call-manager-fallback configuration mode.
Step 4	**after-hours block pattern** *tag pattern* [**7-24**] **Example:** `Router(config-cm-fallback)# after-hours block pattern 1 91900`	Defines a pattern of outgoing digits to be blocked. Up to 32 patterns can be defined, using individual commands. • If the **7-24** keyword is specified, the pattern is always blocked, 7 days a week, 24 hours a day. • If the **7-24** keyword is not specified, the pattern is blocked during the days and dates that are defined using the **after-hours day** and **after-hours date** commands.
Step 5	**after-hours day** *day start-time stop-time* **Example:**	Defines a recurring time period based on the day of the week during which calls are blocked to outgoing dial

	Command or Action	Purpose
	`Router(config-cm-fallback)# after-hours day mon 19:00 07:00`	patterns that are defined using the **after-hours block pattern** command. • *day* : Day of the week abbreviation. The following are valid day abbreviations: **sun**, **mon**, **tue**, **wed**,**thu**, **fri**, **sat**. • *start-time stop-time* : Beginning and ending times for call blocking, in an HH:MM format using a 24-hour clock. If the stop time is a smaller value than the start time, the stop time occurs on the day following the start time. For example, “mon 19:00 07:00” means “from Monday at 7 p.m. until Tuesday at 7 a.m.” The value 24:00 is not valid. If 00:00 is entered as a stop time, it is changed to 23:59. If 00:00 is entered for both start time and stop time, calls are blocked for the entire 24-hour period on the specified date.
Step 6	**after-hours date** *month date start-time stop-time* **Example:** `Router(config-cm-fallback)# after-hours date jan 1 00:00 00:00`	Defines a recurring time period based on month and date during which calls are blocked to outgoing dial patterns that are defined using the **after-hours block pattern** command. • *month* : Month abbreviation. The following are valid month abbreviations: **jan**, **feb**, **mar**, **apr**, **may**,**jun**, **jul**, **aug**, **sep**, **oct**, **nov**,**dec**. • *date* : Date of the month. Range is from 1 to 31. • *start-time stop-time* : Beginning and ending times for call blocking, in an HH:MM format using a 24-hour clock. The stop time must be larger than the start time. The value 24:00 is not valid. If 00:00 is entered as a stop time, it is changed to 23:59. If 00:00 is entered for both start time and stop time, calls are blocked for the entire 24-hour period on the specified date.
Step 7	**exit** **Example:** `Router(config-cm-fallback)# exit`	Exits call-manager-fallback configuration mode.
Step 8	**voice register pool** *tag* **Example:** `Router(config)# voice register pool 12`	Enters voice register pool configuration mode. • Use this command to control which registrations are accepted or rejected by a Cisco Unified SIP SRST device.

	Command or Action	Purpose
Step 9	**after-hour exempt** **Example:** `Router(config-register-pool)# after-hour exempt`	Specifies that for a particular voice register pool, none of its outgoing calls are blocked although call blocking is enabled.

Step 10	**end**
	Example:
	`Router(config-register-pool)# end`
		Returns to privileged EXEC mode.

Verification

To verify the feature's configuration, enter one of the following commands:

- **show voice register dial-peer** : Displays all the dial peers created dynamically by phones that have registered. This command also displays configurations for after hours blocking and call forwarding.
- **show voice register pool** : Displays information about a specific pool.
- **debug ccsip message** : Debugs basic B2BUA calls.

For more information about these commands, see Cisco Unified SRST and Cisco Unified SIP SRST Command Reference (All Versions).

SIP Call Hold and Resume

Unified SRST supports the ability for SIP phones to place calls on hold and to resume from calls placed on hold. This also includes support for a consultative hold where A calls B, B places A on hold, B calls C, and B disconnects from C and then resumes with A. Support for call hold is signaled by SIP phones using "re-INVITE c=0.0.0.0" and also by the receive-only mechanism.

No configuration is necessary.

Configure Music On Hold for Unified SRST

Unified SRST supports the ability for SIP phones to play music for calls placed on hold. The following is the recommended configuration for Music On Hold (MOH) on a SIP Phone that falls back to Unified SRST.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **no telephony-service**
4. **call-manager-fallback**
5. **moh enable-g711 "bootflash:** *filename"*
6. **moh enable-g729 "bootflash:** *filename"*
7. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**no telephony-service** **Example:** `Router# no telephony-service`	Removes all the configurations for IP phones configured under the telephony-service configuration mode.
Step 4	**call-manager-fallback** **Example:** `Router(config)# call-manager-fallback`	Enters call-manager-fallback configuration mode.
Step 5	**moh enable-g711 "bootflash:** *filename"* **Example:** `Router(config-cm-fallback)# moh enable-g711 "bootflash:music-on-hold.au"`	Generates an audio stream from a router flash file that supports G.711 codec for Music On Hold (MOH) in Unified. SRST.
Step 6	**moh enable-g729 "bootflash:** *filename"* **Example:** `Router(config-cm-fallback)# moh g729 "flash:SampleAudioSource.g729.wav"`	Generates an audio stream from a router flash file that supports G.729 codec for MOH in Unified SRST.

Step 7	**end**
	Example:
	`Router(config-cm-fallback)# end`
		Returns to privileged EXEC mode.

Enabling KPML for SIP Phones

Perform the following steps to enable KPML digit collection on a SIP phone.

Restrictions

A dial plan assigned to a phone has priority over KPML.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register pool** *pool-tag*
4. **digit collect kpml**
5. **end**

6. **show voice register dial-peers**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice register pool** *pool-tag* **Example:** `Router(config)# voice register pool 4`	Enters voice register pool configuration mode to set phone-specific parameters for a SIP phone. • *pool-tag*: Unique sequence number of the SIP phone to be configured. Range is version and platform-dependent; type **?** to display range. You can modify the upper limit for this argument with the **max-pool** command.
Step 4	**digit collect kpml** **Example:** `Router(config-register-pool)# digit collect kpml`	Enables KPML digit collection for the SIP phone. **Note** This command is enabled by default for supported phones in Cisco Unified CME and Cisco Unified SRST.

Step 5	**end**
	Example:
	`Router(config-register-pool)# end`
		Exits to privileged EXEC mode.

Step 6	**show voice register dial-peers** **Example:** `Router# show voice register dial-peer`	Displays details of all dynamically created VoIP dial peers associated with the Cisco Unified CME SIP register including the defined digit collection method.

Disabling SIP Supplementary Services for Call Forward and Call Transfer

Perform the following steps to disable REFER messages for call transfers and redirect responses for call forwarding from being sent to the destination by Unified SRST. You can disable these supplementary features if the destination gateway does not support them.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip** OR **dial-peer voice** *tag* **voip**
4. **no supplementary-service sip** {**moved-temporarily** |**refer**}

5. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.

Step 3	**voice service voip** OR **dial-peer voice** *tag* **voip**
	Example:
	`Router(config)# voice service voip`
	`or`
	`Router(config)# dial-peer voice 99 voip`
		Enters voice-service configuration mode to set global parameters for VoIP features.
		or
		Enters dial peer configuration mode to set parameters for a specific dial peer.

Step 4	**no supplementary-service sip** {**moved-temporarily** |**refer**}
	Example:
	`Router(conf-voi-serv)# no supplementary-service sip refer`
	`or`
	`Router(config-dial-peer)# no supplementary-service sip refer`
		Disables SIP call forwarding or call transfer supplementary services globally or for a dial peer.
		• **moved-temporarily**: SIP redirect response for call forwarding.
		• **refer**: SIP REFER message for call transfers.
		• Sending REFER and redirect messages to the destination is the default behavior.
		Note This command is supported for calls between SIP phones and calls between SCCP phones. It is not supported for a mixture of SCCP and SIP endpoints.

Step 5	**end**
	Example:
	`Router(config-voi-serv)# end`
	`OR`
	`Router(config-dial-peer)# end`
		Exits to privileged EXEC mode.

Configuring idle Prompt Status for SIP Phones

Perform the following steps to customize the message that displays on SIP phones after the phones failover to Cisco Unified SRST.

Note You do not need to create new configuration files with the **create profile** command and restart the phones after changing the idle status message in Cisco Unified SRST. Modifying the status message takes effect immediately in Cisco Unified SRST.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register global**
4. **system message** *string*
5. **end**
6. **show voice register global**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice register global** **Example:** `Router(config)# voice register global`	Enters voice register global configuration mode to set global parameters for all supported SIP phones in a Cisco Unified CME environment.
Step 4	**system message** *string* **Example:** `Router(config-register-global)# system message fallback active`	Defines a status message that displays on SIP phones registered to Cisco Unified SRST. • *string*: Up to 32 alphanumeric characters. Default is “CM Fallback Service Operating.”

Step 5	**end**
	Example:
	`Router(config-register-global)# end`
		Exits to privileged EXEC mode.

Step 6	**show voice register global** **Example:** `Router# show voice register global`	Displays all global configuration parameters associated with SIP phones.

Examples

The following are sample configurations for supporting SIP SRST on Cisco 4000 Series Integrated Services Router.

Example for Configuring Unified SIP SRST on Cisco 4000 Series Integrated Services Routers

The following example shows how to configure Unified SIP SRST on Cisco 4000 Series Integrated Services Routers.

```
!
voice service voip
allow-connections sip to sip
no supplementary-service sip moved-temporarily
no supplementary-service sip refer
supplementary-service media-renegotiate
sip
registrar server expires max 120 min 60
!
!
voice register global
default mode
max-dn 40
max-pool 40
!
voice register pool 1
id network 8.55.0.0 mask 255.255.0.0
dtmf-relay rtp-nte
codec g711ulaw
no vad
!
!
```

Example for Configuring Voice Hunt Groups in Unified SIP SRST

The following example shows how to configure longest-idle hunt group 20 with pilot number 4701, final number 5000, and 6 numbers in the list. After a call is redirected six times (makes 6 hops), it is redirected to the final number 5000.

```
Router(config)# voice hunt-group 20 longest-idle
Router(config-voice-hunt-group)# pilot 4701
Router(config-voice-hunt-group)# list 4001, 4002, 4023, 4028, 4045, 4062
Router(config-voice-hunt-group)# final 5000
Router(config-voice-hunt-group)# hops 6
Router(config-voice-hunt-group)# timeout 20
Router(config-voice-hunt-group)# exit
```

Examples for Configuring IPv6 Pools for SIP IP Phones

The following example provides configuration of IPv6 pools for SIP IP Phones:

```
ipv6 unicast-routing
voice service voip
sip
```

```
no anat
call service stop
exit
exit
sip-ua
protocol mode dual-stack
exit
voice service voip
sip
no call service stop
exit
voice register global
default mode
max-dn 50
max-pool 40
exit
voice register pool 1
id network 2001:420:54FF:13::901:0/117
end
```

The following example provides interface configuration for IPv6 supported on Unified SRST:

```
configure terminal
interface GigabitEthernet0/0/1
ip address 10.64.86.229 255.255.255.0
negotiation auto
ipv6 address 2001:420:54FF:13::312:82/119
ipv6 enable
```

The following example provides IP route configuration for IPv6 supported on Unified SRST:

```
ipv6 route 2001:420:54FF:13::312:0/119 2001:420:54FF:13::312:1
ipv6 route 2001:420:54FF:13::901:0/119 2001:420:54FF:13::312:1
```

The following example displays output when SIP call service is shut down with the call service stop CLI command:

```
Router# show sip service
SIP service is shut
under 'voice service voip', 'sip' submode
```

The following example displays output when SIP call service is active with the no call service stop CLI command:

```
Router# show sip-ua service
SIP Service is up
under 'voice service voip', 'sip' submode
```

Example for Configuring Call Blocking Based on Time of Day, Day of Week, or Date

The following example defines several patterns of digits for which outgoing calls are blocked. Patterns 1 and 2, which block calls to external numbers that begin with 1 and 011, are blocked on Monday through Friday before 7 a.m. and after 7 p.m. Pattern 3 blocks calls to 900 numbers 7 days a week, 24 hours a day.

```
call-manager-fallback
after-hours block pattern 1 91
after-hours block pattern 2 9011
after-hours block pattern 3 91900 7-24
after-hours day mon 19:00 07:00
after-hours day tue 19:00 07:00
after-hours day wed 19:00 07:00
```

```
after-hours day thu 19:00 07:00
after-hours day fri 19:00 07:00
```

The following example exempts a Cisco SIP phone pool from the configured blocking criteria:

```
voice register pool 1
after-hour exempt
```

Example for Configuring Music On Hold for Unified SIP SRST

The following example shows how to configure Music On Hold (MOH) for Unified SIP SRST on Cisco 4000 Series Integrated Services Routers.

```
enable
configure terminal
no telephony-service
call-manager-fallback
moh enable-g711 "flash:music-on-hold.au"
moh g729 "flash:SampleAudioSource.g729.wav"
```

Example for Configuring SIP-to-SIP Call Forwarding on Unified SRST

The following is a sample configuration for SIP-to-SIP Call Forwarding on Unified SRST.

```
enable
configure terminal
voice register pool 15
call-forward b2bua busy 5006
call-forward b2bua mailbox 5007
call-forward b2bua noan 5010 timeout 8
```

Example for Configuring idle Prompt Status for SIP Phones

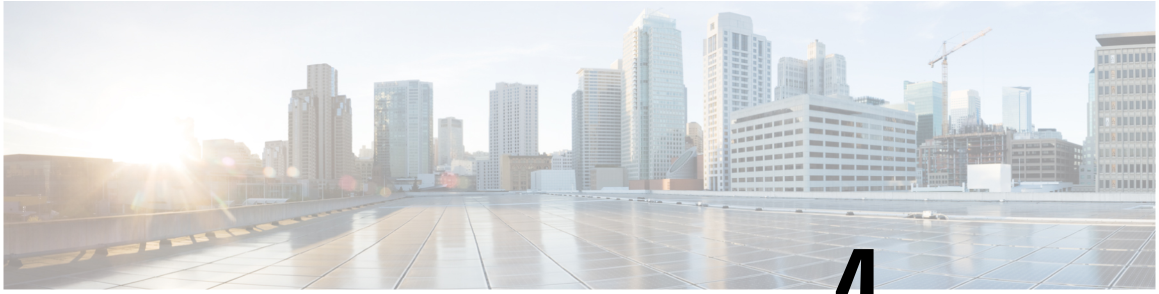
The following is a sample configuration for idle prompt status for SIP phones on Unified SRST.

```
enable
configure terminal
voice register global
system message fallback active
end
show voice register global
```

Example for Disabling SIP Supplementary Services for Call Forward and Call Transfer

The following is a sample configuration for disabling SIP supplementary services for call forward and call transfer on Unified SRST.

```
enable
configure terminal
voice service voip
no supplementary-service sip {moved-temporarily | refer}
end
```



CHAPTER 4

Enhanced SRST

This chapter describes the Unified Enhanced Survivable Remote Site Telephony (Unified E-SRST) feature which is an enhancement of the SRST feature that provides advanced services compared to the classic Unified SRST.

Migration from Cisco Unified SRST Manager to Unified E-SRST

Cisco Unified Survivable Remote Site Telephony Manager is End-of-Life (EOL). Hence, provisioning for Unified E-SRST through Cisco Unified SRST Manager is not supported for Unified E-SRST Release 12.2 and later releases. Unified E-SRST is provisioned only using CLI commands (manual provisioning) to support fall back of phones registered to Cisco Unified Communications Manager. For more information on configuring Unified E-SRST see SIP: Configure Unified E-SRST and SCCP: Configure Unified E-SRST.

For information on Cisco Unified Survivable Remote Site Telephony Manager End-of-Life announcement, see Cisco Unified Survivable Remote Site Telephony Manager Product Bulletin.

Cisco Unified SRST Manager is a GUI-based tool that helps to monitor, report, and troubleshoot remote sites. It performs automatic sync up between the Cisco Unified Communications Manager and the Unified E-SRST gateway that helps in adding, deleting, and modifying the users and phones including dial-plan mapping. It also provides centralized management and control of all remote sites. For more information on the Cisco Unified SRST Manager that is End-of-Life, see Administration Guide for Cisco Unified SRST Manager.

Benefits

When you configure Unified E-SRST, it provides the following feature benefits in comparison to the classic Cisco Unified SRST:

- Voice Hunt Group
    - Shared Lines

    - Mixed Shared Lines (SIP and SCCP Phones)
    - Hunt Statistics Collection
    - Mixed Deployment (SIP and SCCP Phones)

- Shared Line
- BLF
- Video
- B-ACD

For more information on configuring VHG with Unified E-SRST, see Unified E-SRST with Support for Voice Hunt Group.

For more information on configuring Shared Line, BLF, and Video with Unified E-SRST, see SIP: Configure Unified E-SRST.

Restrictions

- Supports the Version Negotiation feature only on the Cisco Unified 9951, 9971, 8961 SIP IP phones, Cisco IP Phone 7800, and 8800 Series.
- The phone firmware version is version 9.4.1 or later versions.
- This feature supports video calls only between the local Cisco Unified SIP IP phones and the No Time-Division Multiplexing (TDM) video calls during the SRST failovers.
- To enable phone-specific features like shared-line & BLF work, configure the individual voice register Pools.

Restrictions for Unified E-SRST, Release 12.2

The Unified E-SRST deployment with the voice hunt group has the following restrictions:

- Does not support the auto logout.
- Does not support Programmable Line Keys (PLK).
- Does not support HLog Softkey.

Note The existing support for Cisco Jabber is now End of Life (EOL). Hence, does not support Cisco Jabber on Cisco Unified SRST, Unified E-SRST.

Support for Cisco Unified IP Phones and Platforms

The following section provides information about platform support for Cisco Unified IP Phones:

- Unified E-SRST is supported on Cisco 1100 Series Integrated Services Router (ISR) Platforms with Cisco IOS XE Bengaluru 17.5.1a and later releases.

- Unified E-SRST is supported on Cisco 4000 Series ISR Platforms (4321, 4331, 4351, 4431, and 4451) on all Cisco IOS XE releases.
- Unified E-SRST is supported on Cisco 4461 Series ISR Platforms with Cisco IOS XE 16.10.1a and later releases.
- Unified E-SRST is supported on Cisco Catalyst 8300 Series Edge Platforms with Cisco IOS XE Amsterdam 17.3.2 and later releases.
- Unified E-SRST is supported on Cisco Catalyst 8200 Series Edge Platforms with Cisco IOS XE Bengaluru 17.4.1a and later releases.
- Unified E-SRST is supported on Cisco Catalyst 8200L Series Edge Platforms with Cisco IOS XE Bengaluru 17.5.1a and later releases.

Licensing

This section provides information on licensing of Cisco Unified Enhanced Survivable Remote Site Telephony (Unified E-SRST).

Cisco Smart Licensing for Unified E-SRST

Cisco Smart Licensing is a software licensing model that provides visibility of ownership and usage through the Cisco Smart Software Manager (CSSM) portal. CSSM is a central license repository that manages licenses across all Cisco products that you own, including Unified E-SRST. Devices send license usage to CSSM either directly or use an on-premises satellite. Your Smart Account Administrator controls your access to CSSM. Use your Cisco credentials to access the CSSM portal using http://software.cisco.com.

Smart Licensing applies to all platform technology (UCK9, Security) and Unified E-SRST feature licenses that the router uses. Unified E-SRST requires one license entitlement (SRST_E_EP) for each configured SIP or SCCP phone.

CSSM shows license usage across all registered devices to a virtual account. A Virtual Account License Inventory displays the quantity of licenses that are purchased, those licenses in use, and a balance. An **Insufficient Licenses** alert is displayed if the license balance is below 0.

For example, consider a smart account in CSSM with 50 SRST_E_EP licenses. If you have a single registered Unified E-SRST router with 20 configured phones, the CSSM licenses page shows **Purchased** as 50, **In Use** as 20 and **Balance** as 30.

For more information on Smart Software Manager, see the Cisco Smart Software Manager User Guide.

Note The SRST_E_EP license count reflects the total phone count for both the ephones and voice register Pools that are configured in the Unified E-SRST irrespective of registered or nonregistered phones. Reports license usage three minutes after the last configuration change, to avoid unnecessary reporting while configuring Unified E-SRST.

Note Unified E-SRST Smart Licenses also provide RTU entitlement for routers that are not configured for Smart Licensing.

Smart License Operation

Cisco IOS XE Everest 16.5.1 Release to Cisco IOS XE Fuji 16.9.1 Release

Cisco 4000 Series Integrated Services Routers support Smart Licensing as an alternative to Cisco Software RTU Licensing. Use the **license smart enable** command to enable Smart Licensing. To disable Smart Licensing, use the **no** form of the command and re-accept the EULA using the **license accept end user agreement** command.

Cisco IOS XE Gibraltar 16.10.1 Release Onwards

The Cisco RTU Licensing and the CLI **license smart enable** command are deprecated. Smart Licensing is mandatory from this release.

Cisco IOS XE Everest 16.5.1 Release to Cisco IOS XE Amsterdam 17.3.1a Release

Routers configured to use Smart Licensing offer a 90-day evaluation period, during which you can use all the features without registering to CSSM. A Unified E-SRST device is associated with CSSM using a registration token. You can obtain the registration token from the virtual CSSM account or from an on-premises satellite. Once registered, the evaluation period pauses and you can use the balance later. You cannot renew the evaluation period on its expiry.

Warning Unified E-SRST shuts down when the router is unregistered and allowed to pass into the Evaluation Expired state.

To register the Unified E-SRST router with CSSM, use **license smart register idtoken** command. For information on registering the device with CSSM, see Software Activation Configuration Guide.

Upon successful registration, the device sends an authorization request to CSSM for the licenses in use. For each license type requested, if the Smart Account has sufficient licenses, CSSM responds with **Authorized**. If the Smart Account does not have sufficient licenses, CSSM responds with **Out of Compliance**.

Post successful authorization of the request, licenses are bound to the requesting device until the next authorization request submission. An authorization request is sent every 30 days or when there is any change in license consumption, to maintain the registration with CSSM. The authorization expires if you do not update the license request for the router within 90 days. The certificate issued to identify the router at the time of registration is valid for one year and renewed every six months.

```
Router# show license summary
Smart Licensing is ENABLED
Registration:
Status: REGISTERED
Smart Account: ABC
Virtual Account: XYZ
Export-Controlled Functionality: Not Allowed
Last Renewal Attempt: None
```

```
Next Renewal Attempt: Jun 07 12:08:10 2017 UTC
License Authorization:
Status: AUTHORIZED
Last Communication Attempt: SUCCESS
Next Communication Attempt: Apr 13 07:11:48 2017 UTC
License Usage:
License                    Entitlement tag         Count     Status
-----------------------------------------------------------------------------
ISR_4351_UnifiedCommun.. (ISR_4351_UnifiedCommun..) 1       AUTHORIZED
SRST v12 Endpoint Li...  (SRST_EP)                  4       AUTHORIZED
```

Cisco IOS XE Gibraltar 16.12.1 Release to Cisco IOS XE Amsterdam 17.3.1a Release

Cisco 4000 Series Integrated Services Routers supports Specific License Reservation (SLR). SLR allows reservation and utilization of Cisco Smart Licenses without communicating the license information to CSSM. To reserve specific licenses for a device, generate the request code from the device. Enter the request code in CSSM along with the required licenses and their quantity, and generate authorization code. Enter the authorization code on the device to map the license to the Unique Device identifier (UDI).

Cisco IOS XE Amsterdam 17.3.2 and Cisco IOS XE Bengaluru 17.4.1a Release Onwards

This release introduces a new paradigm for tracking license usage across your business. In earlier releases, license authorization was forward looking, binding licenses to a device until the next authorization request. Actual license usage during the proceeding reporting period is sent to CSSM, allowing you to plan ongoing license requirements based on historical usage data. Initial device registration is no longer required to use most platform functionality and deprecates the evaluation period.

Submits the license usage reports periodically according to a minimum reporting policy set for your account. Typically, this period could be once per year. However, you can generate reports more frequently if the use of licensed features varies over time. CSSM acknowledges each Resource Utilization Monitoring (RUM) report to ensure reliable recording of the usage. If the router does not receive an acknowledgment within the minimum reporting period, disables the call processing. Resumes the call processing on receiving a valid acknowledgment.

Submit the reports directly to the CSSM or through a satellite. Cisco Smart Licensing Utility (CSLU) applications can also receive usage reports, providing you with more flexibility in managing your license usage. Also, when a device is not able to communicate directly with a licensing server, a signed usage report can be generated and manually uploaded to CSSM. The acknowledgment generated by CSSM must be uploaded to the device within the license reporting policy period to ensure continued use.

As license reporting is now based on historical usage, the registration process used previously has been replaced with a trust association that also defines the reporting policy set in your account. Establishing trust with CSSM or Cisco Smart Software Manager Satellite uses an identity token similar to earlier registrations. Use the **license smart trust idtoken** *token* command to establish the trust relationship within the initial reporting period set for the device. The CLI **license smart register** command is deprecated from this release.

Current license usage for Unified E-SRST is displayed using the **show license summary** command:

```
Router# sh license summary
                License Usage:

        License         Entitlement tag             Count     Status
        -----------------------------------------------------------------------------
        securityk9     (ISR_4400_Security)            1        IN USE
        AdvUCSuiteK9   (ISR_4400_AdvancedUCSuite)     1        IN USE
        uck9           (ISR_4400_UnifiedCommun...)    1        IN USE
        SRST_E_EP      (SRST_E_EP)                    8        IN USE
        SRST_EP        (SRST_EP)                      592      IN USE
```

Toll Fraud Prevention for SIP Line Side on Unified E-SRST

Unified E-SRST Release 12.6 enhances the existing Toll Fraud Prevention feature by enforcing security on the SIP line side of Unified E-SRST. The feature enhancement secures the Unified E-SRST system against potential toll fraud exploitation by unauthorized users from the SIP line side.

The configuration and characteristics of toll fraud prevention offered on the SIP line side of Unified E-SRST is same as the support available on Cisco Unified SRST. For more information on the feature, see Toll Fraud Prevention for SIP Line Side on Unified SRST.

Unified E-SRST with Support for Voice Hunt Group

The Unified E-SRST Release 12.2 supports the Voice Hunt Group with Cisco Unified Enhanced Survivable Remote Site Telephony (Unified E-SRST). The deployment supports the SIP and SCCP phones. The Cisco IP Phone 7800 and 8800 Series are the supported SIP phones for this deployment. The Unified E-SRST deployment introduces the voice hunt group enhancement on the Cisco 4000 Series Integrated Services Routers.

As part of the enhancement, supports the voice hunt group features in the E-SRST mode. The Unified E-SRST 12.2 and later releases supports the voice hunt group deployments with Sequential, Parallel, Longest idle, and Peer call blasting.

During a WAN outage, the SIP phones on the Cisco Unified Communications Manager (Cisco Unified Communications Manager) fallback to Unified E-SRST router in **mode esrst**. By default, logs the SIP phones in to the hunt group. However, if the CLI command **members logout** is configured under the voice hunt group configuration mode, the phones are in logged out state. In the Unified E-SRST mode, the phone that falls back on Unified E-SRST can toggle state. It can also log in (or log out) to the voice hunt group using HLog in Feature Access Code (FAC). Displays the DN status (logged in or logged out) on the registered phones with Unified E-SRST. The following FAC codes are available as part of the enhancement introduced on Unified E-SRST:

- FAC Standard (Code: *5)
- FAC Custom (Code: Customizable, with maximum character string length of 10. For example, *89, 8888888888)

When the user inputs FAC from a phone with multiple lines, the log out behavior is different across a deployment with the common voice register Pool configuration and the individual voice register Pool configuration.

- Common Voice Register Pool Configuration: The DN's log out individually, and not at the phone level.
- Individual Voice Register Pool Configuration: The DN's log out at the phone level, irrespective of the user providing the DN (primary, secondary, and so on) from which FAC input.

When the WAN is available, the phones register back with Cisco Unified Communications Manager. For a sample configuration of Unified E-SRST with voice hunt group enhancements, see Example for Configuring Unified E-SRST with Voice Hunt Group Enhancements.

The Unified E-SRST 12.2 Release introduces support for the voice hunt group with shared lines and mixed shared lines (SCCP and SIP phones). For a mixed shared line supported with the voice hunt group, configure only individual voice register Pools. Does not support the common voice register Pools. For a sample configuration of mixed shared lines configured for a voice hunt group on Unified E-SRST, see Example for Configuring Shared Line with Voice Hunt Group on Unified E-SRST.

Also, supports hunt statistic collection for Unified E-SRST 12.2 and later releases.

A mixed deployment of SIP and SCCP phones supports the Unified E-SRST, Release 12.2. Supports Hunt Group Logout from a mixed deployment of SIP and SCCP phones using:

- FAC
- Feature Button, or DND

Supports Line level logout and phone level log out using FAC (*4).

Note Does not support Hunt Group logout for shared lines. Shared lines retain their logged in status.

Support for B-ACD in Unified E-SRST

The Unified E-SRST Release 12.2 enhancement supports B-ACD. For SIP phones that fall back to Unified E-SRST router in **mode esrst**, you must ensure that the CLI command **members logout** is configured. The Members Logout functionality handles the login back from the phones using FAC. It also supports call Delivery to Voice Hunt Group from B-ACD.

For a sample configuration, see Example for Configuring B-ACD with Unified E-SRST.

Recommendations for Configuring Voice Hunt Group on Unified E-SRST

The Unified E-SRST Release with Support for voice hunt group has the following design characteristics:

- For all the directory numbers falling back from Cisco Unified Communications Manager, a common voice register Pool configuration and an individual voice register Pool configuration is supported for this deployment. An individual **voice register pool** configured with the CLI command **id device-id-name**, along with **voice register dn** configuration, is recommended.
- Ensure that the CLI command **mode esrst** is configured under **voice register global** configuration mode for phones to fall back to Unified E-SRST.
- Ensure that the CLI command **id ip** or **id device-id-name** is configured under **voice register pool** configuration mode, along with **voice register dn** configuration, for a deployment with individual voice register Pool configuration. For a sample configuration, see Example for Configuring Unified E-SRST with Voice Hunt Group Enhancements.
- Ensure that the CLI command **id device-id-name** is preferred over **id ip** as the CLI command to configure under **voice register pool** configuration mode. This scenario occurs where the IP address of the phone changes due to the DHCP configured on the phone.
- Ensure that the CLI command **id network** is configured under **voice register pool** configuration mode for a deployment with common voice register Pool configuration. The recommended configuration is **id network** *8.55.0.0 255.255.0.0* so as to facilitate registration of phones falling back on Unified E-SRST from Cisco Unified Communications Manager.

- Ensure that the CLI command **members logout** is configured under **voice hunt-group** configuration mode. The CLI is applied by default when the SIP phones fall back to Unified E-SRST from Cisco Unified Communications Manager.
- Ensure that the CLI command **fac standard** is configured under **telephony-service** configuration mode. If you want to configure a FAC code other than *5, you must configure the CLI command **fac custom** under **telephony-service** configuration mode.
- Ensure that the CLI commands **call-park system application** and**hunt-group logout hlog** are configured under **telephony-service** configuration mode. The CLI commands are mandatory configuration for FAC functionality to work.

For steps on configuring voice hunt groups on Unified E-SRST, see Configure Voice Hunt Groups on Unified E-SRST.

For a sample configuration of voice hunt groups on Unified E-SRST, see Example for Configuring Unified E-SRST with Voice Hunt Group Enhancements.

SIP: Configure Unified E-SRST

The Enhanced SRST for Cisco Unified SIP IP Phones feature supports version negotiation between the SIP phones and ESRST to enable more features in the Cisco Unified E-SRST mode. In the current scenario, when the SIP phones fall back to the SRST mode, disables features such as Shared-Line, Busy-Lamp-Field (BLF), and Video call on the phones. The SRST mode does not support these features. However, with the Enhanced Survivable Remote Site Telephony (E-SRST) deployment, you can enable the basic and supplementary call features. Also, you can enable the following features using version negotiation:

- Shared-Line
- Busy-Lamp-Field (BLF)
- Video Calls

The following table contains a list of supported features and the expected behavior of the features in the E-SRST mode.

Feature	Supported Features	Expected Behavior in the E-SRST Mode
Shared-Line	cBarge	Not Supported (After the failover, the phone does not retain the key.)
Privacy-on-hold	Supported	Supported
Transfer	Supported	
Conference	Supported	
BLF	BLF DN monitoring	
BLF device-based monitoring	Not supported (Not supported in RT phones)	
BLF call-list monitoring	Supported	

Feature	Supported Features	Expected Behavior in the E-SRST Mode
Monitoring of a Call-park slot	Not supported	
Monitoring of Paging DN	Not supported	
Monitoring of Conference DN	Not supported	

- To enable version negotiation feature between ESRST & phone, you must configure "mode esrst" under the voice register global mode.
- We recommended using the SRST manager to automate the CLI provisioning of ESRST branch routers.

For more information on SRST, see the Cisco Unified SRST Manager Administration Guide.

Restrictions

- Supports the Version Negotiation feature only on the Cisco Unified 9951, 9971, 8961 SIP IP phones, Cisco IP Phone 7800, and 8800 Series.
- The phone firmware version is version 9.4.1 or later versions.
- This feature supports video calls only between the local Cisco Unified SIP IP phones and the No Time-Division Multiplexing (TDM) video calls during the SRST failovers.
- To enable phone-specific features like shared-line & BLF work, configure the individual voice register Pools.

Enable the E-SRST Mode

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register global**
4. **mode esrst**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:**	Enters global configuration mode.

	Command or Action	Purpose
	`Router# configure terminal`	
Step 3	**voice register global** **Example:** `Router(config)# voice register global`	Enters the voice register global configuration mode to set the parameters for all the supported SIP phones in Cisco Unified Communications Manager Express.
Step 4	**mode esrst** **Example:** `Router(config-register-global)# mode esrst`	Configures the E-SRST mode under the voice register global mode.
Step 5	**exit** **Example:** `Router(config-register-global)# exit`	Exits the voice register-global configuration mode.

Configure SIP shared-line

To configure SIP shared-line, perform the following procedure:

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register dn** *dn-tag*
4. **shared-line** [**max-calls** *number-of-calls*]
5. **huntstop channel** *number-of-channels*
6. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice register dn** *dn-tag*	
Step 4	**shared-line** [**max-calls** *number-of-calls*]	
Step 5	**huntstop channel** *number-of-channels*	
Step 6	**end**	Returns to privileged EXEC mode.

Configure BLF

Before you begin

To enable the version negotiation feature in the Unified E-SRST mode, perform the following procedure.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **sip-ua**
4. **presence enable**
5. **exit**
6. **max-subscription** *number*
7. **presence call-list**
8. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**sip-ua**	
Step 4	**presence enable**	
Step 5	**exit**	
Step 6	**max-subscription** *number*	
Step 7	**presence call-list**	
Step 8	**end**	

Enable a SIP Directory Number to Be Watched

To enable a directory number to be watched, perform the following procedure:

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register dn** *dn-tag*
4. **number***number*

5. **allow watch**
6. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable**	
Step 2	**configure terminal**	
Step 3	**voice register dn** *dn-tag*	
Step 4	**number***number*	
Step 5	**allow watch**	
Step 6	**end**	

Enable BLF on a Voice Register Pool

To enable BLF on a **voice register pool**, perform the following steps:

For configuration information, see the Cisco Unified Communications Manager Administration Guide.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register pool** *pool-tag*
4. **number** *tag***dn** *dn-tag*]
5. **blf-speed-dial** *tag number***label***string***[device]**
6. **presence call-list**(To enable Presence feature for all the missed/received/placed calls)
7. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable**	
Step 2	**configure terminal**	
Step 3	**voice register pool** *pool-tag*	
Step 4	**number** *tag***dn** *dn-tag*]	
Step 5	**blf-speed-dial** *tag number***label***string***[device]**	
Step 6	**presence call-list**(To enable Presence feature for all the missed/received/placed calls)	
Step 7	**end**	

Example: ESRST Mode

The following example shows how to enable the E-SRST mode:

```
Router# configure terminal
Router(config)# voice register global
Router(config-register-global)# mode esrst
```

Example: Configuring Shared Line

The following example shows how to configure shared-line:

```
Router(config)#voice register dn 1
Router (config-register-dn)#number 1111
Router (config-register-dn)#shared-line max-calls 7

Router(config)#voice register pool 1
Router(config-register-pool)#Id mac 002D.264E.54FA
Router(config-register-pool)#type 9971
Router(config-register-pool)#number 1 dn 1

Router(config)#voice register pool 2
Router(config-register-pool)#id mac 000D.39F9.3A58
Router(config-register-pool)#type 7965
Router(config-register-pool)#number 1 dn 1
```

Example: Configuring BLF

The following example shows how to configure BLF:

```
Router(config)#voice register dn  1Router (config-register-dn)#number 1111Router
(config-register-dn)#allow watchRouter(config)#voice register dn  1Router
(config-register-dn)#number 2222Router(config)#voice register pool
1Router(config-register-pool)#id mac 0015.6247.EF90Router(config-register-pool)#type
7971Router(config-register-pool)#number 1 dn 1Router(config)#voice register pool
2Router(config-register-pool)#id mac 0012.0007.8D82Router(config-register-pool)#type
7912Router(config-register-pool)#number 1 dn 2Router(config-register-pool)#blf-speed-dial
1 1111 label "1111"
```

Note If the phone and the Unified E-SRST router are in different subnets and you are using **id mac** in the **voice register pool** configuration mode. Configure the digest credentials on Cisco Unified Communications Manager, and username password configuration under **voice register pool** on Unified E-SRST. Digest Configuration is not required with the **id device-id-name** CLI command in Cisco Unified SRST Release 12.2.

Configure Unified E-SRST

The **mode esrst** under **telephony-service** and **voice register global** configuration mode supports SCCP and SIP phones respectively to enable the enhanced services in Unified E-SRST mode. While Cisco Unified SRST supports only the basic voice hunt group features, Unified E-SRST supports the advanced voice hunt group features such as HLog, shared lines, and B-ACD. To configure the basic Unified E-SRST, perform the following procedure:

SUMMARY STEPS

1. **enable**
2. **configure terminal**

3. **telephony-service**
4. **mode esrst**
5. **max-ephones** *max-phones*
6. **max-dn***max-directory-numbers*
7. **ip source-address** *ip-address* [**port** *port*] [**any-match** | **strict-match**]
8. **call-park system {application |redirect}**
9. **hunt-group logout {DND | HLog}**
10. **transfer-system full-consult**
11. **transfer-pattern** *transfer-pattern*
12. **fac { standard | custom { alias** *alias-tag* | *feature* **} }**
13. **create cnf-files**
14. **exit**
15. **voice register global**
16. **mode esrst**
17. **max-dn** *max-directory-numbers*
18. **max-pool** *max-phones*
19. **exit**
20. **voice register dn** *dn-tag*
21. **number** *number*
22. **exit**
23. **voice register pool** *pool-tag*
24. **id** [{**network** *address* **mask** *mask* | **ip** *address* **mask** *mask* | **mac** *address*}] [**device-id-name** *devicename*]
25. **dtmf-relay rtp-nte**
26. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**telephony-service** **Example:** `Router(config)# telephony-service`	Enters telephony-service configuration mode.
Step 4	**mode esrst** **Example:** `Router(config)# telephony-service`	Configures the E-SRST mode under the telephony-service configuration mode.

	Command or Action	Purpose
Step 5	**max-ephones** *max-phones* **Example:** `Router(config-telephony)# max-ephones 40`	Configures the maximum supported IP phones by the router. The default is 0. The maximum number is platform-dependent.
Step 6	**max-dn***max-directory-numbers* **Example:** `Router(config-telephony)# max-dn 15`	Sets the maximum supported directory numbers (DNs) by the router. • Max-directory-numbers: Maximum supported directory numbers (DNS) or virtual voice ports by the router. The maximum number is platform-dependent. The default is 0.
Step 7	**ip source-address** *ip-address* [**port** *port*] [**any-match** | **strict-match**] **Example:** `Router(config-telephony)# ip source-address 8.39.23.24 port 2000`	Enables the router to receive messages from the Cisco IP phones through the specified IP addresses and supports strict IP address verification. The default port number is 2000.

Step 8	**call-park system {application |redirect}**
	Example:
	`Router(config-telephony)# call-park system application`
		Defines system parameters for the Call Park feature.
		• **application** : Enables the Call Park features supported in Cisco Unified SRST.

Step 9	**hunt-group logout {DND | HLog}**
	Example:
	`Router(config-telephony)# hunt-group logout HLog`
		Sets the hunt-group logout options with Hlog in telephony-service configuration mode.

Step 10	**transfer-system full-consult** **Example:** `Router(config-telephony)# transfer-system full-consult`	Specifies the Call Transfer method. • **full-consult**—Calls are transferred with consultation using H.450.2 standard methods and a second phone line if available. Calls fall back to full-blind if the second line is unavailable.
Step 11	**transfer-pattern** *transfer-pattern* **Example:** `Router(config-telephony)# transfer-pattern .T`	Allows transfer of the phone calls by Cisco Unified IP phones to specified phone number patterns. If you have set no transfer pattern, defaults to other local IP phones. • *transfer-pattern*—A string of digits for permitted Call Transfers.

Step 12	**fac { standard | custom { alias** *alias-tag* | *feature* **} }**
	Example:
	`Router(config-telephony)# fac standard`
		Enables all standard feature access codes (FACs) or creates and enables individual custom FACs in telephony-service configuration mode.

Step 13	**create cnf-files** **Example:**	Builds the required XML configuration files for IP phones in the telephony-service configuration mode.

	Command or Action	Purpose
	`Router(config-telephony)# create cnf-files version-stamp`	
Step 14	**exit** **Example:** `Router(config-telephony)# exit`	Exits the telephony-service configuration mode
Step 15	**voice register global** **Example:** `Router(config)# voice register global`	Enter the voice register global configuration mode.
Step 16	**mode esrst** **Example:** `Router(config-register-global)# mode esrst`	Configures the E-SRST mode under the voice register global mode.
Step 17	**max-dn** *max-directory-numbers* **Example:** `Router(config-register-global)# max-dn 40`	Set the maximum supported SIP phone directory numbers (extensions) by a Cisco router in the voice register global configuration mode.
Step 18	**max-pool** *max-phones* **Example:** `Router(config-register-global)# max-pool 40`	Sets maximum supported SIP phones by the Cisco Unified SRST router. • Version- and platform-dependent; type? For range.
Step 19	**exit** **Example:** `Router(config-register-global)# exit`	Exits the voice register global configuration mode.
Step 20	**voice register dn** *dn-tag* **Example:** `Router(config)# voice register dn 17`	Enter the voice register directory number configuration mode to define a directory number for a SIP phone. Use the same directory number (DN) configured in Cisco Unified Communications Manager to configure the voice register directory number in Unified E-SRST.
Step 21	**number** *number* **Example:** `Router(config-register-dn)# number 7001`	Defines a valid number for a directory number.
Step 22	**exit** **Example:** `Router(config-register-dn)# exit`	Exits the voice register directory number configuration mode.
Step 23	**voice register pool** *pool-tag* **Example:** `Router(config)# voice register pool 1`	Enters the voice register Pool configuration mode to set phone-specific parameters for a SIP phone.

	Command or Action	Purpose
Step 24	**id** [{**network** *address* **mask** *mask* | **ip** *address* **mask** *mask* | **mac** *address*}] [**device-id-name** *devicename*] **Example:** `Router(config-register-pool)# id network 8.55.0.0 mask 255.255.0.0`	Explicitly identifies a locally available individual or set of SIP IP phones. The keywords and arguments are defined as follows: • **network** *address* **mask** *mask*: The **network** *address* mask *mask* keyword/argument combination is used to accept SIP Register messages for the indicated phone numbers from any IP phone within the indicated IP subnet. • **ip***address* **mask***mask* : The **ip** *address* **mask** *mask* keyword/argument combination is used to identify an individual phone. • **mac***address* : MAC address of a particular Cisco Unified IP Phone. • **device-id-name***devicename* : Defines the device name to be used to download the phone's configuration file.
Step 25	**dtmf-relay rtp-nte** **Example:** `Router(config-register-pool)# dtmf-relay rtp-nte`	Forwards DTMF tones by using Real-Time Transport Protocol (RTP) with the Named phone Event (NTE) payload type and enables the DTMF relay using the RFC 2833 standard method.
Step 26	**exit** **Example:** `Router(config-register-pool)# exit`	Exits the voice register Pool configuration mode.

Configure Voice Hunt Groups on Unified E-SRST

To configure Voice Hunt Group feature on Unified E-SRST, perform the following procedure:

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice hunt-group** *hunt-tag* **{longest-idle | parallel | peer | sequential}**
4. **members logout**
5. **list** *number [, number...]*
6. **timeout** *seconds*
7. **statistics collect**
8. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.

Step 3	**voice hunt-group** *hunt-tag* **{longest-idle | parallel | peer | sequential}**
	Example:
	`Router(config)# voice hunt-group 1 sequential`
		Enters voice hunt-group configuration mode to define a hunt group.
		• Hunt-tag—Unique sequence number for configuring the hunt group. Range is 1–100.
		• Longest idle—Hunt group in which calls go to the directory number that has been idle for the longest time.
		• Sequential—Hunt group in which directory numbers ring in the order in which they are listed, left to right.
		• Parallel—Hunt group in which all directory numbers ring simultaneously.
		• Peer—Hunt group in which the call placed to a directory number rings for the next directory number in line.

Step 4	**members logout** **Example:** `Router(config-voice-hunt-group)# members logout`	(optional) Configures a Cisco Unified SRST system for all non-shared static members or agents in a voice hunt group with the Hlogout initial state.
Step 5	**list** *number [, number...]* **Example:** `Router(config-voice-hunt-group)# list 1812, 1813, 1814`	Defines a list of extensions that are members of a voice hunt group.
Step 6	**timeout** *seconds* **Example:** `Router(config-voice-hunt-group)# timeout 30`	Defines the number of seconds after which directs the unanswered calls to the next number in a voice hunt-group list.
Step 7	**statistics collect** **Example:** `Router(config-voice-hunt-group)# statistics collect`	Enables the collection of call statistics for a voice hunt group.
Step 8	**exit** **Example:**	Exits the voice hunt group configuration mode.

	Command or Action	Purpose
	Router(config-voice-hunt-group)# exit	

Example for Configuring Unified E-SRST with Voice Hunt Group Enhancements

The following is a sample configuration for Unified E-SRST Release 12.2 under **telephony-service**, **voice register global**,**voice register pool**, and **voice hunt-group** configuration modes, for a deployment with common voice register Pool configuration.

```
Router#
telephony-service
call-park system application
hunt-group logout HLog
transfer-system full-consult
fac standard
```

```
Router#sh run | sec global
voice register global
mode esrst
max-dn 40
max-pool 40
```

```
Router#
voice register pool 1
id network 8.55.0.0 mask 255.255.0.0
dtmf-relay rtp-nte
Router#
telephony-service
max-ephones 40
max-dn 50
ip source-address 8.39.23.24 port 2000
call-park system application
transfer-system full-consult
transfer-pattern .T
fac standard
create cnf-files version-stamp Jan 01 2002 00:00:00
```

```
Router#sh run | sec hunt
voice hunt-group 1 sequential
members logout
list 1812,1813,1814
timeout 30
statistics collect
pilot 1111
```

The following is a sample configuration for Unified E-SRST Release 12.2, for a deployment with individual voice register Pool configuration, with the CLI command **id ip** configured.

```
voice register dn 2
number 4000
!
voice register dn 3
number 4002
!
voice register pool 2
busy-trigger-per-button 2
id ip 8.55.0.241 mask 255.255.0.0
type 8811
number 1 dn 2
dtmf-relay rtp-nte
codec g711ulaw
!
```

```
voice register pool 3
busy-trigger-per-button 2
id ip 8.55.0.242 mask 255.255.0.0
type 7861
number 1 dn 3
dtmf-relay rtp-nte
codec g711ulaw
```

The following is a sample configuration for Unified E-SRST Release 12.2, for a deployment with individual voice register Pool configuration, with the CLI command **id device-id-name** configured.

```
voice register dn 2
number 4000
!
voice register dn 3
number 4002
!
voice register pool 2
busy-trigger-per-button 2
id device-id-name SEP00EBD5CD77ED
type 8811
number 1 dn 2
dtmf-relay rtp-nte
codec g711u;aw
voice register pool 3
busy-trigger-per-button 2
id device-id-name SEP0076861A7EDC
type 7861
number 1 dn 3
dtmf-relay rtp-nte
codec g71ulaw
```

Example for Configuring B-ACD with Unified E-SRST

The following is a sample configuration for B-ACD functionality supported with Unified E-SRST:

```
application
service aa-bcd bootflash:/app-b-acd-aa-3.0.0.4_thd_v4.tcl
paramspace english index 0
param second-greeting-time 60
param welcome-prompt _bacd_welcome.au
param call-retry-timer 8
param voice-mail 1811
paramspace english language en
param max-time-call-retry 16param service-name callq
param number-of-hunt-grps 2
param handoff-string aa-bcd
paramspace english location flash:
param max-time-vm-retry 2
param aa-pilot 1117
!
service clid_col_npw_npw
param uid-length 4
!
service aa-ccd bootflash:/app-b-acd-aa-3.0.0.4_thd_v4.tcl
paramspace english index 0
param drop-through-prompt _bacd_welcome.au
param second-greeting-time 60
paramspace english language en
param call-retry-timer 8
param voice-mail 1811
param max-time-call-retry 16
param service-name callq
param number-of-hunt-grps 1
param drop-through-option 1
paramspace english location flash:
param handoff-string aa-ccd
param max-time-vm-retry 2
param aa-pilot 1118
!
service callq bootflash:/imanage-b-acd-3.0.0.4_Q60.tcl
param queue-len 1
param aa-hunt1 1111
param number-of-hunt-grps 4
param queue-manager-debugs 1
!
call-park system application
```

Example for Configuring Shared Line with Voice Hunt Group on Unified E-SRST

The following is a sample configuration of Unified E-SRST, Release 12.2 with support for mixed shared lines (SIP and SCCP Phones) in a voice hunt group deployment.

```
Router# sh run | sec global
voice register global
mode esrst
no allow-hash-in-dn
max-dn 40
max-pool 40
Router# sh run | sec pool
max-pool 40
voice register pool 1
busy-trigger-per-button 2
id device-id-name SEP00CCFC4AA4DC
type 8811
number 1 dn 1
number 2 dn 21
dtmf-relay rtp-nte
username xxxx password uvwx
codec g711ulaw
no vad
voice register pool 2
busy-trigger-per-button 2
id device-id-name SEP00CCFC177A4E
type 8841
number 1 dn 2
dtmf-relay rtp-nte
username xxxx password uvwx
codec g711ulaw
no vad
voice register pool 3
busy-trigger-per-button 2
id device-id-name SEP0076861ADEF0
type 7841
number 1 dn 3
number 2 dn 22
dtmf-relay rtp-nte
username xxxx password uvwx
codec g711ulaw
no vad
voice register pool 4
busy-trigger-per-button 2
id device-id-name SEP00EBD5CD270C
type 8811
number 1 dn 4
number 2 dn 22
```

```
dtmf-relay rtp-nte
username xxxx password uvwx
codec g711ulaw
no vad
voice register pool 5
busy-trigger-per-button 2
id device-id-name SEP94D4692A2553
type 8841
number 1 dn 5
dtmf-relay rtp-nte
username xxxx password uvwx
codec g711ulaw
no vad
voice register pool 6
busy-trigger-per-button 2
id device-id-name SEP00CAE540C4B5
type 8811
number 1 dn 6
number 2 dn 21
dtmf-relay rtp-nte
username xxxx password uvwx
codec g711ulaw
no vad
alias exec pool show voice register pool all br
```

```
Router# sh run | sec dn
no allow-hash-in-dn
max-dn 40
voice register dn 1
voice-hunt-groups login
number 1811
voice register dn 2
voice-hunt-groups login
number 1812
voice register dn 3
voice-hunt-groups login
number 1813
voice register dn 4
voice-hunt-groups login
number 1814
voice register dn 5
voice-hunt-groups login
number 1815
voice register dn 6
voice-hunt-groups login
number 1816
voice register dn 21
voice-hunt-groups login
number 1821
shared-line
voice register dn 22
voice-hunt-groups login
number 1822
shared-line
```

```
Router# sh run | sec ephone
max-ephones 40
ephone-dn 11
number 1911
ephone-dn 12
number 1912
ephone-dn 13
number 1913
ephone-dn 14
number 1914
```

```
ephone-dn 21
number 1921
ephone-dn 22
number 1822
shared-line sip
ephone 11
device-security-mode none
mac-address 1111.1111.1911
feature-button 1 HLog
type 7970
button 1:11
ephone 12
device-security-mode none
mac-address 1111.1111.1912
feature-button 1 HLog
type 7970
button 1:12 2:21
ephone 13
device-security-mode none
mac-address 1111.1111.1913
feature-button 1 HLog
type 7970
button 1:13 2:21
ephone 14
device-security-mode none
mac-address 1111.1111.1914
feature-button 1 HLog
type 7970
button 1:14 2:22
alias ephone show ephone summary brief
alias exec ephone show ephone summary brief
```

```
Router# sh run | sec tele
telephony-service
conference transfer-pattern
mode esrst
max-ephones 40
max-dn 50
ip source-address 8.39.23.24 port 2000
service phone sshAccess 0
service phone webAccess 0
max-conferences 8 gain -6
call-park system application
hunt-group logout HLog
transfer-system full-consult
fac standard
```

SCCP: Configure Unified E-SRST

You need to configure mode esrst under telephony-service to enable ESRST mode for SCCP Phones.

Before you begin

To enable the version negotiation feature in the Unified E-SRST mode, perform the following procedure.

- Cisco Unified Communications Manager Express 10.5 or later version
- Configure the telephony-services command.

Note For SCCP phones, CME-as-SRST mode is provisioned using the SRST mode autoprovision command. From 10.5 release onwards, deprecates this command. When you try to configure CME-as-SRST mode, displays the following message: *“Note: This configuration is being deprecated. Please configure "mode esrst" to use the enhanced SRST mode.”*

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **telephony-service**
4. **mode esrst**
5. **max-ephones***max-phones*
6. **max-dn max-directory-numbers [preference preference-order] [no-reg primary | both]**
7. **ip source-address** *ip-address [port port] [any-match | strict-match]*
8. **exit**
9. **ephone-dn dn-tag** *[dual-line]*
10. **number** *number [secondary number] [no-reg [both |primary]]*
11. (Optional) **name***name*
12. **exit**
13. **ephone phone-tag**
14. **mac-address***[mac-address]*
15. **type** *phone-type [addon 1 module-type [2 module-type]]*
16. **button button-number{separator}dn-tag** *[,dn-tag...][button-number{x}overlay-button-number] [button-number...]*
17. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**telephony-service** **Example:** `Router(config)# telephony-service`	Enters telephony-service configuration mode.
Step 4	**mode esrst** **Example:** `Router(config-telephony)# mode esrst`	Enters telephony-service configuration mode.

	Command or Action	Purpose
Step 5	**max-ephones***max-phones* **Example:** `Router(config-telephony)# max-ephones 24`	Enters telephony-service configuration mode.
Step 6	**max-dn max-directory-numbers [preference preference-order] [no-reg primary | both]** **Example:** `Router(config-telephony)# max-dn 24 no-reg primary`	Limits the number of directory numbers supported by this router. • Maximum number is the platform and version-specific. Type? For value.
Step 7	**ip source-address** *ip-address [port port] [any-match | strict-match]* **Example:** `Router(config-telephony)# ip source-address 192.168.11.1 port 2000`	Identifies the IP address and port number that the Cisco Unified SRST router uses for IP phone registration. • port port—(Optional) TCP/IP port number to use for SCCP. Range is 2000–9999. Default is 2000. • Any-match—(Optional) Disables the strict IP address checking for registration. It is the default setting. • Strict-match—(Optional) Instructs the router to reject IP phone registration attempts if the IP server address used by the phone does not exactly match the source address.
Step 8	**exit** **Example:** `Router(config-telephony)# exit`	Exits telephony-service configuration mode.
Step 9	**ephone-dn dn-tag** *[dual-line]* **Example:** `Router(config)# ephone-dn 1`	Enters ephone dn configuration mode to define a directory number for an IP phone, intercom line, voice port, or a message-waiting indicator (MWI). • Dn-tag—Identifies a particular directory number during configuration tasks. Range is 1 to the maximum number of directory numbers allowed on the router platform. Type? To display range.
Step 10	**number** *number [secondary number] [no-reg [both |primary]]* **Example:** `Router(config-ephone-dn)# number 1001`	Associates an extension number with this directory number. • Number—String of up to 16 digits that represents an extension or E.164 phone number.
Step 11	(Optional) **name***name* **Example:** `Router(config-ephone-dn)# name Smith, John`	Associates a name with this directory number. • Uses the Name for caller-ID displays and in the local directory listings. • Follows the name order in the directory command.

	Command or Action	Purpose
Step 12	**exit** **Example:** `Router(config-telephony)# end`	Exits ephone-dn configuration mode.
Step 13	**ephone phone-tag** **Example:** `Router(config)# ephone 1`	Enters ephone configuration mode to set ephone specific parameters. • Phone-tag—Unique sequence number that identifies the phone. Range is version and platform-dependent; type? To display range.
Step 14	**mac-address***[mac-address]* **Example:** `Router(config-ephone)# mac-address 0022.555e.00f1`	Associates the MAC address of a Cisco IP phone with an ephone configuration in a Unified E-SRST system. • Mac-address—Identifying MAC address of an IP phone found on a sticker on the bottom of the phone.
Step 15	**type** *phone-type [addon 1 module-type [2 module-type]]* **Example:** `Router(config-ephone)# type 7960`	Specifies the type of phone.
Step 16	**button button-number{separator}dn-tag** *[,dn-tag...][button-number{x}overlay-button-number] [button-number...]* **Example:** `Router(config-ephone)# button 1:7`	Associates a button number and line characteristics with an ephone-dn. Determines the maximum number of buttons by phone type.

Step 17	**end**
	Example:
	`Router(config-telephony)# end`
		Returns to privileged EXEC mode.

Example

The following example shows the status of the device in E-SRST mode:

```
show telephony-service
CONFIG (Version=10.5)
=====================
Version 10.5
Max phoneload sccp version 17
Max dspfarm sccp version 18
Cisco Unified Enhanced SRST
```

Note For SCCP phones, switching the mode from CME to ESRST and vice versa, results in wiping out the entire CME or ESRST configurations (including ephone, DNs, templates etc.).

Configure Mixed Shared Lines with SCCP Phones

To configure mixed shared lines between SCCP and SIP IP Phones on Unified E-SRST, perform the following procedure:

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **ephone-dn** *dn-tag [dual-line]*
4. **number** [**secondary** *[number]* [**no-reg** [**both**|**primary**]]
5. **shared-line sip**
6. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**ephone-dn** *dn-tag [dual-line]* **Example:** `Router(config)# ephone-dn 1`	Enters ephone dn configuration mode to define a directory number for an IP phone, intercom line, voice port, or a message-waiting indicator (MWI). • Dn-tag—Identifies a particular directory number during configuration task. Range is 1 to the maximum number of directory numbers allowed on the router platform. Type? To display the range.
Step 4	**number** [**secondary** *[number]* [**no-reg** [**both**|**primary**]] **Example:** `Router(config-ephone-dn)# number 1001`	Associates an extension number with this directory number. • number—String of up to 16 digits that represents an extension or E.164 phone number.
Step 5	**shared-line sip** **Example:** `Router(config-ephone-dn)# shared-line sip`	Adds an ephone-dn as a member of a shared directory number for a mixed shared line between Unified SIP and Unified SCCP IP phones.

Step 6	**end**
	Example:
	`Router(config-ephone-dn)# end`
		Returns to privileged EXEC mode.

Configure BLF for SCCP Phones

Before you begin

To enable the version negotiation feature in the Unified E-SRST mode, perform the following procedure.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **presence**
4. **max-subscription***number*
5. **presence call-list**
6. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**presence**	
Step 4	**max-subscription***number*	
Step 5	**presence call-list**	(To enable Presence feature for all the missed or received or placed calls)
Step 6	**end**	

Enable an SCCP Directory Number to Be Watched

To enable a directory number to be watched, perform the following procedure:

SUMMARY STEPS

1. **ephone-dn***dn-tag*
2. **number***number*
3. **allow watch**
4. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**ephone-dn***dn-tag*	
Step 2	**number***number*	
Step 3	**allow watch**	
Step 4	**end**	

Enable BLF on an Ephone

To enable BLF on an **ephone**, perform the following steps:

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **ephone***ephone-tag*
4. **button***button-number{separator}dn-tag [,dn-tag...] [button-number{x}overlay-button-number][button-number...]*
5. **blf-speed-dial** *tag number* **label** *string* **[device]**
6. **presence call-list**(To enable Presence feature for all the missed/received/placed calls)
7. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable**	
Step 2	**configure terminal**	
Step 3	**ephone***ephone-tag*	
Step 4	**button***button-number{separator}dn-tag [,dn-tag...] [button-number{x}overlay-button-number][button-number...]*	
Step 5	**blf-speed-dial** *tag number* **label** *string* **[device]**	
Step 6	**presence call-list**(To enable Presence feature for all the missed/received/placed calls)	
Step 7	**end**	

Configure Digest Credentials on Cisco Unified Communications Manager

To configure the username and password with Digest Authentication on Cisco Unified Communications Manager, perform the following steps:

SUMMARY STEPS

1. Log in to Cisco Unified Communications Manager.
2. Go to **System**>**Security**->**Phone Security Profile.**
3. Go to **User Management** > **End User.**
4. Go to the **Phone Settings** page and associate the user in the **Digest User** field.

DETAILED STEPS

	Command or Action	Purpose
Step 1	Log in to Cisco Unified Communications Manager.	
Step 2	Go to **System**>**Security**->**Phone Security Profile.**	
Step 3	Go to **User Management** > **End User.**	
Step 4	Go to the **Phone Settings** page and associate the user in the **Digest User** field.	

Configure Digest Credentials on Unified E-SRST for SIP

To configure credentials under a specific voice register pool, perform the following procedure:

Note Digest authentication does not work with 'id network' configuration in 'voice register pool'. It requires 'id device-id-name' or 'id Mac' configuration for individual pools. Also DN association on 'voice register pool' is required.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register pool** *<pool-tag>*
4. **username** *<username>* **password** *<password>*
5. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice register pool** *<pool-tag>*	

	Command or Action	Purpose
Step 4	**username** <*username*> **password** <*password*>	
Step 5	**end**	

Example: Configuring Digest Credentials on ESRST

The following example shows how to configure digest credentials on ESRST:

```
Router# conf terminal
Router(config)#voice register pool 10
Router (config-register-pool)# username abc password xyz
```

Configure Digest Credentials on Unified E-SRST for SCCP

To configure credentials under a specific ephone, perform the following procedure:

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **ephone** *ephone tag*
4. **username** <*username*> **password** <*password*>
5. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**ephone** *ephone tag*	
Step 4	**username** <*username*> **password** <*password*>	
Step 5	**end**	

CHAPTER 5

Setting Up the Network

This chapter describes how to configure your Cisco Unified Survivable Remote Site Telephony (SRST) router to run DHCP and to communicate with the IP phones during Cisco Unified Communications Manager fallback.

Information About Setting Up the Network

When the WAN link fails, the Cisco Unified IP Phones detect that they are no longer receiving keepalive packets from Cisco Unified Communications Manager. The Cisco Unified IP Phones then register with the router. The Cisco Unified SRST software is automatically activated and builds a local database of all Cisco Unified IP Phones attached to it (up to its configured maximum). The IP phones are configured to query the router as a backup call-processing source when the central Cisco Unified Communications Manager does not acknowledge keepalive packets. The Cisco Unified SRST router now performs call setup and processing, call maintenance, and call termination.

Cisco Unified Communications Manager uses DHCP to provide Cisco Unified IP Phones with the IP address of Cisco Unified Communications Manager. In a remote branch office, DHCP service is provided either by the SRST router itself or through the Cisco Unified SRST router using DHCP relay. Configuring DHCP is one of two main tasks in setting up network communication. The other task is configuring the Cisco Unified SRST router to receive messages from the Cisco IP phones through the specified IP addresses. Keepalive intervals are also set now.

MGCP Gateways and SRST

MGCP fallback is a different feature than SRST and, when configured as an individual feature, can be used by a PSTN gateway. To use SRST as your fallback mode on an MGCP gateway, SRST and MGCP fallback must both be configured on the same gateway. MGCP and SRST have had the capability to be configured on the same gateway since Cisco IOS Release 12.2(11)T.

To make outbound calls while in SRST mode on your MGCP gateway, two fallback commands must be configured on the MGCP gateway. These two commands allow SRST to assume control over the voice port and over call processing on the MGCP gateway. With Cisco IOS earlier than 12.3(14)T, the two commands are the **ccm-manager fallback-mgcp** and **call application alternate**commands. With Cisco IOS releases after 12.3(14)T, the **ccm-manager fallback-mgcp** and **service** commands must be configured. A complete configuration for these commands is shown in the section the Enabling Cisco Unified SRST on an MGCP Gateway section.

Note The commands listed above are ineffective unless both commands are configured. For instance, your configuration will not work if you only configure the **ccm-manager fallback-mgcp** command.

For more information on the fallback methods for MGCP gateways, see Configuring MGCP Gateway Support for Cisco Unified Communications Manager document or the MGCP Gateway Fallback Transition to Default H.323 Session Application document.

How to Set Up the Network

Enabling Cisco Unified SRST on an MGCP Gateway

To use SRST as your fallback mode with an MGCP gateway, SRST and MGCP fallback must both be configured on the same gateway. The configuration in the following section allows SRST to assume control over the voice port and over call processing on the MGCP gateway. Due to command changes that were made in Cisco IOS Release 12.3(14)T, use the configuration task that corresponds with the Cisco IOS Release you have installed.

Note The commands in the configuration section are ineffective unless both commands are configured. For instance, your configuration will not work if you only configure the **ccm-manager fallback-mgcp** command.

Note When an MGCP-controlled PRI goes into SRST mode, do not make or save configuration changes to the NVRAM on the router. If configuration changes are made and saved in SRST mode, the MGCP-controlled PRI fails when normal MGCP operation is restored.

Configuring Cisco Unified SRST on an MGCP Gateway Before Cisco IOS Release 12.3(14)T

Perform this task to enable SRST on an MGCP Gateway if you are using software release before Cisco IOS Release 12.3(14)T.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **ccm-manager fallback-mgcp**
4. **call application alternate** [*application-name*] OR **service** [**alternate** |**default**] *service-name location*
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable**	Enables privileged EXEC mode.

	Command or Action	Purpose
	Example: `Router> enable`	• Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**ccm-manager fallback-mgcp** **Example:** `Router(config)# ccm-manager fallback-mgcp`	Enables the gateway fallback feature and allows an MGCP voice gateway to provide call processing services through SRST or other configured applications when Cisco Unified Communications Manager is unavailable.

Step 4	**call application alternate** [*application-name*] OR **service** [**alternate** |**default**] *service-name location*
	Example:
	`Router(config)# call application alternate`
	OR
	`Router(config)# service default`
		The **call application alternate** command specifies that the default voice application takes over if the MGCP application is not available. The *application-name* argument is optional and indicates the name of the specific voice application to use if the application in the dial peer fails. If a specific application name is not entered, the gateway uses the default application.
		OR
		The service command loads and configures a specific, standalone application on a dial peer. The keywords and arguments are as follows:
		• Alternate (Optional). Alternate service to use if the service configured on the dial peer fails.
		• Default (Optional). Specifies that the default service **DEFAULT** on the dial peer is used if the alternate service fails.
		• Service-name: Name that identifies the voice application.
		• Location: Directory and filename of the Tcl script or VoiceXML document in URL format. For example, flash memory `flash:filename` , a TFTP `tftp://../filename`, or an HTTP server `http://../filename` are valid locations.

Step 5	**exit** **Example:** `Router(config)# exit`	Exits global configuration mode and returns to privileged EXEC mode.

Configuring SRST on an MGCP Gateway Using Cisco IOS Release 12.3(14)T or Later Releases

Perform this task to enable SRST on an MGCP Gateway if you are using Cisco IOS Release 12.3(14)T or later version.

Before you begin

Effective with Cisco IOS Release 12.3(14)T, the call application alternate command is replaced by the service command. The service command can be used in all releases after Cisco IOS Release 12.3(14)T.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **ccm-manager fallback-mgcp**
4. **application** [*application-name*]
5. **global**
6. **service**[*alternate* | *default*] *service-name location*
7. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**ccm-manager fallback-mgcp** **Example:** `Router(config)# ccm-manager fallback-mgcp`	Enables the gateway fallback feature and allows an MGCP voice gateway to provide call processing services through SRST or other configured applications when Cisco Unified Communications Manager is unavailable.
Step 4	**application** [*application-name*] **Example:** `Router(config) application app-xfer`	The **application-name** argument is optional and indicates the name of the specific voice application to use if the application in the dial peer fails. If a specific application name is not entered, the gateway uses the **DEFAULT** application.
Step 5	**global** **Example:** `Router(config)# global`	Enters global configuration mode.
Step 6	**service**[*alternate* | *default*] *service-name location* **Example:** `Router(config) service myapp https://myserver/myfile.vxml`	Loads and configures a specific, standalone application on a dial peer. • Alternate (Optional). Alternate service to use if the service configured on the dial peer fails. • Default (Optional). Specifies that the default service **DEFAULT** on the dial peer is used if the alternate service fails.

	Command or Action	Purpose
		• Service-name: Name that identifies the voice application. • Location: Directory and filename of the Tcl script or VoiceXML document in URL format. For example, flash memory `flash:filename` , a TFTP `tftp://../filename`, or an HTTP server `http://../filename` are valid locations.
Step 7	**exit** **Example:** `Router(config)# exit`	Exits global configuration mode and returns to privileged EXEC mode.

Configuration Example of Enabling SRST on a MGCP Gateway using Cisco IOS Release 12.3(14)T

The following is an example of configuring SRST on an MGCP Gateway if you are using Cisco IOS Release 12.3(14)T or later release:

```
isdn switch-type primary-net5
!
!
ccm-manager fallback-mgcp
ccm-manager mgcp
ccm-manager config
mta receive maximum-recipients 0
!
controller E1 1/0
pri-group timeslots 1-12,16 service mgcp
!
controller E1 1/1
!
!
!
interface Ethernet0/0
ip address 10.48.80.9 255.255.255.0
half-duplex
!
interface Serial1/0:15
no ip address
no logging event link-status
isdn switch-type primary-net5
isdn incoming-voice voice
isdn bind-l3 ccm-manager
no cdp enable
!
!
!
call rsvp-sync
!
call application alternate DEFAULT
!--- For Cisco IOS® Software Release 12.3(14)T or later,
this command was replaced by the service command
in global application configuration mode.
application
global
service alternate Default
!
```

```
voice-port 1/0:15
!
mgcp
mgcp dtmf-relay voip codec all mode cisco
mgcp package-capability rtp-package
mgcp sdp simple
!
mgcp profile default
!
!
!
dial-peer cor custom
!
!
!
dial-peer voice 10 pots
application mgcpapp
incoming called-number
destination-pattern 9T
direct-inward-dial
port 1/0:15
!
!
call-manager-fallback
limit-dn 7960 2
ip source-address 10.48.80.9 port 2000
max-ephones 10
max-dn 32
dialplan-pattern 1 704.... extension-length 4
keepalive 20
default-destination 5002
alias 1 5003 to 5002
call-forward busy 5002
call-forward noan 5002 timeout 12
time-format 24
!
!
line con 0
exec-timeout 0 0
line aux
```

Configuring DHCP for Cisco Unified SRST Phones

To perform this task, you must have your network configured with DHCP. For further details about DHCP configuration, see the Cisco IOS DHCP Server document and see your Cisco Unified Communications Manager documentation.

When a Cisco IP phone is connected to the Cisco Unified SRST system, it automatically queries for a DHCP server. The DHCP server responds by assigning an IP address to the Cisco IP phone and providing the IP address of the TFTP server through DHCP option 150. Then, the phone registers with the Cisco Unified Communications Manager system server and attempts to get configuration and phone firmware files from the Cisco Unified Communications Manager TFTP server address provided by the DHCP server.

When setting up your network, configure your DHCP server local to your site. You may use your SRST router to provide DHCP service (recommended). If your DHCP server is across the WAN and there is an extended WAN outage, the DHCP lease times on your Cisco Unified IP Phones may expire. This may cause your phones to lose their IP addresses, resulting in a loss of service. Rebooting your phones when there is no DHCP server available after the DHCP lease has expired will not reactivate the phones, because they will be unable to obtain an IP address or other configuration information. Having your DHCP server local to your remote site ensures that the phones can continue to renew their IP address leases in the event of an extended WAN failure.

Choose one of the following tasks to set up DHCP service for your Cisco UnifiedIP Phones:

- Defining a Single DHCP IP Address Pool, on page 159—Use this method if the Cisco Unified SRST router is a DHCP server and if you can use a single shared address pool for all your DHCP clients.
- Defining a Separate DHCP IP Address Pool for Each Cisco Unified IP Phone, on page 160—Use this method if the Cisco Unified SRST router is a DHCP server and you need separate pools for non-IP-phone DHCP clients.
- Defining the DHCP Relay Server, on page 161—Use this method if the Cisco Unified SRST router is not a DHCP server and you want to relay DHCP requests from IP phones to a DHCP server on a different router.

Defining a Single DHCP IP Address Pool

This task creates a large shared pool of IP addresses in which all DHCP clients receive the same information, including the option 150 TFTP server IP address. The benefit of selecting this method is that you set up only one DHCP pool. However, defining a single DHCP IP address pool can be a problem if non-IP phone clients need to use a different TFTP server address.

SUMMARY STEPS

1. **ip dhcp pool***pool-name*
2. **network** *ip-address*[*mask* | *prefix -length*
3. **option 150 ip** *ip-address*
4. **default-router** *ip-address*
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**ip dhcp pool***pool-name* **Example:** `Router(config)# ip dhcp pool mypool`	Creates a name for the DHCP server address pool and enters DHCP pool configuration mode.
Step 2	**network** *ip-address*[*mask* | *prefix -length* **Example:** `Router(config-dhcp)# network 10.0.0.0 255.255.0.0`	Specifies the IP address of the DHCP address pool and the optional mask or number of bits in the address prefix, preceded by a forward slash.
Step 3	**option 150 ip** *ip-address* **Example:** `Router(config-dhcp)# option 150 ip 10.0.22.1`	Specifies the TFTP server address from which the Cisco IP phone downloads the image configuration file. This needs to be the IP address of Cisco Unified CM.
Step 4	**default-router** *ip-address* **Example:**	Specifies the router to which the Cisco Unified IP phones are connected directly.

	Command or Action	Purpose
	`Router(config-dhcp)# default-router 10.0.0.1`	This router should be the Cisco Unified SRST router because this is the default address that is used to obtain SRST service in the event of a WAN outage. As long as the Cisco IP phones have a connection to the Cisco Unified SRST router, the phones are able to get the required network details.
Step 5	**exit** **Example:** `Router(config-dhcp)# exit`	Exits DHCP pool configuration mode.

Defining a Separate DHCP IP Address Pool for Each Cisco Unified IP Phone

This task creates a name for the DHCP server address pool and specifies IP addresses. This method requires you to make an entry for every Cisco Unified IP phone.

SUMMARY STEPS

1. **ip dhcp pool***pool-name*
2. **host** *ip-address subnet-mask*
3. **option 150 ip** *ip-address*
4. **default-router** *ip-address*
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**ip dhcp pool***pool-name* **Example:** `Router(config)# ip dhcp pool pool2`	Creates a name for the DHCP server address pool and enters DHCP pool configuration mode.
Step 2	**host** *ip-address subnet-mask* **Example:** `Router(config-dhcp)# host 10.0.0.0 255.255.0.0`	Specifies the IP address that you want the phone to use.
Step 3	**option 150 ip** *ip-address* **Example:** `Router(config-dhcp)# option 150 ip 10.0.22.1`	Specifies the TFTP server address from which the Cisco IP phone downloads the image configuration file. This needs to be the IP address of Cisco Unified CM.
Step 4	**default-router** *ip-address* **Example:** `Router(config-dhcp)# default-router 10.0.0.1`	Specifies the router to which the Cisco Unified IP phones are connected directly. This router should be the Cisco Unified SRST router because this is the default address that is used to obtain SRST service in the event of a WAN outage. As long as the Cisco IP phones have a connection to the Cisco Unified

	Command or Action	Purpose
		SRST router, the phones are able to get the required network details.
Step 5	**exit** **Example:** `Router(config-dhcp)# exit`	Exits DHCP pool configuration mode.

Defining the DHCP Relay Server

This task sets up DHCP relay on the LAN interface where the Cisco Unified IP phones are connected and enables the Cisco IOS DHCP server feature to relay requests from DHCP clients (phones) to a DHCP server. For further details about DHCP configuration, see the Cisco IOS DHCP Server document. The Cisco IOS DHCP server feature is enabled on routers by default. If the DHCP server is not enabled on your Cisco Unified SRST router, use the following steps to enable it.

SUMMARY STEPS

1. **service dhcp**
2. **interface** *type number*
3. **ip helper-address** *ip-address*
4. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**service dhcp** **Example:** `Router(config)# service dhcp`	Enables the Cisco IOS DHCP Server feature on the router.
Step 2	**interface** *type number* **Example:** `Router(config)# interface serial 0`	Enters interface configuration mode for the specified interface. See Cisco IOS Interface and Hardware Component Command Reference, Release 12.3T for more information.
Step 3	**ip helper-address** *ip-address* **Example:** `Router(config-if)# ip helper-address 10.0.22.1`	Specifies the helper address for any unrecognized broadcast for TFTP server and Domain Name System (DNS) requests. For each server, a separate **ip helper-address** command is required if the servers are on different hosts. You can also configure multiple TFTP server targets by using the **ip helper-address** command for multiple servers.
Step 4	**exit** **Example:** `Router(config-if)# exit`	Exits interface configuration mode.

Specifying Keepalive Intervals

The keepalive interval is the period of time between keepalive messages sent by a network device. A keepalive message is a message sent by one network device to inform another network device that the virtual circuit between the two is still active.

Note If you plan to use the default time interval between messages, which is 30 seconds, you do not have to perform this task.

SUMMARY STEPS

1. **call-manager-fallback**
2. **keepalive** *seconds*
3. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**call-manager-fallback** **Example:** `Router(config)# call-manager-fallback`	Enters call-manager-fallback configuration mode.
Step 2	**keepalive** *seconds* **Example:** `Router(config-cm-fallback)# keepalive 60`	Sets the time interval, in seconds, between keepalive messages that are sent to the router by Cisco Unified IP Phones. Seconds: Range is 10 to 65535. Default is 30.
Step 3	**exit** **Example:** `Router(config-cm-fallback)# exit`	Exits call-manager-fallback configuration mode.

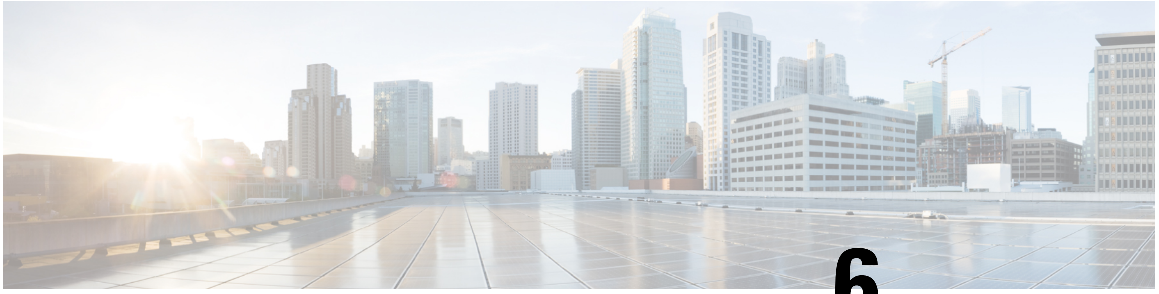
Example

The following example sets a keepalive interval of 45 seconds:

```
call-manager-fallback
keepalive 45
```

What to do next

The next step is setting up the phone and getting a dial tone. For instructions, see the Cisco Unified SIP SRST section.



CHAPTER 6

Cisco Unified SIP SRST

This chapter describes the features and provides the configuration information for Cisco Unified SIP SRST 4.1:

- Out-of-Dialog REFER(OOD-R)
- Digit Collection on SIP Phones
- Caller ID Display
- Disabling SIP Supplementary Services for Call Forward and Call Transfer
- Idle Prompt Status

Note With Cisco IOS Release 12.4(15)T, the number of SIP phones supported on each platform is now equivalent to the number of SCCP phones supported. For example, 3845 now supports 720 phones regardless of whether these are SIP or SCCP.

Prerequisites for Cisco Unified SIP SRST 4.1

- Cisco IOS Release 12.4(15)T or a later release.
- Cisco Unified IP Phones 7911G, 7941G, 7941GE, 7961G, 7961GE, 7970G, and 7971GE require firmware load 8.2(1) or a later version.
- For the prerequisites for the Enhanced 911 Services for Cisco Unified SRST feature introduced in Version 4.1, see Prerequisites for Enhanced 911 Services..

Restrictions for Cisco Unified SIP SRST 4.1

- Cisco Unified SRST does not support line status speed-dial notification, Call Forward All synchronization, dial plans, directory services, or Music On Hold (MOH).
- Before SIP phone load 8.0, SIP phones maintained dual registration with both Cisco Unified Communications Manager and Cisco Unified SRST simultaneously. In SIP phone load 8.0 and later versions, SIP phones use keepalive to maintain a connection with Cisco Unified SRST during active registration with Cisco Unified Communications Manager. Every 2 minutes, a SIP phone sends a keepalive message to Cisco Unified SRST. Cisco Unified SRST responds to this keepalive with a 404 message. This process repeats until fallback to Cisco Unified SRST occurs. After fallback, SIP phones send a keepalive message every two minutes to Cisco Unified Communications Manager while the phones are registered with Cisco Unified SRST. Cisco Unified SRST continues to support dual registration for SIP phone loads older than 8.0.

Information About Cisco Unified SIP SRST 4.1

Out-of-Dialog REFER

Out-of-dialog REFER (OOD-R) enables remote applications to establish calls by sending a REFER message to Cisco Unified SRST without an initial INVITE. After the REFER is sent, the remainder of the call setup is independent of the application and the media stream does not flow through the application. The application using OOD-R triggers a call setup request that specifies the Referee address in the Request-URI and the Refer-Target in the Refer-To header. The SIP messaging used to communicate with Cisco Unified SRST is independent of the end-user device protocol, which can be H.323, plain old telephone service (POTS), SCCP, or SIP. Click-to-dial is an example of an application that can be created using OOD-R.

A click-to-dial application enables users to combine multiple steps into one click for a call setup. For example, a user can click a web-based directory application from his or her PC to look up a phone number, off-hook the desk phone, and dial the called number. The application initiates the call setup without the user having to outdial from his or her own phone. The directory application sends a REFER message to Cisco Unified SRST, which sets up the call between both parties based on this REFER.

For more information about OOD-R, see Out-of-Dialog REFER from the Cisco Unified Communications Manager Express System Administrator Guide.

Digit Collection on SIP Phones

When you dial a phone, the digit strings must be collected and matched against predefined patterns to place calls to the destination corresponding to your input. Previously, SIP phones in a Cisco Unified SRST system required you to press the DIAL softkey or # key, or wait for the interdigit-timeout to trigger the call processing. This could cause delays in processing the call.

Two new methods of collecting and matching digits are supported for SIP phones depending on the model of the phone:

- KPML Digit Collection
- SIP Dial Plans

KPML Digit Collection

The Key Press Markup Language (KPML) uses SIP SUBSCRIBE and NOTIFY methods to report a user input digit by digit. Each digit you dial generates its own signaling message to Cisco Unified SRST. Cisco Unified SRST performs a pattern recognition by matching the destination pattern to the dial peer as it collects the dialed digits. This process of relaying each digit immediately is similar to the process used by SCCP phones. It eliminates the need to press the dial softkey or wait for the interdigit timeout before the digits are sent to the Cisco Unified SRST for processing.

KPML is supported on Cisco Unified IP Phones 7911G, 7941G, 7941GE, 7961G, 7961GE, 7970G, and 7971GE. For configuration information, see Enabling KPML for SIP Phones section.

SIP Dial Plans

A dial plan is a set of dial patterns that SIP phones use to determine when a digit collection is complete after you go off-hook and dial a destination number. Dial plans enable SIP phones to perform local digit collection and recognize dial patterns that you have keyed. After a pattern is recognized, the SIP phone sends an INVITE message to Cisco Unified SRST to initiate the call to the number matching your input. All the digits entered by the user are presented as a block to Cisco Unified SRST for processing. Because digit collection is done by the phone, dial plans reduce signaling messages overhead compared to KPML digit collection.

SIP dial plans eliminate the need for a user to press the Dial softkey or # key or to wait for the interdigit timeout to trigger an outgoing INVITE. You configure a SIP dial plan and associate the dial plan with a SIP phone. The dial plan is downloaded to the phone in the configuration file.

You can configure SIP dial plans and associate them with the following SIP phones:

- Cisco Unified IP Phone 7911G, 7941G, 7941GE, 7961G, 7961GE, 7970G, and 7971GE: These phones use dial plans and support KPML. If both a dial plan and KPML are enabled, the dial plan has priority.

  If a matching dial plan is not found and KPML is disabled, the user must wait for the interdigit timeout before the SIP NOTIFY message is sent to Cisco Unified SRST. Unlike other SIP phones, these phones do not have a Dial softkey to indicate the end of dialing, except when on-hook dialing is used.

- Cisco Unified IP Phone 7905, 7912, 7940, and 7960: These phones use dial plans and do not support KPML. If you do not configure a SIP dial plan for these phones, or if the dialed digits do not match a dial plan, the user must press the Dial softkey or wait for the interdigit timeout before digits are sent to Cisco Unified SRST for processing.

When you reset a phone, the phone requests its configuration files from the TFTP server, which builds the appropriate configuration files depending on the type of phone.

- Cisco Unified IP Phone 7905 and 7912: The dial plan is a field in their configuration files.
- Cisco Unified IP Phone 7911G, 7940, 7941G, 7941GE, 7960, 7961G, 7961GE, 7970G, and 7971GE: The dial plan is a separate XML file that is pointed to from the normal configuration file.

The Cisco Unified SRST supports SIP dial plans if they are provisioned in Cisco Unified Communications Manager. You cannot configure dial plans in Cisco Unified SRST.

Caller ID Display

The Caller ID display includes the name and number of the caller on the Cisco Unified IP Phone 7911G, 7941G, 7941GE, 7961G, 7961GE, 7970G, and 7971GE. Other SIP phones display only the number of the caller. Also, the caller ID information is updated on the destination phone when there is a change in the caller

ID. The change in the caller ID is of the originating party such as with the call forwarding or Call Transfer. No new configuration is required to support these enhancements.

Disabling SIP Supplementary Services for Call Forward and Call Transfer

Perform the following steps to disable REFER messages for Call Transfers and redirect responses for call forwarding from being sent to the destination by Cisco Unified SRST. You can disable these supplementary features if the destination gateway does not support them.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip** OR **dial-peer voice** *tag* **voip**
4. **no supplementary-service sip** {**moved-temporarily** | **refer**}
5. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.

Step 3	**voice service voip** OR **dial-peer voice** *tag* **voip**
	Example:
	`Router(config)# voice service voip`
	OR
	`Router(config)# dial-peer voice 99 voip`
		Enters voice-service configuration mode to set global parameters for VoIP features.
		OR
		Enters dial peer configuration mode to set parameters for a specific dial peer.

Step 4	**no supplementary-service sip** {**moved-temporarily** | **refer**}
	Example:
	`Router(conf-voi-serv)# no supplementary-service sip refer`
	OR
	`Router(config-dial-peer)# no supplementary-service sip refer`
		Disables SIP call forwarding or Call Transfer supplementary services globally or for a dial peer.
		• Moved-temporarily: SIP redirect response for call forwarding.
		• Refer: SIP REFER message for Call Transfers.
		• Sending REFER and redirect messages to the destination is the default behavior.
		Note This command is supported for calls between SIP phones and calls between SCCP phones. It is not supported for a mixture of SCCP and SIP endpoints.

	Command or Action	Purpose

Step 5	**end**
	Example:
	`Router(config-voi-serv)# end`
	OR
	`Router(config-dial-peer)# end`
		Exits to privileged EXEC mode.

Idle Prompt Status

A message displays on the status line of a SIP phone after the phone registers to Cisco Unified SRST to indicate that Cisco Unified SRST is providing fallback support for the Cisco Unified Communications Manager. This message informs the user that the phone is operating in fallback mode and that not all features are available. The default message that displays **CM Fallback Service Operating** is taken from the phone dictionary file. You can customize the message by using the **system message** command on the Cisco Unified SRST router. Cisco Unified SRST updates the idle prompt message when you register a SIP phone or when you modify the message through the configuration. The message displays until a phone switches back to the Cisco Unified Communications Manager.

The idle prompt status message supports the Cisco Unified IP Phone 7911G, 7941G, 7941GE, 7961G, 7961GE, 7970G, and 7971GE with Cisco Unified SRST 4.1 onwards. For versions earlier than Cisco Unified SRST 4.1, the phones display the default message from the dictionary file.

Enhanced 911 Services

Enhanced 911 Services for Cisco Unified SRST enable 911 operators to:

- Immediately pinpoint the location of the 911 caller based on the calling number.
- Call back the 911 caller if a disconnect occurs.

Before this feature was introduced, Cisco Unified SRST supported only outbound calls to 911. With basic 911 functionality, calls were routed to a Public Safety Answering Point (PSAP). The 911 operator at the PSAP would then have to verbally gather the emergency information and location from the caller, before dispatching a response team from the ambulance service, fire department, or police department. Calls could not be routed to different PSAPs, based on the specific geographic areas that they cover.

With Enhanced 911 Services, emergency calls are selectively routed to the closest PSAP based on the caller's location. In addition, the caller's phone number and address automatically display on a terminal at the PSAP. Therefore, the PSAP can quickly dispatch emergency help, even if the caller is unable to communicate the location. Also, if the caller disconnects prematurely, the PSAP has the information to contact the 911 caller.

See Configuring Enhanced 911 Services from Cisco Unified Communications Manager Express System Administrator Guide for more information.

How to Configure Cisco Unified SIP SRST 4.1 Features

Enabling KPML for SIP Phones

Perform the following steps to enable KPML digit collection on a SIP phone.

Before you begin

- This feature is supported only on Cisco Unified IP Phone 7911G, 7941G, 7941GE, 7961G, 7961GE, 7970G, and 7971GE.
- A dial plan assigned to a phone has priority over KPML.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register pool** *pool-tag*
4. **digit collect kpml**
5. **end**
6. **show voice register dial-peers**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice register pool** *pool-tag* **Example:** `Router(config)# voice register pool 4`	Enters voice register Pool configuration mode to set phone-specific parameters for a SIP phone. **pool-tag**: Unique sequence number of the SIP phone to be configured. Range is version and platform-dependent; type **?** to display range. You can modify the upper limit for this argument with the **max-pool** command.
Step 4	**digit collect kpml** **Example:** `Router(config-register-pool)# digit collect kpml`	Enables KPML digit collection for the SIP phone. **Note** This command is enabled by default for supported phones in Cisco Unified Communications Manager Express and Cisco Unified SRST.

	Command or Action	Purpose

Step 5	**end**
	Example:
	`Router(config-register-pool)# end`
		Exits to privileged EXEC mode.

Step 6	**show voice register dial-peers** **Example:** `Router# show voice register dial-peers`	Displays details of all dynamically created VoIP dial peers associated with the Cisco Unified Communications Manager Express SIP register including the defined digit collection method.

What to do next

After changing the KPML configuration in Cisco Unified SRST, you do not need to create new configuration profiles and restart the phones. Enabling or disabling KPML is effective immediately in Cisco Unified SRST.

Disabling SIP Supplementary Services for Call Forward and Call Transfer

Perform the following steps to disable REFER messages for Call Transfers and redirect responses for call forwarding from being sent to the destination by Cisco Unified SRST. You can disable these supplementary features if the destination gateway does not support them.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip** OR **dial-peer voice** *tag* **voip**
4. **no supplementary-service sip** {**moved-temporarily** | **refer**}
5. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.

Step 3	**voice service voip** OR **dial-peer voice** *tag* **voip**
	Example:
	`Router(config)# voice service voip`
	OR
	`Router(config)# dial-peer voice 99 voip`
		Enters voice-service configuration mode to set global parameters for VoIP features.
		OR
		Enters dial peer configuration mode to set parameters for a specific dial peer.

	Command or Action	Purpose

Step 4	**no supplementary-service sip** {**moved-temporarily** | **refer**}
	Example:
	`Router(conf-voi-serv)# no supplementary-service sip refer`
	OR
	`Router(config-dial-peer)# no supplementary-service sip refer`
		Disables SIP call forwarding or Call Transfer supplementary services globally or for a dial peer.
		• Moved-temporarily: SIP redirect response for call forwarding.
		• Refer: SIP REFER message for Call Transfers.
		• Sending REFER and redirect messages to the destination is the default behavior.
		Note This command is supported for calls between SIP phones and calls between SCCP phones. It is not supported for a mixture of SCCP and SIP endpoints.

Step 5	**end**
	Example:
	`Router(config-voi-serv)# end`
	OR
	`Router(config-dial-peer)# end`
		Exits to privileged EXEC mode.

Configuring Idle Prompt Status for SIP Phones

Perform the following steps to customize the message that displays on SIP phones after the phones failover to Cisco Unified SRST.

Before you begin

Cisco Unified SRST 4.1 or a later version.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register global**
4. **system message** *string*
5. **end**
6. **show voice register global**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.

	Command or Action	Purpose
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice register global** **Example:** `Router(config)# voice register global`	Enters voice register global configuration mode to set global parameters for all supported SIP phones in a Cisco Unified Communications Manager Express environment.
Step 4	**system message** *string* **Example:** `Router(config-register-global)# system message fallback active`	Defines a status message that displays on SIP phones registered to Cisco Unified SRST. • String: Up to 32 alphanumeric characters. Default value is **CM Fallback Service Operating**.

Step 5	**end**
	Example:
	`Router(config-register-global)# end`
		Exits to privileged EXEC mode.

Step 6	**show voice register global** **Example:** `Router# show voice register global`	Displays all global configuration parameters associated with SIP phones.

What to do next

The next step is configuring Cisco Unified IP phones using SCCP. For instructions, see Setting Up Cisco Unified IP Phones using SCCP section.

CHAPTER 7

Setting Up Cisco Unified IP Phones using SCCP

This chapter describes how to set up the displays and features that callers will see and use on Cisco Unified IP Phones during Cisco Unified CM fallback.

Note Ciso Unified IP Phones discussed in this chapter are just examples. For a complete list of IP phones, see Compatibility Information.

Information About Setting Up Cisco Unified IP Phones

Cisco Unified IP Phone configuration is limited for Cisco Unified SRST because IP phones retain nearly all Cisco Unified CM settings during Cisco Unified CM fallback. You can configure the date format, time format, language, and system messages that appear on Cisco Unified IP Phones during Cisco Unified Communications Manager fallback. All four of these settings have defaults, and the available language options depend on the IP phones and Cisco Unified CM version in use. Also available for configuration is a secondary dial tone, which can be generated when a phone user dials a predefined PSTN access prefix and can be terminated when additional digits are dialed. Dual-line phone configuration is required for dual-line phone operation during Cisco Unified CM fallback.

How to Set Up Cisco Unified IP Phones

This section contains the following tasks:

- Configuring Cisco Unified SRST to Support Phone Functions (Required)
- Configuring Cisco Unified 8941 and 8945 SCCP IP Phones (Required)
- Verifying That Cisco Unified SRST Is Enabled(Optional)
- Configuring IP Phone Clock, Date, and Time Formats (Optional)

- Configuring IP Phone Language Display (Optional)
- Configuring Customized System Messages for Cisco Unified IP Phones (Optional)
- Configuring a Secondary Dial Tone (Optional)
- Configuring Dual-Line Phones (Required Under Certain Conditions)
- Configuring Eight Calls per Button (Octo-Line)(Optional)
- Configuring the Maximum Number of Calls (Optional)
- Troubleshooting (Optional)

Configuring Cisco Unified SRST to Support Phone Functions

Note When the Cisco Unified SRST is enabled, Cisco Unified IP Phones do not have to be reconfigured while in Cisco Unified Communications Manager fallback mode because phones retain the same configuration that was used with Cisco Unified Communications Manager.

To configure Cisco Unified SRST on the router to support the Cisco Unified IP Phone functions, use the following commands beginning in global configuration mode.

SUMMARY STEPS

1. **call-manager-fallback**
2. **ip source-address** *ip-address* [**port** *port*] [**any-match** | **strict-match**]
3. **max-dn***max-directory-numbers*[**dual-line**][**preference***preference-order*]
4. **max-ephones** *max-ephones*
5. **limit-dn** *phone-type max-lines*
6. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**call-manager-fallback** **Example:** `Router(config)# call-manager-fallback`	Enters call-manager-fallback configuration mode.
Step 2	**ip source-address** *ip-address* [**port** *port*] [**any-match** | **strict-match**] **Example:** `Router(config-cm-fallback)# ip source-address 10.6.21.4 port 2002 strict-match`	Enables the router to receive messages from the Cisco IP phones through the specified IP addresses and provides for strict IP address verification. The default port number is 2000.

	Command or Action	Purpose
Step 3	**max-dn***max-directory-numbers*[**dual-line**][**preference***preference-order*] **Example:** `Router(config-cm-fallback)# max-dn 15 dual-line preference 1`	Sets the maximum number of directory numbers (DNs) or virtual voice ports that can be supported by the router and activates dual-line mode. • **max-directory-numbers**:Maximum number of directory numbers (dns) or virtual voice ports supported by the router. The maximum number is platform-dependent. The default is 0. See Compatibility Information for further details. • **dual-line** (Optional). Allows IP phones in Cisco Unified Communications Manager fallback mode to have a virtual voice port with two channels. • **preference** *preference-order* (Optional). Sets the global preference for creating the VoIP dial peers for all directory numbers that are associated with the primary number. Range is from 0 to 10. Default is 0, which is the highest preference. The **alias** command also has a **preference** keyword that sets **alias** command preference values. Setting the **alias** command**preference** keyword allows the default preference set with the **max-dn** command to be overridden. See the Configuring Call Rerouting section for more information on using the **max-dn** command with the **alias** command. **Note** You must reboot the router to reduce the limit of the directory numbers or virtual voice ports after the maximum allowable number is configured.
Step 4	**max-ephones** *max-ephones* **Example:** `Router(config-cm-fallback)# max-ephones 24`	Configures the maximum number of Cisco IP phones that can be supported by the router. The default is 0. The maximum number is platform dependent. See Compatibility Information for further details. **Note** You must reboot the router to reduce the limit of Cisco IP phones after the maximum allowable number is configured.
Step 5	**limit-dn** *phone-type max-lines* **Example:** `Router(config-cm-fallback)# limit-dn 7945 2`	Optional) Limits the directory number lines on Cisco IP phones during Cisco Unified CM fallback. **Note** You must configure this command during initial Cisco Unified SRST router configuration, before any phone actually registers with the Cisco Unified SRST router. However, you can modify the number of lines at a later time. For a list of available phones, see Cisco SRST and SIP SRST Command Reference (All Versions).

	Command or Action	Purpose
Step 6	**exit** **Example:** `Router(config-cm-fallback)# exit`	Exits call-manager-fallback configuration mode.

Configuring Cisco Unified 8941 and 8945 SCCP IP Phones

To configure Cisco Unified 8941 and 8945 SCCP IP Phones in Unified SRST mode, perform the following commands:

Note This section is required only in SRST version 8.6 and is not required for version 8.8 and higher.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **ephone-type** *phone-type*
4. **device-id***number*
5. **device-type** *phone-type*
6. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**ephone-type** *phone-type* **Example:** `phone-type`	Enters phone type to configure. • 8941 • 8945
Step 4	**device-id***number* **Example:** `Router(config-ephone-type)# device-id 586`	Specifies the device ID for the phone type. • 8941—586 • 8945—585
Step 5	**device-type** *phone-type*	Specifies the device type for the phone.

	Command or Action	Purpose
	Example: `Router(config-ephone-type)# device-type 8941`	• 8941 • 8945

Step 6	**end**
	Example:
	`Router(config-ephone-type)# end`
		Returns to privileged EXEC mode.

Verifying That Cisco Unified SRST Is Enabled

To verify that the Cisco Unified SRST feature is enabled, perform the following steps:

1. Enter the **show running-config** command to verify the configuration.
2. Enter the **show call-manager-fallback all** command to verify that the Cisco Unified SRST feature is enabled.
3. Use the Settings display on the Cisco IP phones in your network to verify that the default router IP address on the phones matches the IP address of the Cisco Unified SRST router.
4. To temporarily block the TCP port 2000 Skinny Client Control Protocol (SCCP) connection for one of the Cisco IP phones to force the Cisco IP phone to lose its connection to the Cisco Unified Communications Manager and register with the Cisco Unified SRST router, perform the following steps:
   a. Use the appropriate IP **access-list** command to temporarily disconnect a Cisco Unified IP Phone from the Cisco Unified Communications Manager. During a WAN connection failure, when Cisco Unified SRST is enabled, Cisco Unified IP Phones display a message informing you that they are operating in Cisco Unified Communications Manager fallback mode. The Cisco IP Phone 7960 and Cisco IP Phone 7940 display a “CM Fallback Service Operating” message, and the Cisco IP Phone 7910 displays a “CM Fallback Service” message when operating in Cisco Unified Communications Manager fallback mode. When the Cisco Unified Communications Manager is restored, the message goes away and full Cisco IP phone functionality is restored.
   b. Use the **debug ephone register** command to observe the registration process of the Cisco IP phone on the Cisco Unified SRST router.
   c. Use the **show ephone** command to display the Cisco IP phones that have registered to the Cisco Unified SRST router.
   d. Enter the **no** form of the appropriate **access-list** command to restore normal service for the phone.

Configuring IP Phone Clock, Date, and Time Formats

The Cisco Unified IP Phone 7970G and Cisco Unified IP Phone 7971G-GE IP phones obtain the correct timezone from Cisco Unified Communications Manager. They also receive the Coordinated Universal Time (UTC) time from the SRST router during SRST registration. When in SRST mode, the phones take the timezone and the UTC time, and apply a timezone offset to produce the correct time display.

Cisco IP Phone 7960 IP phones and other similar SCCP phones such as the Cisco IP Phone 7940, get their display clock information from the local time of the SRST router during SRST registration. If the Cisco Unified

SRST router is configured to use the Network Time Protocol (NTP) to automatically sync the Cisco Unified SRST router time from an NTP time server, only UTC time is delivered to the router. This is because the NTP server could be physically located anywhere in the world, in any timezone. As it is important to display the correct local time, use the clock timezone command to adjust or offset the Cisco Unified SRST router time.

The date and time formats that appear on the displays of all Cisco Unified IP Phones in Cisco Unified CM fallback mode are selected using the **date-format** and **time-format** commands as configured below:

SUMMARY STEPS

1. **clock timezone***zone hours-offset*[*minutes-offset*]
2. **call-manager-fallback**
3. **date-format** {**mm-dd-yy**|**dd-mm-yy**|**yy-dd-mm**|**yy-mm-dd**}
4. **time-format** [**12** | **24**]
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**clock timezone***zone hours-offset*[*minutes-offset*] **Example:** `Router(config)# clock timezone PST -8`	Sets the time zone for display purposes. • **zone**: Name of the time zone to be displayed when standard time is in effect. The length of the zone argument is limited to 7 characters. • **hours-offset**: The number of hour difference from Coordinated Universal Time (UTC). • **minutes-offset** (Optional). Minutes difference from UTC.
Step 2	**call-manager-fallback** **Example:** `Router(config)# call-manager-fallback`	Enters call-manager-fallback configuration mode.

Step 3	**date-format** {**mm-dd-yy**|**dd-mm-yy**|**yy-dd-mm**|**yy-mm-dd**}
	Example:
	`Router(config-cm-fallback)# date-format yy-dd-mm`
		Sets the date format for IP phone display. The choices are **mm-dd-yy**, **dd-mm-yy**, **yy-dd-mm**, and **yy-mm-dd**, where
		• **dd**: day
		• **mm**: month
		• **yy**: year
		The default is set to mm-dd-yy.

Step 4	**time-format** [**12** | **24**] **Example:** `Router(config-cm-fallback)# time-format 24`	Sets the time display format on all Cisco Unified IP Phones registered with the router. The default is set to a 12-hour clock.
Step 5	**exit** **Example:**	Exits call-manager-fallback configuration mode.

	Command or Action	Purpose
	`Router(config-cm-fallback)# exit`	

Example

The following example sets the time zone to Pacific Standard Time (PST), which is 8 hours behind UTC and sets the time display format to a 24 hour clock:

```
Router(config)# clock timezone PST -8
Rounter(config)# call-manager-fallback
Rounter(config-cm-fallback)# time-format 24
```

Configuring IP Phone Language Display

During Cisco Unified CM fallback, the language displays shown on Cisco Unified IP Phones default to the ISO-3166 country code of US (United States). The Cisco Unified IP Phone 7940 and Cisco Unified IP Phone 7960 can be configured for different languages (character sets and spelling conventions) using the **user-locale** command.

Note This configuration option is available in Cisco SRST V2.1 and later versions running under Cisco Unified CM V3.2 and later versions. Systems with software prior to Cisco Unified SRST V2.1 and Cisco Unified CM V3.2 can use the default country, United States (US), only.

SUMMARY STEPS

1. **call-manager-fallback**
2. **configure terminal**
3. **user-locale** *country-code*
4. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**call-manager-fallback** **Example:** `Router(config)# call-manager-fallback`	Enters call-manager-fallback configuration mode.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**user-locale** *country-code* **Example:**	Selects a language by country for displays on the Cisco IP Phone 7940 and Cisco IP Phone 7960.

	Command or Action	Purpose
	`Router(config-cm-fallback)# user-locale ES`	The following ISO-3166 codes are available to Cisco SRST and Cisco Unified SRST systems running under Cisco Communications Manager V3.2 or later versions: • **DE** : German. • **DK**: Danish. • **ES**: Spanish. • **FR**: French. • **IT**: Italian. • **JP**: Japanese Katakana (available under Cisco Unified Communications Manager V4.0 or later versions). • **NL**: Dutch. • **NO**: Norwegian. • **PT**: Portuguese. • **RU**: Russian. • **SE**: Swedish. • **US**: United States English (default).
Step 4	**exit** **Example:** `Router(config-cm-fallback)# exit`	Exits call-manager-fallback configuration mode.

Example

The following example offers a configuration for the Portugal user locale:

```
call-manager-fallback
user-locale PT
```

Configuring Customized System Messages for Cisco Unified IP Phones

Use the system message command to customize the system message displayed on all Cisco Unified IP Phones during Cisco Unified CM fallback.

One of two keywords, primary and secondary, must be included in the command. The primary keyword is for IP phones that can support static text messages during fallback. The default display message for primary IP phones in fallback mode is “CM Fallback Service Operating.”

The secondary keyword is for Cisco Unified IP Phones that do not support static text messages and have a limited display space. Secondary IP phones flash messages during fallback. The default display message for secondary IP phones in fallback mode is “CM Fallback Service.”

Changes to the display message will occur immediately after configuration or at the end of each call.

Note The normal in-service static text message is controlled by Cisco Unified Communications Manager.

SUMMARY STEPS

1. **call-manager-fallback**
2. **system message** {**primary***primary-string*|**secondary***secondary-string*}
3. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**call-manager-fallback** **Example:** `Router(config)# call-manager-fallback`	Enters call-manager-fallback configuration mode.

Step 2	**system message** {**primary***primary-string*|**secondary***secondary-string*}
	Example:
	`Router(config-cm-fallback)# system message primary Custom Message`
		Declares the text for the system display message on IP phones in fallback mode.
		• **primary** *primary-string*: For Cisco Unified IP Phones that can support static text messages during fallback, such as the Cisco Unified IP Phone 7940 and Cisco Unified IP Phone 7960 units. A string of approximately 27 to 30 characters is allowed.
		• **secondary** *secondary-string*: For Cisco Unified IP Phones that do not support static text messages, such as the Cisco Unified IP Phone 7910. A string of approximately 20 characters is allowed.

Step 3	**exit** **Example:** `Router(config-cm-fallback)# exit`	Exits call-manager-fallback configuration mode.

Example

The following example sets “SRST V3.0” as the system display message for all Cisco Unified IP Phones on a router:

```
call-manager-fallback
 system message primary SRST V3.0
 system message secondary SRST V3.0
 exit
```

Configuring a Secondary Dial Tone

A secondary dial tone can be generated when a phone user dials a predefined PSTN access prefix and can be terminated when additional digits are dialed. An example is when a secondary dial tone is heard after the number 9 is dialed to reach an outside line.

SUMMARY STEPS

1. **call-manager-fallback**
2. **secondary-dialtone***digit-string*
3. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**call-manager-fallback** **Example:** `Router(config)# call-manager-fallback`	Enters call-manager-fallback configuration mode.
Step 2	**secondary-dialtone***digit-string* **Example:** `Router(config-cm-fallback)# secondary-dialtone 9`	Activates a secondary dial tone when a digit string is dialed.
Step 3	**exit** **Example:** `Router(config-cm-fallback)# exit`	Exits call-manager-fallback configuration mode.

Example

The following example sets the number 8 to trigger a secondary dial tone:

```
call-manager-fallback
secondary-dialtone 8
```

Configuring Dual-Line Phones

Dual-line phone configuration is required for dual-line phone operation during Cisco Unified CM fallback, see the Enabling Consultative Call Transfer and Forward Using H.450.2 and H.450.3 with Cisco Unified SRST 3.0 section.

Dual-line IP phones are supported during Cisco Unified CM fallback using the **max-dn** command. Dual-line IP phones have one voice port with two channels to handle two independent calls. This capability enables call waiting, call transfer, and conference functions on a phone-line button.

In dual-line mode, each IP phone and its associated line button can support one or two calls. Selection of one of two calls on the same line is made using the blue Navigation button located below the phone display. When one of the dual-line channels is used on a specific phone, other phones that share the ephone-dn will be unable to use the secondary channel. The secondary channel will be reserved for use with the primary dual-line channel.

It is recommended that hunting be disabled to the second channel. For more information, see the Configuring Dial-Peer and Channel Hunting section.

SUMMARY STEPS

1. **call-manager-fallback**
2. **max-dn***max-directory-numbers*[**dual-line**][**preference***preference-order*]
3. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**call-manager-fallback** **Example:** `Router(config)# call-manager-fallback`	Enters call-manager-fallback configuration mode.
Step 2	**max-dn***max-directory-numbers*[**dual-line**][**preference***preference-order*] **Example:** `Router(config-cm-fallback)# max-dn 15 dual-line preference 1`	Sets the maximum number of directory numbers (DNs) or virtual voice ports that can be supported by the router and activates dual-line mode. • **max-directory-numbers**:Maximum number of directory numbers (dns) or virtual voice ports supported by the router. The maximum number is platform-dependent. The default is 0. See Compatibility Information for further details. • **dual-line** (Optional). Allows IP phones in Cisco Unified Communications Manager fallback mode to have a virtual voice port with two channels. • **preference** *preference-order* (Optional). Sets the global preference for creating the VoIP dial peers for all directory numbers that are associated with the primary number. Range is from 0 to 10. Default is 0, which is the highest preference. The **alias** command also has a **preference** keyword that sets **alias** command preference values. Setting the **alias** command**preference** keyword allows the default preference set with the **max-dn** command to be overridden. See the Configuring Call Rerouting section for more information on using the **max-dn** command with the **alias** command.
Step 3	**exit** **Example:** `Router(config-cm-fallback)# exit`	Exits call-manager-fallback configuration mode.

Example

The following example sets the maximum number of DNs or virtual voice ports that can be supported by a router to 10 and activates the dual-line mode for all IP phones in Cisco Unified CM fallback mode:

```
call-manager-fallback
 max-dn 10 dual-line
 exit
```

Configuring Eight Calls per Button (Octo-Line)

The octo-line feature supports up to eight active calls, both incoming and outgoing, on a single button. Eight incoming calls to an octo-line directory number ring simultaneously. After an incoming call is answered, the ringing stops and the remaining seven incoming calls hear a call waiting tone.

After an incoming call on an octo-line directory number is answered, the answering phone is in the connected state. Other phones that share the directory number are in the remoteMultiline state. A subsequent incoming call sends the call waiting tone to the phone connected to the call, and sends the ringing tone to the other phones that are in the remoteMultiline state. All phones sharing the directory number can pick up any of the incoming unanswered calls.

When multiple incoming calls ring on an octo-line directory number that is shared among multiple phones, the ringing tone stops on the phone that answers the call, and the call waiting tone is heard for other unanswered calls. The multiple instances of the ringing calls is displayed on other ephones sharing the directory number. After a connected call on an octo-line directory number is put on-hold, any phone that shares this directory number can pick up the held call. If a phone is in the process of transferring a call or creating a conference, other phones that share the octo-line directory number cannot steal the call.

As new calls come in on an octo-line, the system searches for the next available idle line using the **huntstop chan** *tag*command, where *tag* is a number from 1 to 8. An idle channel is selected from the lowest number to the highest. When the highest number of allowed calls is received, the system stops hunting for available channels. Use this command to limit the number of incoming calls on an octo-line directory number and reserve channels for outgoing calls or features such as call transfer or conference calls.

With the new feature, you can:

- Configure only dual-line mode
- Configure only octo-line mode
- Configure dual-line mode and octo-line mode

Restrictions

Octo-line directory numbers are not supported by the Cisco Unified IP Phone 7902, 7920, or 7931, or by analog phones connected to Cisco ATA or Cisco VG224.

Before you begin

- Cisco Unified SRST 7.0/4.3
- Cisco Unified CM 6.0

- Cisco IOS Release 12.4(15)XZ

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **call-manager-fallback**
4. **max-dn max-no-of-directories**[*dual-line* |*octo-line*] [*number octo-line*]
5. **huntstop channel***1-8*
6. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**call-manager-fallback** **Example:** `Router(config)# call-manager-fallback`	Enters call-manager-fallback configuration mode.
Step 4	**max-dn max-no-of-directories**[*dual-line* |*octo-line*] [*number octo-line*] **Example:** `Router(config-cm-fallback)# max-dn 15 dual-line 6 octo-line`	Sets the maximum number of DNs or virtual voice ports that can be supported by the router and activates dual-line mode, octo-line mode, or both modes. • **max-no-of-directories**: Maximum number of directory numbers (dns) or virtual voice ports supported by the router. The maximum number is platform-dependent. The default is 0. • **dual-line**: (Optional) Allows IP phones in Cisco Unified Communications Manager fallback mode to have a virtual voice port with two channels. • **octo-line**: (Optional) Allows IP phones in Cisco Unified Communications Manager fallback mode to have a virtual voice port with eight channels. • **number** (Optional): Sets the number of directory numbers for octo-mode.
Step 5	**huntstop channel***1-8* **Example:** `Router(config-cm-fallback)# huntstop channel 4`	Enables channel huntstop on an octo-line, which keeps a call from hunting to the next channel of a directory number if the last allowed channel is busy or does not answer.

	Command or Action	Purpose
		• **number**: Number of channels available to accept incoming calls. The remaining channels are reserved for outgoing calls and features such as call transfer, call waiting, and conferencing. The range is 1 to 8 and the default is 8. • The command is supported for octo-line directory numbers only.

Step 6	**end**
	Example:
	`Router(config)# end`
		Returns to privileged EXEC mode.

Example

In the following example, octo-line mode is enabled, there are 8 octo-line directory numbers, there are a maximum of 23 directory numbers, and a maximum of 6 channels are available for incoming calls:

```
!
call-manager-fallback
max-dn 23 octo-line 8
huntstop channel 6
```

Configuring the Maximum Number of Calls

To configure the maximum number of calls on a Cisco Unified SCCP IP phone in Cisco Unified SRST 9.0, perform the following steps.

Before you begin

- Cisco Unified SRST 9.0 and later versions.
- Correct firmware, 9.2(1) or a later version, is installed.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **call-manager-fallback**
4. **max-dn** *max-no-of-directories* [**dual-line** | **octo-line**]
5. **timeouts busy** *seconds*
6. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**call-manager-fallback** **Example:** `Router(config)# call-manager-fallback`	Enables Cisco Unified SRST support and enters call-manager-fallback configuration mode.
Step 4	**max-dn** *max-no-of-directories* [**dual-line** | **octo-line**] **Example:** `Router(config-cm-fallback)# max-dn 10 octo-line`	Sets the maximum possible number of directory numbers or virtual voice ports that can be supported by a router and enables dual-line mode, octo-line mode, or both modes. • **max-no-of-directories**—Maximum number of directory numbers or virtual voice ports supported by the router. The maximum possible number is platform-dependent. The default is 0 directory numbers and 1 channel per virtual port. • **dual-line**—(Optional) Sets all Cisco Unified IP phones connected to a Cisco Unified SRST router to one virtual voice port with two channels. • **octo-line**—(Optional) Sets all Cisco Unified IP phones connected to a Cisco Unified SRST router to one virtual voice port with eight channels.
Step 5	**timeouts busy** *seconds* **Example:** `Router(config-cm-fallback)# timeouts busy 10`	Sets the timeout value for call transfers to busy destinations. • **seconds**—Number of seconds after connection to a busy destination before a transferred call is disconnected. Range is 0 to 30. Default: 10.

Step 6	**end**
	Example:
	`Router(config-cm-fallback)# end`
		Exits configuration mode and enters privileged EXEC mode.

Troubleshooting

To troubleshoot your Cisco Unified SRST configuration, use the following commands:

- To set keepalive debugging for Cisco IP phones, use the **debug ephone keepalive** command.
- To set registration debugging for Cisco IP phones, use the **debug ephone register** command.

- To set state debugging for Cisco IP phones, use the **debug ephone state** command.
- To set detail debugging for Cisco IP phones, use the **debug ephone detail** command.
- To set error debugging for Cisco IP phones, use the **debug ephone error** command.
- To set call statistics debugging for Cisco IP phones, use the **debug ephone statistics** command.
- To provide voice-packet-level debugging and to display the contents of one voice packet in every 1024 voice packets, use the **debug ephone pak** command.
- To provide raw low-level protocol debugging display for all SCCP messages, use the **debug ephone raw** command.

For further debugging, see Cisco IOS Debug Command Reference.

How to Set Up Cisco IP Communicator for Cisco Unified SRST

Cisco IP Communicator is a software-based application that delivers enhanced telephony support on personal computers. Cisco IP Communicator appears on a user’s computer monitor as a graphical, display-based IP phone with a color screen, a keypad, feature buttons, and soft keys.

For information about operation, see the Cisco IP Communicator online help and user documentation.

Prerequisites

You should have the following before you begin this task:

- IP address of the Cisco Unified CM (Call Manager) TFTP server
- IP address of the Cisco Unified SRST TFTP server
- Headset with microphone for your PC (Optional; you can use PC internal speakers and microphone)

1. Download the latest version of the Cisco IP Communicator software and install it on your PC. The software is available for download at http://www.cisco.com/cisco/web/download/index.html.
   a. Click **Voice and Unified Communication**.
   b. Click **IP Telephony**.
   c. Click **IP Phones**.
   d. Click **Cisco IP Communicator**.
2. (Optional) Attach a headset to your PC.
3. Start the Cisco IP Communicator software application.
4. Define the IP address of the Cisco Unified CM as primary TFTP server
   a. Open the **Network** > **User Preferences window**.
   b. Enter the IP address of the Cisco Unified CM TFTP server.
5. Define the IP address of the Cisco Unified SRST as secondary TFTP server

a. Open the **Network** > **User Preferences window**.

b. Enter the IP address of the Cisco Unified SRST TFTP server.

6. Ensure that Cisco IP Communicator has at least once registered to Cisco Unified CM. For more details, see Install and Configure IP Communicator with CallManager.
7. Wait for the Cisco IP Communicator to connect to the Cisco Unified SRST system (upon Cisco Unified CM Failure) and register itself.
8. Cisco IP Communicator should have retained the original buttons and numbers for Cisco IP Communicator.

Verifying Cisco IP Communicator

SUMMARY STEPS

1. Use the **show running-config** command to display ephone-dn and ephone information associated with this phone.
2. After Cisco IP Communicator registers with Cisco Unified SRST, it displays the phone extensions and soft keys in its configuration. Verify that these are correct.
3. Make a local call from the phone and ask someone to call you. Verify that you have a two-way voice path.

DETAILED STEPS

	Command or Action	Purpose
Step 1	Use the **show running-config** command to display ephone-dn and ephone information associated with this phone.	
Step 2	After Cisco IP Communicator registers with Cisco Unified SRST, it displays the phone extensions and soft keys in its configuration. Verify that these are correct.	
Step 3	Make a local call from the phone and ask someone to call you. Verify that you have a two-way voice path.	

Troubleshooting Cisco IP Communicator

Use the **debug ephone detail** command to diagnose problems with calls. For more information, see Cisco IOS Debug Command Reference.

Multicast Music On Hold

For Unified SRST 3.0 and later versions, you can configure the MOH audio stream as a multicast source. A Unified SRST router that is configured for multicast MOH also transmits the audio stream on the physical IP interfaces of the specified router to permit access to the stream by external devices. Certain IP phones do not support multicast MOH because they do not support IP multicast. You can disable multicast MOH to individual phones that do not support multicast. Callers hear a repeating tone when they are placed on hold.

Multicast MOH on Unified SRST is supported for both SIP and SCCP phones. Support is offered for G.711 and G.729 codecs with multicast MOH on Unified SRST. Multicast MOH is supported on Cisco Integrated Services Router Generation 2 (ISR G2) and the Cisco 4000 Series Integrated Services Routers.

For SIP phones to play the Multicast MOH, you need to configure the CLI command **moh** *enable-g711filename* (for example, **moh***enable-g711flash:en_bacd_music_on_hold.au* or **moh** *g729 flash:SampleAudioSource.g729.wav*). For SCCP phones to play Multicast MOH, you need to configure the CLI command **multicast moh** *ip-address* **port** *port-number* [**route** *ip-address-list*] (for example, **multicast moh** *239.1.1.1* **port** *2000*), apart from the CLI command **moh***filename*. If both the CLI commands are not configured, SCCP phones will only play tone on hold.

For more information on supporting Multicast MOH with Unified SRST for a scenario where WAN is available, see Information About Using Cisco Unified SRST Gateways as a Multicast MOH Resource.

Configure Multicast Music On Hold for Unified SRST

Before you begin

To configure multicast MOH for Unified SRST, perform the following steps:

- Unified SRST 3.0 or later versions.
- IP phones do not support multicast at 224.x.x.x addresses.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **call-manager-fallback**
4. **moh***filename*
5. **multicast moh***ip-address***port***port number* [**route***ip-address-list*]
6. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**call-manager-fallback** **Example:** `Router(config)# call-manager-fallback`	Enters call-manager-fallback configuration mode.
Step 4	**moh***filename*	Enables music on hold using the specified file.

	Command or Action	Purpose

	Example:
	`Router(config-cm-fallback)# moh enable-g711 "flash:en_bacd_music_on_hold.au"`
	OR
	`Router(config-cm-fallback)# moh g729 flash:SampleAudioSource.g729.wav`
		- If you specify a file with this command and later want to use a different file, you must disable use of the first file with the no moh command before configuring the second file.

Step 5	**multicast moh***ip-address***port***port number* [**route***ip-address-list*] **Example:** `Router(config-cm-fallback)# multicast moh 239.1.1.1 port 2000`	Specifies that this audio stream is to be used for multicast and also for MOH. **Note** This command is required to use MOH for internal calls and it must be configured after MOH is enabled with the **moh** command. - ip-address—Destination IP address for multicast. - port port-number—Media port for multicast. Range is 2000 to 65535. We recommend port 2000 because it is already used for normal RTP media transmissions between IP phones and the router. **Note** Valid port numbers for multicast include even numbers that range from 16384 to 32767. (The system reserves odd values.) - route—(Optional) List of explicit router interfaces for the IP multicast packets. - ip-address-list—(Optional) List of up to four explicit routes for multicast MOH. The default is that the MOH multicast stream is automatically output on the interfaces that correspond to the address that was configured with the **ip source-address** command. **Note** For MOH on internal calls, packet flow must be enabled to the subnet on which the phones are located.
Step 6	**exit** **Example:** `Router(config-cm-fallback)# exit`	Exits call-manager-fallback configuration mode.

Where to Go Next

The next step is configuring Cisco Unified IP Phones using SIP. For more information, see the Setting Up Cisco Unified IP Phones using SIP section.

For additional information, see the Related Documents and References, on page 58 section.

CHAPTER 8

Setting Up Cisco Unified IP Phones using SIP

Session Initiation Protocol (SIP) registrar functionality in Cisco IOS software is an essential part of Cisco Unified SIP Survivable Remote Site Telephony (SRST). According to RFC 3261, a SIP registrar is a server that accepts Register requests and is typically collocated with a proxy or redirect server. A SIP registrar may also offer location services.

Prerequisites for Configuring the SIP Registrar

Complete the prerequisites documented in the Prerequisites for Configuring Cisco Unified SIP SRST section in Cisco Unified SRST Feature Overview chapter.

Restrictions for Configuring the SIP Registrar

See the restrictions documented in the Restrictions for Configuring Cisco Unified SIP SRST section in the Cisco Unified SRST Feature Overview chapter.

Information About Configuring the SIP Registrar

Cisco Unified SIP SRST provides backup to an external SIP call control (IP-PBX) by providing basic registrar and call handling services. These services are used by a SIP IP phone in the event of a WAN connection outage when the SIP phone is unable to communicate with its primary SIP proxy. The Cisco Unified SIP SRST device also provides PSTN gateway access for placing and receiving PSTN calls.

Cisco Unified SIP SRST works for the following types of calls:

- Local SIP IP phone to local SIP phone, if the main proxy is unavailable.
- Additional services like class of restriction (COR) for local SIP IP phones to the outgoing PSTN. For example, to block outgoing 1-900 numbers.

How to Configure the SIP Registrar

Configuring the SIP Registrar

The local SIP gateway that becomes the SIP registrar acts as a backup SIP proxy and accepts SIP Register messages from SIP phones. It becomes a location database of local SIP IP phones.

A registrar accepts SIP Register requests and dynamically builds VoIP dial peers, allowing the Cisco IOS voice gateway software to route calls to SIP phones.

If a SIP Register request has a Contact header that includes a DNS address, the Contact header is resolved before the contact is added to the SIP registrar database. This is done because during a WAN failure (and the resulting Cisco Unified SIP SRST functionality), DNS servers may not be available.

SIP registrar functionality is enabled with the following configuration. By default, Cisco Unified SIP SRST is not enabled and cannot accept SIP Register messages. The following configuration must be set up to accept incoming SIP Register messages.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **allow-connections sip to sip**
5. **sip**
6. **registrar server** [**expires** [**max***sec*] [**min***sec*]]
7. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice service voip** **Example:** `Router(config)# voice service voip`	Enters voice service configuration mode.
Step 4	**allow-connections sip to sip** **Example:** `Router(config-voi-srv)# allow-connections sip to sip`	Allows connections from SIP to SIP endpoints.

	Command or Action	Purpose
Step 5	**sip** **Example:** `Router(config-voi-srv)# sip`	Enters SIP configuration mode.
Step 6	**registrar server** [**expires** [**max***sec*] [**min***sec*]] **Example:** `Router(conf-serv-sip)# registrar server expires max 600 min 60`	Enables SIP registrar functionality. The keywords and arguments are defined as follows: • expires: (Optional) Sets the active time for an incoming registration. • max sec: (Optional) Maximum expiration time for a registration, in seconds. The range is from 600 to 86400. The default is 3600. **Note** Ensure that the registration expiration timeout is set to a value smaller than the TCP connection aging timeout to avoid disconnection from the TCP. • min sec: (Optional) Minimum expiration time for a registration, in seconds. The range is from 60 to 3600. The default is 60.

Step 7	**end**
	Example:
	`Router(conf-serv-sip)# end`
		Returns to privileged EXEC mode.

What to do next

For incoming SIP Register messages to be successfully accepted, users must also set up a voice register pool. See the section Configuring Backup Registrar Service to SIP Phones.

Configuring Backup Registrar Service to SIP Phones

Backup registrar service to SIP IP phones can be provided by configuring a voice register pool on SIP gateways. The voice register pool configuration provides registration permission control and can also be used to configure some dial-peer attributes that are applied to the dynamically created VoIP dial peers when SIP phone registrations match the pool. The following call types are supported:

SIP IP phone to or from:

- Local PSTN
- Local analog FXS phones
- Local SIP IP phone

The commands in the configuration below provide registration permission control and set up a basic voice register pool. The pool gives users control over which registrations are accepted by a Cisco Unified SIP SRST device and which can be rejected. Registrations that match this pool create VoIP SIP dial peers with the dial-peer attributes set to these configurations. Although only the id command is mandatory, this configuration example shows basic functionality.

For command-level information, see the appropriate command page in Cisco Unified SRST and Cisco Unified SIP SRST Command Reference (All Versions).

Before you begin

The SIP registrar must be configured before a voice register pool is set up. See the section Configuring the SIP Registrar.

Restrictions

- The **id** command identifies the individual SIP IP phone or sets of SIP IP phones that are to be configured. Thus, the**id**command configured in Step 5 is required and must be configured before any other voice register pool commands. When the**mac***address* keyword and argument are used, the IP phone must be in the same subnet as that of the router's LAN interface, such that the phone's MAC address is visible in the router's Address Resolution Protocol (ARP) cache. Once a MAC address is configured for a specific voice register pool, remove the existing MAC address before changing to a new MAC address.
- Proxy dial peers are autogenerated dial peers that route all calls from the PSTN to Cisco Unified SIP SRST. When a SIP phone registers to Cisco Unified SIP SRST and the **proxy** command is enabled, two dial peers are automatically created. The first dial peer routes to the proxy, and the second (or fallback) dial peer routes to the SIP phone. The same functionality can also be achieved with the appropriate creation of static dial peers (manually creating dial peers that point to the proxy). Proxy dial peers can be monitored to one proxy IP address, only. That is, only one proxy from a voice registration pool can be monitored at a time. If more than one proxy address needs to be monitored, you must manually create and configure additional dial peers.
- If Jabber for desktop clients must register with Unified SRST, ensure that**voice register pools** are configured for all desktop computer networks.

Note To monitor SIP proxies, the **call fallback active** command must be configured, as described in Step 3

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **call fallback active**
4. **voice register pool** *tag*
5. **id** { **network** *address* **mask** *mask* | **ip** *address* **mask** *mask* | **mac** *address* }
6. **preference** *preference-order*
7. **proxy** *ip-address* [**preference** *value* [**monitor probe {icmp-ping | rtr }** *alternate-ip-address*]]
8. **voice-class codec** *tag*
9. (Optional) **application** *application-name*
10. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**call fallback active** **Example:** `Router(config)# call fallback active`	Enables a call request to fall back to alternate dial peers in case of network congestion. This command is used if you want to monitor the proxy dial peer and fallback to the next preferred dial peer. For full information on the call fallback active command, see PSTN Fallback Feature.
Step 4	**voice register pool** *tag* **Example:** `Router(config)# voice register pool 12`	Enters voice register pool configuration mode for SIP phones. Use this command to control which registrations are accepted or rejected by a Cisco Unified SIP SRST device.

Step 5	**id** { **network** *address* **mask** *mask* | **ip** *address* **mask** *mask* | **mac** *address* }
	Example:
	`Router(config-register-pool)# id network 172.16.0.0 mask 255.255.0.0`
		Explicitly identifies a locally available individual or set of SIP IP phones. The keywords and arguments are defined as follows:
		• **network** *address* **mask** *mask* : The **network** *address* **mask** *mask* keyword/argument combination is used to accept SIP Register messages for the indicated phone numbers from any IP phone within the indicated IP subnet.
		• **ip** *address* **mask** *mask* : The **ip** *address* **mask** *mask* keyword/argument combination is used to identify an individual phone.
		• **mac** *address* : MAC address of a particular Cisco Unified IP Phone.

Step 6	**preference** *preference-order* **Example:** `Router(config-register-pool)# preference 2`	Sets the preference order for the VoIP dial peers to be created. Range is from 0 to 10. Default is 0, which is the highest preference. The preference must be greater (lower priority) than the preference configured with the **preference** keyword in the **proxy** command.
Step 7	**proxy** *ip-address* [**preference** *value* [**monitor probe {icmp-ping | rtr }** *alternate-ip-address*]]	Autogenerates additional VoIP dial peers to reach the main SIP proxy whenever a Cisco Unified SIP IP Phone registers

	Command or Action	Purpose
	Example: `Router(config-register-pool)# proxy 10.2.161.187 preference 1`	with a Cisco Unified SIP SRST gateway. The keywords and arguments are defined as follows: • *ip-address* : The *ip-address* of the SIP Proxy. • **preference** *value* : Defines the preference of the proxy dial peers that are created. The preference must be less (higher priority) than the preference configured with the **reference** command. Range is from 0 to 10. The highest preference is 0. There is no default. • **monitor probe** : Enables monitoring of proxy dial peers. • **icmp-ping** : Enables monitoring of proxy dial peers using ICMP ping. **Note** The dial peer on which the probe is configured will be excluded from call routing only for outbound calls. Inbound calls can arrive through this dial peer. • **rtr** : Enables monitoring of proxy dial peers using RTR probes. • *alternate-ip-address* : Enables monitoring of alternate IP addresses other than the proxy address. For example, to monitor a gateway front end to a SIP proxy.
Step 8	**voice-class codec** *tag* **Example:** `Router(config-register-pool)# voice-class codec 15`	Sets the voice class codec parameters. The tag argument is a codec group number between 1 and 10000.
Step 9	(Optional) **application** *application-name* **Example:** `Router(config-register-pool)# application SIP.App`	Selects the session-level application on the VoIP dial peer. Use the *application-name* argument to define a specific interactive voice response (IVR) application.

Step 10	**end**
	Example:
	`Router(config-register-pool)# end`
		Returns to privileged EXEC mode.

What to do next

There are several more voice register pool commands that add functionality, but that are not required. See the section Configuring Backup Registrar Service to SIP Phone (Using Optional Commands) for these commands.

Configuring Backup Registrar Service to SIP Phone (Using Optional Commands)

The prior configurations set up a basic voice register pool. The configuration in this procedure adds optional attributes to increase functionality.

Before you begin

- Prerequisites as described in the Configuring Backup Registrar Service to SIP Phones section.
- Configuration of the required commands as described in the Configuring Backup Registrar Service to SIP Phones section .
- Before configuring the **alias** command, translation rules must be set using the **translate-outgoing (voice register pool)** command.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register pool** *tag*
4. **translation-profile outgoing** *profile-tag*
5. **alias** *tag pattern* **to** *target* [**preference** *value*]
6. **cor {incoming | outgoing}** *cor-list-name {cor-list-number starting-number [- ending-number] |* **default }**
7. **incoming called-number** *[number]*
8. **number** *tag number-pattern {* **preference** *value }* [**huntstop**]
9. **dtmf-relay [cisco-rtp] [rtp-nte] [sip-notify]**
10. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice register pool** *tag* **Example:** `Router(config)# voice register pool 12`	Enters voice register pool configuration mode. Use this command to control which registrations are accepted or rejected by a Cisco Unified SIP SRST device.
Step 4	**translation-profile outgoing** *profile-tag* **Example:** `Router(config-register-pool)# voice translation-rule 1`	Use this command to apply the translation profile to a specific directory number or to all directory numbers on a SIP phone.

	Command or Action	Purpose

	`rule 1 /1000/ /1006/`
	`!`
	`!`
	`voice translation-profile 1`
	`translate called 1`
	`!`
	`voice register pool xxx`
	`translation-profile outgoing 1`
		Profile-tag: Translation profile name to handle translation to outgoing calls.

Step 5	**alias** *tag pattern* **to** *target* [**preference** *value*] **Example:** `Router(config-register-pool)# alias 1 94... to 91011 preference 8`	Allows Cisco Unified SIP IP Phones to handle inbound PSTN calls to telephone numbers that are unavailable when the main proxy is not available. The keywords and arguments are defined as follows: • tag : Number from 1 to 5 and the distinguishing factor when there are multiple alias commands. • *pattern* : The prefix number; matches the incoming telephone number and may include wildcards. • **to**: Connects the tag number pattern to the alternate number. • *target* : The target number; an alternate telephone number to route incoming calls to match the number pattern. • **preference***value* : Assigns a dial-peer preference value to the alias. The *value* argument is the value of the associated dial peer, and the range is from 1 to 10. There is no default.

Step 6	**cor {incoming | outgoing}** *cor-list-name {cor-list-number starting-number [- ending-number] |* **default }**
	Example:
	`Router(config-register-pool)# cor incoming call91 1 91011`
		Configures a class of restriction (COR) on the VoIP dial peers associated with directory numbers. COR specifies which incoming dial peers can use which outgoing dial peers to make a call. Each dial peer can be provisioned with an incoming and outgoing COR list. The keywords and arguments are defined as follows:
		• **incoming** : COR list to be used by incoming dial peers.
		• **outgoing** : COR list to be used by outgoing dial peers.
		• *cor-list-name* : COR list name.
		• *cor-list-number* : COR list identifier. The maximum number of COR lists that can be created is four, comprised of incoming or outgoing dial peers.
		• *starting-number* : Start of a directory number range, if an ending number is included. Can also be a standalone number.
		• Indicator that a full range is configured.

	Command or Action	Purpose
		• *ending-number* : End of a directory number range. • **default**: Instructs the router to use an existing default COR list.
Step 7	**incoming called-number** *[number]* **Example:** `Router(config-register-pool)# incoming called-number 308`	Applies incoming called parameters to dynamically created dial peers. The number argument is optional and indicates a sequence of digits that represent a phone number prefix.
Step 8	**number** *tag number-pattern {* **preference** *value }* [**huntstop**] **Example:** `Router(config-register-pool)# number 1 50.. preference 2`	Indicates the E.164 phone numbers that the registrar permits to handle the Register message from the Cisco Unified SIP IP Phone. The keywords and arguments are defined as follows: • *tag* : Number from 1 to 10 and the distinguishing factor when there are multiple number commands. • *number-pattern* : Phone numbers (including wildcards and patterns) that are permitted by the registrar to handle the Register message from the SIP IP phone. • **preference value** : Defines the number list preference order. • **huntstop** : Stops hunting if the dial peer is busy.
Step 9	**dtmf-relay [cisco-rtp] [rtp-nte] [sip-notify]** **Example:** `Router(config-register-pool)# dtmf-relay rtp-nte`	Specifies how a SIP gateway relays dual tone multifrequency (DTMF) tones between telephony interfaces and an IP network. The keywords are defined as follows: • **cisco-rtp**: Forwards DTMF tones by using Real-Time Transport Protocol (RTP) with a Cisco proprietary payload type. • **rtp-nte**: Forwards DTMF tones by using RTP with the Named Telephone Event (NTE) payload type. • **sip-notify**: Forwards DTMF tones using SIP NOTIFY messages.

Step 10	**end**
	Example:
	`Router(config-register-pool)# end`
		Returns to privileged EXEC mode.

Example

The following partial output from the show running-config command shows that voice register pool 12 is configured to accept all registrations from SIP IP phones with extension number 50xx from the

172.16.0.0/16 network. Autogenerated dial peers for registrations that match pool 12 have attributes configured in this pool.

```
.
.
.
voice register pool 12
id network 172.16.0.0 mask 255.255.0.0
number 1 50.. preference 2
application SIP.app
preference 2
incoming called-number
cor incoming allowall default
translate-outgoing called 1
voice-class codec 1
.
.
.
```

Verifying SIP Registrar Configuration

To help you troubleshoot a SIP registrar and voice register pool, perform the following steps.

SUMMARY STEPS

1. **debug voice register errors**
2. **debug voice register events**
3. **show sip-ua status registrar**

DETAILED STEPS

	Command or Action	Purpose

Step 1	**debug voice register errors**
	Example:
	`Router# debug voice register errors`
	`*Apr 22 11:52:54.523 PDT: VOICE_REG_POOL: Contact doesn't match any pools`
	`*Apr 22 11:52:54.539 PDT: VOICE_REG_POOL: Register request for (33015) from (10.2.152.39)`
	`*Apr 22 11:52:54.539 PDT: VOICE_REG_POOL: Contact doesn't match any pools.`
	`*Apr 22 11:52:54.559 PDT: VOICE_REG_POOL: Register request for (33017) from (10.2.152.39)`
	`*Apr 22 11:53:04.559 PDT: VOICE_REG_POOL: Maximum registration threshold for pool(3) hit`
		Use this command to debug errors that happen during registration.
		If there are no voice register pools configured for a particular registration request, the message `Contact doesn’t match any pools` is displayed.

Step 2	**debug voice register events**
	Example:
	`Router# debug voice register events`
	`Apr 22 10:50:21.731 PDT: VOICE_REG_POOL: Contact matches pool 1`
	`Apr 22 10:50:21.731 PDT: VOICE_REG_POOL: key(91011) contact(192.168.0.2) add to contact table`
		Using the **debug voice register events** command should suffice to display registration activity. Registration activity includes matching of pools, registration creation, and automatic creation of dial peers. For more details and error conditions, you can use the **debug voice register errors** command.

	Command or Action	Purpose

	`Apr 22 10:50:21.731 PDT: VOICE_REG_POOL: key(91011) exists in contact table`
	`Apr 22 10:50:21.731 PDT: VOICE_REG_POOL: contact(192.168.0.2) exists in contact table, ref updated`
	`Apr 22 10:50:21.731 PDT: VOICE_REG_POOL: Created dial-peer entry of type 1`
	`Apr 22 10:50:21.731 PDT: VOICE_REG_POOL: Registration successful for 91011, registration id is 257`
		The phone number 91011 registered successfully, and type 1 is reported, which means there is a pre-existing VoIP dial peer.

Step 3	**show sip-ua status registrar**
	Example:
	`Router# show sip-ua status registrar`
	`Line    destination expires(sec) contact`
	`======= =========== ============ =======`
	`91021   192.168.0.3 227          192.168.0.3`
	`91011   192.168.0.2 176          192.168.0.2`
	`95021   10.2.161.50 419          10.2.161.50`
	`95012   10.2.161.50 419          10.2.161.50`
	`95011   10.2.161.50 420          10.2.161.50`
	`95500   10.2.161.50 420          10.2.161.50`
	`94011   10.2.161.40 128          10.2.161.40`
	`94500   10.2.161.40 129          10.2.161.40`
		Use this command to display all the SIP endpoints currently registered with the contact address.

Verifying Proxy Dial-Peer Configuration

To use the **icmp-ping** keyword with the **proxy** command to assist in troubleshooting proxy dial peers, perform the following steps.

SUMMARY STEPS

1. **configure terminal**
2. **voice register pool**
3. **proxy** *ip-address*[**preference***value*] [**monitor probe {icmp-ping|rtr}**[*alternate-ip-address*]]
4. **end**
5. **show voice register dial-peers**
6. **show dial-peer voice**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**configure terminal** **Example:** `Router# configure terminal`	Use this command to enter global configuration mode.
Step 2	**voice register pool** **Example:** `Router(config)# voice register pool 1`	Use this command to enter voice register pool configuration mode.

	Command or Action	Purpose
Step 3	**proxy** *ip-address*[**preference***value*] [**monitor probe {icmp-ping|rtr}**[*alternate-ip-address*]] **Example:** `Router(config-register-pool)# proxy 10.2.161.187 preference 1 monitor probe icmp-ping`	Set the **proxy** command to monitor with **icmp-ping**.

Step 4	**end**
	Example:
	`Router(config-register-pool)# end`
		Returns to privileged EXEC mode.

Step 5	**show voice register dial-peers**
	Example:
	`Router# show voice register dial-peers`
	`dial-peer voice 40035 voip`
	`preference 5`
	`destination-pattern 91011`
	`session target ipv4:192.168.0.2`
	`session protocol sipv2`
	`voice-class codec 1`
	`dial-peer voice 40036 voip`
	`preference 1`
	`destination-pattern 91011`
	`session target ipv4:10.2.161.187`
	`session protocol sipv2`
	`voice-class codec 1`
	`monitor probe icmp-ping 10.2.161.187`
		Use this command to verify dial-peer configurations, and notice that **icmp-ping** monitoring is set.

Step 6	**show dial-peer voice** **Example:** Router# show dial-peer voice VoiceOverIpPeer40036 peer type = voice, information type = voice, description = `', tag = 40036, destination-pattern = `91011', answer-address = `', preference=1, CLID Restriction = None CLID Network Number = `' CLID Second Number sent source carrier-id = `', target carrier-id = `', source trunk-group-label = `', target trunk-group-label = `', numbering Type = `unknown' group = 40036, Admin state is up, Operation state is up, incoming called-number = `', connections/maximum = 0/unlimited, ! Default output for incoming called-number command DTMF Relay = disabled, modem transport = system, huntstop = disabled, in bound application associated: 'DEFAULT' out bound application associated: '' dnis-map = permission :both	Use the show dial-peer voice command on dial peer 40036, and notice the monitor probe status. **Note** Also highlighted is the output of the **cor** and **incoming called-number** commands.

	Command or Action	Purpose

```
incoming COR list:maximum capability
! Default output for cor command
outgoing COR list:minimum requirement
! Default output for cor command
Translation profile (Incoming):
Translation profile (Outgoing):
incoming call blocking:
translation-profile = `'
disconnect-cause = `no-service'
advertise 0x40 capacity_update_timer 25 addrFamily
 4
oldAddrFamily 4
type = voip, session-target = `ipv4:10.2.161.187',
technology prefix:
settle-call = disabled
ip media DSCP = ef, ip signaling DSCP = af31,
ip video rsvp-none DSCP = af41,ip video rsvp-pass
DSCP = af41
ip video rsvp-fail DSCP = af41,
UDP checksum = disabled,
session-protocol = sipv2, session-transport =
system,
req-qos = best-effort, acc-qos = best-effort,
req-qos video = best-effort, acc-qos video =
best-effort,
req-qos audio def bandwidth = 64, req-qos audio
max
bandwidth = 0,
req-qos video def bandwidth = 384, req-qos video
max
bandwidth = 0,
RTP dynamic payload type values: NTE = 101
Cisco: NSE=100, fax=96, fax-ack=97, dtmf=121,
fax-relay=122
S=123, ClearChan=125, PCM switch over
u-law=0,A-law=8
RTP comfort noise payload type = 19
fax rate = voice, payload size = 20 bytes
fax protocol = system
fax-relay ecm enable
fax NSF = 0xAD0051 (default)
codec = g729r8, payload size = 20 bytes,
Media Setting = flow-through (global)
Expect factor = 0, Icpif = 20,
Playout Mode is set to adaptive,
Initial 60 ms, Max 300 ms
Playout-delay Minimum mode is set to default, value
40 ms
Fax nominal 300 ms
Max Redirects = 1, signaling-type = cas,
VAD = enabled, Poor QOV Trap = disabled,
Source Interface = NONE
voice class sip url = system,
voice class sip rel1xx = system,
monitor probe method: icmp-ping ip address:
10.2.161.187,
Monitored destination reachable
voice class perm tag = `'
Time elapsed since last clearing of voice call
statistics never
Connect Time = 0, Charged Units = 0,
Successful Calls = 0, Failed Calls = 0, Incomplete
```

	Command or Action	Purpose

	`Calls = 0`
	`Accepted Calls = 0, Refused Calls = 0,`
	`Last Disconnect Cause is "",`
	`Last Disconnect Text is "",`
	`Last Setup Time = 0.`	

What to do next

The next step is configuring incoming and outgoing calls for Cisco Unified SRST. For more information, see the Configuring Call Handling section.

IPv6 Support for Unified SRST SIP IP Phones

Internet Protocol version 6 (IPv6) is the latest version of the Internet Protocol (IP). IPv6 uses packets to exchange data, voice, and video traffic over digital networks. Also, IPv6 increases the number of network address bits from 32 bits in IPv4 to 128 bits. From Unified SRST Release 12.0 onwards, Unified SRST supports IPv6 protocols for SIP IP phones.

IPv6 support in Unified SRST allows the network to behave transparently in a dual-stack (IPv4 and IPv6) environment and provides additional IP address space to SIP IP phones that are connected to the network. If you do not have a dual-stack configuration, configure the CLI command **call service stop** under **voice service voip** configuration mode before changing to dual-stack mode. For an example of switching to dual-stack mode, see Examples for Configuring IPv6 Pools for SIP IP Phones, on page 210.

The Cisco IP Phone 7800 Series and 8800 Series are supported on IPv6 for Unified SRST.

For more information on configuring SIP IP phones for IPv6 source address, see Configure IPv6 Pools for SIP IP Phones, on page 207.

For an example of configuring IPv6 Support on Unified SRST, see Examples for Configuring IPv6 Pools for SIP IP Phones, on page 210.

For more details about IPv6 deployment, see IPv6 Deployment Guide for Cisco Collaboration Systems Release 12.0.

Feature Support for IPv6 in Unified SRST SIP IP Phones

The basic feature supported for a IPv6 WAN down scenario is:

Basic SIP Line (IPv4 or IPv6) to SIP Line calls (IPv4 or IPv6) when Unified SRST is in dual-stack **no anat** mode.

The following supplementary services are supported as part of IPv6 in Unified SRST IP Phones:

- Hold/Resume
- Call Forward
- Call Transfer
- Three-way Conference (with BIB conferencing only)
- Line to T1/E1 Trunk and Trunk to Line with Supplementary Service Features

- Fax to and from PSTN (IPv4 ATA to ISDN T1/E1) for both T.38 Fax Relay and Fax Passthrough

Restrictions

The following are the known restrictions for IPv6 support on Unified SRST:

- SIP Trunks are not supported on Unified SRST for IPv6 deployment. PSTN calls are supported only through T1/E1 trunks.
- SCCP IP Phones are not supported in a deployment of IPv6 for Unified SRST.
- SIP Phones can be either in IPv4 only or IPv6 only mode (**no anat**).
- Trancoding and Transrating are not supported.
- H.323 trunks are not supported.
- Secure SIP lines or trunks are not supported.
- IPv6 on Unified SRST is not supported on the Cisco IOS platform. The support is restricted to Cisco IOS XE platform with Cisco IOS Release 16.6.1 or later versions.
- For IPv6 Support on Unified SRST, all the legacy IP Phones and Voice Gateways must be converted or reconfigured to IPv4-Only SIP signaling from SCCP signaling, if applicable.

Configure IPv6 Pools for SIP IP Phones

Before you begin

- Unified SRST 12.0 or a later version.
- IPv6 option only appears if protocol mode is dual-stack configured under sip-ua configuration mode or IPv6.
- Cisco Unified SRST License must be configured for the gateway to function as a Unified SRST gateway to support IPv6 functionality. For more information on licenses, see Licensing.
- Cisco Unified Communications Manager (Unified Communications Manager) is provisioned with the IPv6 address of Unified SRST. For information on configuration of Unified SRST on Unified Communications Manager, see Survivable Remote Site Telephony Configurationin Cisco Unified Communications Manager Administration Guide.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **ipv6 unicast-routing**
4. **voice service voip**
5. **sip**
6. **no anat**
7. **call service stop**
8. **exit**

9. **exit**
10. **sip-ua**
11. **protocol mode** { **ipv4** | **ipv6** | **dual-stack** [**preference** { **ipv4** | **ipv6** }] }
12. **exit**
13. **voice service** { **voip** }
14. **sip**
15. **no call service stop**
16. **exit**
17. **voice register global**
18. **default mode**
19. **max-dn***max-directory-numbers*
20. **max-pool***max-voice-register-pools*
21. **exit**
22. **voice register pool***pool-tag*
23. **id** { **network***address***mask***mask* | **ip address mask***mask* | **mac***address* }
24. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router #configure terminal`	Enters global configuration mode.
Step 3	**ipv6 unicast-routing** **Example:** `Router(config)# ipv6 unicast-routing`	Enables the forwarding of IPv6 unicast datagrams.
Step 4	**voice service voip** **Example:** `Router (config)# voice service voip`	Enters voice-service configuration mode to specify a voice encapsulation type. • voip — Specifies Voice over IP (VoIP) parameters.
Step 5	**sip** **Example:** `Router(config-voi-serv)# sip`	Enters SIP configuration mode.
Step 6	**no anat** **Example:** `Router(config-serv-sip)# no anat`	Disables Alternative Network Address Types (ANAT) on a SIP trunk.

	Command or Action	Purpose
Step 7	**call service stop** **Example:** `Router(config-serv-sip)# call service stop`	Shuts down SIP call service.
Step 8	**exit** **Example:** `Router(config-serv-sip)# exit`	Exits SIP configuration mode.
Step 9	**exit** **Example:** `Router(config-voi-sip)# exit`	Exits voice service voip configuration mode.
Step 10	**sip-ua** **Example:** `Router(config)# sip-ua`	Enters SIP user-agent configuration mode.

Step 11	**protocol mode**{**ipv4** | **ipv6** | **dual-stack** [**preference**{**ipv4** | **ipv6**}]}
	Example:
	`Router(config-sip-ua)# protocol mode dual-stack preference ipv6`
		Allows phones to interact with phones on IPv6 voice gateways. You can configure phones for IPv4 addresses, IPv6 address es, or for a dual-stack mode.
		• ipv4—Allows you to set the protocol mode as an IPv4 address.
		• ipv6—Allows you to set the protocol mode as an IPv6 address.
		• dual-stack—Allows you to set the protocol mode for both IPv4 and IPv6 addresses.
		• preference—Allows you to choose a preferred IP address family if protocol mode is dual-stack.

Step 12	**exit** **Example:** `Router(config-sip-ua)# exit`	Exits SIP configuration mode.

Step 13	**voice service**{**voip**}
	Example:
	`Router (config)# voice service voip`
		Enters voice-service configuration mode to specify a voice encapsulation type.
		• voip — Specifies Voice over IP (VoIP) parameters.

Step 14	**sip** **Example:** `Router(config-voi-serv)# sip`	Enters SIP configuration mode.
Step 15	**no call service stop** **Example:** `Router(config-serv-sip)# call service stop`	Activates SIP call service.

	Command or Action	Purpose
Step 16	**exit** **Example:** `Router(config-serv-sip)# exit`	Exits SIP configuration mode.
Step 17	**voice register global** **Example:** `Router(config)# voice register global`	Enters voice register global configuration mode to set parameters for all supported SIP phones in Cisco Unified CME.
Step 18	**default mode** **Example:** `Router(config-register-global)# default mode`	Enables mode for provisioning SIP phones in Unified SRST. The default mode is Unified SRST itself.
Step 19	**max-dn***max-directory-numbers* **Example:** `Router(config-register-global)# max-dn 50`	Limits number of directory numbers to be supported by this router. Maximum number is platform and version-specific. Type ? for value.
Step 20	**max-pool***max-voice-register-pools* **Example:** `Router(config-register-global)# max-pool 40`	Sets maximum number of SIP phones to be supported by the Unified SRST router.
Step 21	**exit** **Example:** `Router(config-register-global)# exit`	Exits voice register global configuration mode.
Step 22	**voice register pool***pool-tag* **Example:** `Router(config)# voice register pool 1`	Enters voice register pool configuration mode to set phone-specific parameters for a SIP phone.

Step 23	**id**{**network***address***mask***mask* | **ip address mask***mask* | **mac***address*}
	Example:
	`Router(config-register-pool)# id network 2001:420:54FF:13::901:0/117`
	`Router(config-register-pool)# id network 10.64.88.0 mask 255.255.255.0`
		Explicitly identifies a locally available individual SIP phone to support a degree of authentication.

Step 24	**end**
	Example:
	`Router(config)# end`
		Exits to privileged EXEC mode.

Examples for Configuring IPv6 Pools for SIP IP Phones

The following example provides configuration of IPv6 pools for SIP IP Phones:

```
ipv6 unicast-routing
voice service voip
sip
no anat
call service stop
exit
exit
sip-ua
protocol mode dual-stack
exit
voice service voip
sip
no call service stop
exit
voice register global
default mode
max-dn 50
max-pool 40
exit
voice register pool 1
id network 2001:420:54FF:13::901:0/117
end
```

The following example provides interface configuration for IPv6 supported on Unified SRST:

```
configure terminal
interface GigabitEthernet0/0/1
 ip address 10.64.86.229 255.255.255.0
 negotiation auto
 ipv6 address 2001:420:54FF:13::312:82/119
 ipv6 enable
```

The following example provides IP route configuration for IPv6 supported on Unified SRST:

```
ipv6 route 2001:420:54FF:13::312:0/119 2001:420:54FF:13::312:1
ipv6 route 2001:420:54FF:13::901:0/119 2001:420:54FF:13::312:1
```

The following example displays output when SIP call service is shut down with the **call service stop** CLI command:

```
Router# show sip service
SIP service is shut
under voice service voip, sip submode
```

The following example displays output when SIP call service is active with the **no call service stop** CLI command:

```
Router# show sip-ua service
SIP Service is up
under voice service voip, sip submode
```

CHAPTER 9

Configuring Call Handling

This chapter describes how to configure Cisco Unified Survivable Remote Site Telephony (Cisco Unified SRST) for incoming and outgoing calls for SCCP phones.

This chapter also describes support for standardized RFC 3261 features for SIP phones. Features include call blocking and call forwarding.

Note Configuring Call Handling for SIP phones applies to versions 4.0 and 3.4 only.

Prerequisites for Configuring SIP SRST Features Using Back-to-Back User Agent Mode

- Complete the prerequisites documented in the Prerequisites for Configuring Cisco Unified SIP SRST section in the Cisco Unified SRST Feature Overview, on page 41.

- Configure the SIP registrar. The SIP registrar gives users control of accepting or rejecting registrations. To configure acceptance of incoming SIP Register messages, see the Prerequisites for Configuring the SIP Registrar section.

Restrictions for Configuring SIP SRST Features Using Back-to-Back User Agent Mode

- See the restrictions documented in the Restrictions for Configuring Cisco Unified SIP SRST section in the Cisco Unified SRST Feature Overview, on page 41.

Information About Configuring SCCP SRST Call Handling

Cisco Unified SRST offers a smaller set of call handling capabilities than Cisco Unified Communications Manager, and much of the configuration for this feature involves enabling existing Cisco Unified Communications Manager or Cisco Unified IP Phone settings.

- H.323 VoIP Call Preservation Enhancements for WAN Link Failures
- Toll Fraud Prevention

H.323 VoIP Call Preservation Enhancements for WAN Link Failures

H.323 VoIP call preservation enhancements for WAN link failures sustain connectivity for H.323 topologies where signaling is handled by an entity, such as Cisco Unified Communications Manager, that is different from the other endpoint and brokers signaling between the two connected parties.

Call preservation is useful when a gateway and the other endpoint (typically a Cisco Unified IP phone) are collocated at the same site and call agent is remote and therefore more likely to experience connectivity failures.

For configuration information see Chapter “Configuring H.323 Gateways” in Cisco IOS H.323 Configuration Guide, Release 12.4T.

Note H.323 is deprecated from IOS XE 17.6.1.

Toll Fraud Prevention

When a Cisco router platform is installed with a voice-capable Cisco IOS Software, appropriate features must be enabled on the platform to prevent potential toll fraud exploitation by unauthorized users. Deploy these features on all Cisco router Cisco Unified Communications applications that process voice calls, such as Cisco Unified Communications Manager Express (Cisco Unified Communications Manager Express), Cisco Survivable Remote Site Telephony (SRST), Cisco Unified Border Element (UBE), Cisco IOS-based router and standalone analog and digital PBX and public-switched telephone network (PSTN) gateways, and Cisco contact-center VoiceXML gateways. For more information about Toll Fraud Prevention, see Toll Fraud Prevention in Cisco Unified Communications Manager Express System Administration Guide.

Information About Configuring SIP SRST Features Using Back-to-Back User Agent Mode

A Cisco Unified SRST system can now support SIP phones with standard-based RFC 3261 feature support locally and across SIP WAN networks. With Cisco Unified SIP SRST, SIP phones can place calls across SIP networks with similar features, as SCCP phones do. For example, most SCCP phone features such as caller ID, speed dial, and redial are supported now on SIP networks, which give users the opportunity to choose SCCP or SIP.

Cisco Unified SIP SRST also uses a back-to-back user agent (B2BUA), which is a separate call agent that has more features than Cisco SIP SRST 3.0, which used a redirect server that only accepted and forwarded calls. The main advantage of a B2BUA call agent is in call forwarding, because it forwards calls on behalf of the phone. In addition, it maintains a presence as call middleman in the call path.

Cisco SIP SRST 3.4 supports the following call combinations:

- SIP phone to SIP phone
- SIP phone to PSTN / router voice port
- SIP phone to SCCP phone

Cisco Unified SIP SRST and Cisco SIP Cisco Unified Communications Manager Express Feature Crossover

The voice register directory number, voice register global, and voice register Pool configuration mode commands are accessible in both Cisco Unified SIP Cisco Unified Communications Manager Express and Cisco Unified SIP SRST modes of operation. However, not all the commands within these modes are intended for use in SIP SRST mode. The following table provides a summary guide to which commands are relevant to the Cisco Unified Communications Manager Express or SRST modes of operation.

For more detailed information, refer to the command reference pages for each of the individual commands.

Note The following table is not all-inclusive; more commands may exist.

Command	Dial Peer	Voice Register Mode	Configurable for Cisco Unified (SIP) Cisco Unified Communications Manager Express and Cisco Unified SIP SRST	Applicable to Cisco Unified (SIP) Cisco Unified Communications Manager Express Only
After-hour exempt	X	DN	X	—
Auto-answer	—	DN	—	X
Call forward	X	DN	X	—

Command	Dial Peer	Voice Register Mode	Configurable for Cisco Unified (SIP) Cisco Unified Communications Manager Express and Cisco Unified SIP SRST	Applicable to Cisco Unified (SIP) Cisco Unified Communications Manager Express Only
Huntstop	X	DN	X	—
Label	—	DN	—	X
Name	—	DN	—	X
Number	X	DN	X	—
Preference	X	DN	X	—
Application	X	Global	X	—
Authenticate	—	Global	—	X
Create	—	Global	—	X
Date-format	—	Global	—	X
DST	—	Global	—	X
External ring	—	Global	X	—
File	—	Global	—	X
Hold-alert	—	Global	—	X
Load	—	Global	—	X
Logo	—	Global	—	X
Max-dn	—	Global	X	—
Max-pool	—	Global	X	—
Max-redirect	—	Global	—	X
Mode	—	Global	X	—
MWI	—	Global	—	X
Reset	—	Global	—	X
Tftp-path	—	Global	—	X
Time zone	—	Global	—	X
Upgrade	—	Global	—	X
Url	—	Global	—	X

Command	Dial Peer	Voice Register Mode	Configurable for Cisco Unified (SIP) Cisco Unified Communications Manager Express and Cisco Unified SIP SRST	Applicable to Cisco Unified (SIP) Cisco Unified Communications Manager Express Only
Voicemail	—	Global	—	X
After-hour exempt	X	Pool	X	—
Application	X	Pool	X	—
Call-forward	—	Pool	X	—
Call-waiting	—	Pool	—	X
Codec	X	Pool	X	—
Description	—	Pool	—	X
Dnd-control	—	Pool	—	X
Dtmf-relay	—	Pool	X	—
Id	—	Pool	X	—
Keep-conference	—	Pool	—	X
Max-pool	—	Pool	X	—
Number	X	Pool	X	—
Preference	X	Pool	X	—
Proxy	X	Pool	X	—
Reset	—	Pool	—	X
Speed-dial	—	Pool	—	X
Template	—	Pool	—	X
Translation-profile	X	Pool	X	—
Type	—	Pool	—	X
Username	—	Pool	—	X
VAD	X	Pool	X	—
Anonymous	—	Template	—	X
Caller-id	—	Template	—	X
Conference	—	Template	—	X

Command	Dial Peer	Voice Register Mode	Configurable for Cisco Unified (SIP) Cisco Unified Communications Manager Express and Cisco Unified SIP SRST	Applicable to Cisco Unified (SIP) Cisco Unified Communications Manager Express Only
Dnd-control	—	Template	—	X
Forward	—	Template	—	X
Transfer	—	Template	—	X

How to Configure Cisco Unified SCCP SRST

Configuring Incoming Calls

Incoming call configuration can include the following tasks:

- Call Forwarding and Rerouting
  - Configuring Call Forwarding During a Busy Signal or No Answer (Optional)
  - Configuring Call Rerouting (Optional)
  - Configuring Call Pickup (Optional)
  - Configuring Transfer Digit Collection Method (Optional)
- Phone Number Conversion and Translation
  - Configuring Global Prefixes(Optional)
  - Enabling Digit Translation Rules (Optional)
  - Enable Translation Profiles (Optional)
  - Verifying Translation Profiles (Optional)
- Hunting and Ringing Timeout Behavior
  - Configuring Dial-Peer and Channel Hunting (Optional)
  - Configuring Busy Timeout (Optional)
  - Configuring the Ringing Timeout Default (Optional)

Configuring Call Forwarding During a Busy Signal or No Answer

Configure the incoming calls that reach busy signal or go unanswered during the Cisco Unified Communications Manager fallback to call forwarding to one or more E.164 numbers.

SUMMARY STEPS

1. **call-manager-fallback**
2. **call-forward busy** *directory-number*
3. **call-forward noan** *directory-number* **timeout** *seconds*
4. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**call-manager-fallback** **Example:** `Router(config)# call-manager-fallback`	Enters call-manager-fallback configuration mode.
Step 2	**call-forward busy** *directory-number* **Example:** `Router(config-cm-fallback)# call-forward busy 50..`	Configures call forwarding to another number when the Cisco IP phone is busy. *directory-number* : Selected directory number representing a fully qualified E.164 number. This number can contain "." wildcard characters that correspond to the right-justified digits in the directory number extension.
Step 3	**call-forward noan** *directory-number* **timeout** *seconds* **Example:** `Router(config-cm-fallback)# call-forward noan 5005 timeout 10`	Configures call forwarding to another number when receiving no answer from the Cisco IP phone. • *directory-number* : Selected directory number representing a fully qualified E.164 number or a local extension number. This number can contain "." wildcard characters that correspond to the right-justified digits in the directory number extension. • **timeout** *seconds* : Sets the waiting time, in seconds, before the call is forwarded to another phone. The seconds range is 3–60000.
Step 4	**exit** **Example:** `Router(config-cm-fallback)# exit`	Exits call-manager-fallback configuration mode.

Example

The following example forwards calls to extension number 5005 when an incoming call reaches a busy or unattended IP phone extension number. Incoming calls ring for 15 seconds before forwarding to extension 5005.

```
call-manager-fallback
call-forward busy 5005
call-forward noan 5005 timeout seconds 15
```

The following example transforms an extension number for a call forwarding when the extension number is busy or unattended. The **call-forward busy** command has an argument of 50.., which prepends the digits 50 to the last two digits of the called extension. The resulting extension is the call forwarding number when the original extension number is busy or unattended. For instance, forwards an incoming call to busy extension 6002 to extension 5002, and forwards an incoming call to busy extension 3442 to extension 5042. Incoming calls ring for 15 seconds before being forwarded.

```
call-manager-fallback
call-forward busy 50..
call-forward noan 50.. timeout seconds 15
```

Configuring Call Rerouting

Note We recommend the **alias** command, which obsoletes the **default-destination** command, instead of the **default-destination** command.

The **alias** command provides a mechanism for rerouting calls to phone numbers that are unavailable during fallback. Up to 50 sets of rerouting alias rules can be created for calls to phone numbers that are unavailable during a Cisco Unified Communications Manager fallback. Sets of alias rules are created using the **alias** command. An alias is activated when a phone registers that has a phone number matching a configured *alternate-number* alias. Under that condition, an incoming call is rerouted to the alternate number. The *alternate-number* argument can be used in multiple**alias** commands, allowing you to reroute multiple different numbers to the same target number.

The configured *alternate-number* must be a specific E.164 phone number or extension that belongs to an IP phone registered on the Cisco Unified SRST router. When an IP phone registers with a number that matches an *alternate-number* , an extra POTS dial peer is created. The destination pattern is set to the initial configured *number-pattern* , and the POTS dial peer voice port is set to match the voice port associated with the *alternate-number* .

If other IP phones register with specific phone numbers within the range of the initial *number-pattern* , the call is routed back to the IP phone rather than to the *alternate-number* (according to normal dial-peer longest-match, preference, and huntstop rules).

Configuring Call Pickup

Configuring the **pickup** command enables the PickUp softkey on all SRST phones. You can then press the PickUp key and answer any currently ringing IP phone that has a DID called number that matches the configured *telephone-number* . This command does not enable the Group PickUp (GPickUp) softkey.

When a user presses the PickUp softkey, SRST searches through all the SRST phones to find a incoming call that has a called number that matches the configured telephone-number. When a match is found, the call is automatically forwarded to the extension number of the phone that requested the Call Pickup.

The SRST **pickup** command is designed to operate in a manner compatible with Cisco Unified Communications Manager.

Note The default phone load on Cisco Unified Communications Manager, Release 4.0(1) for the Cisco 7905 and Cisco 7912 IP phones does not enable the PickUp softkey during fallback. To enable the PickUp softkey on Cisco 7905 and Cisco 7912 IP phones, upgrade your default phone load to Cisco Unified Communications Manager, Version 4.0(1) Sr2. Alternatively, you can upgrade the phone load to cmterm-7905g-sccp.3-3-8.exe or cmterm-7912g-sccp.3-3-8.exe, respectively.

SUMMARY STEPS

1. **call-manager-fallback**
2. **no huntstop**
3. **alias** *tag number-pattern* **to** *alternate-number*
4. **pickup** *telephone number*
5. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**call-manager-fallback** **Example:** `Router(config)# call-manager-fallback`	Enters call-manager-fallback configuration mode.
Step 2	**no huntstop** **Example:** `Router(config-cm-fallback)# no huntstop`	Disables huntstop.
Step 3	**alias** *tag number-pattern* **to** *alternate-number* **Example:** `Router(config-cm-fallback)# alias 1 8005550100 to 5001`	Creates a set rule for rerouting calls to sets of phones that are unavailable during Cisco Unified Communications Manager fallback. • *tag* : Identifier for the alias rule range. Range is from 1to 50. • *number-pattern* : Pattern to match the incoming phone number. This pattern may include wildcards. • **to** : Connects the tag number pattern to the alternate number. • *alternate-number* : Alternate phone number to route incoming calls to match the number pattern. The alternate number has to be a specific extension that belongs to an IP phone that is actively registered on the Cisco Unified SRST router. The alternate phone number can be used in multiple **alias** commands.
Step 4	**pickup** *telephone number* **Example:**	Enables the PickUp softkey on all Cisco Unified IP Phones, allowing an external Direct Inward Dialing (DID) call coming into one extension to be picked up from another

	Command or Action	Purpose
	`Router(config-cm-fallback)# pickup 8005550100`	extension during SRST. The telephone-number argument is the phone number to match an incoming called number.

Step 5	**end**
	Example:
	`Router(config-cm-fallback)# end`
		Returns to privileged EXEC mode.

Example

The **pickup** command is best used with the **alias** command. The following partial output from the **show running-config** command shows the **pickup** command and the **alias** command configured to provide call routing for a pilot number of a hunt group:

```
call-manager-fallback
no huntstop
alias 1 8005550100 to 5001
alias 2 8005550100 to 5002
alias 3 8005550100 to 5003
alias 4 8005550100 to 5004
pickup 8005550100
```

When a DID incoming call to 800 555-0100 is received, the **alias** command routes the call at random to one of the four extensions (5001–5004). Because the **pickup** command is configured, if the DID call rings on extension 5002, the call can be answered from any of the other extensions (5001, 5003, 5004) by pressing the PickUp softkey.

The **pickup** command works by finding a match based on the incoming DID called number. In this example, a call from extension 5004 to extension 5001 (an internal call) does not activate the pickup command because the called number (5001) does not match the configured pickup number (800 555-0100). Thus, the **pickup** command distinguishes between internal and external calls if multiple calls are ringing simultaneously.

Configuring Consultative Transfer

Before Cisco Unified SRST 4.3, the consultative transfer feature played dial tone and collected dialed digits until the digits matched the pattern for consultative transfer, blind transfer, or PSTN transfer blocking. The after-hours blocking criteria was applied after the consultative transfer digit collection and pattern matching.

The new feature modifies the transfer digit-collection process to make it consistent with Cisco Unified Communications Manager. This feature is supported only if the **transfer-system full-consult** command (default) is specified in call-manager-fallback configuration mode and an idle line or channel is available for seizing, digit collection, and dialing.

Requires two lines for consultative transfer. When the transferor party is an octo-line directory number, Cisco Unified SRST selects the next available idle channel on that directory number. If the maximum number of channels of the directory number are in use, consider another idle line on the transferor phone. If the **auto-line** command is configured on the phone, the specified autoline (if idle) takes precedence over other nonauto lines. If no idle line is available on the transferor phone, initiates blind transfer instead of the consultative transfer.

During the consultative transfer, blocks the transferor line to the transferee party on the transferor phone to prevent being stolen by other phones sharing the same directory number. When you press the Transfer softkey for consultative transfer, does not display the Transfer softkey while collecting and dialing the digits on this seized consultative transfer call leg. The method for consultative transfer pattern matching, blind transfer, PSTN transfer blocking, or after-hour blocking criteria remain the same although the manipulation after the matching is different. On meeting the criteria for blind transfer, Cisco Unified SMST stops the consultative transfer call leg, informs the Cisco IOS Software to transfer the call, and then stops the original call bubble. Handles the PARK FAC code in the same way as an incoming call which requires applying a ten-second timer by the Cisco IOS Software.

Note The enhancement, by default, collects the transfer digits from the incoming call leg. If necessary, you can configure the system to collect the transfer digits from the original call leg. See the Configuring Transfer Digit Collection Method section.

The error handling for transfer failure because of transfer blocking or interdigit timer expiration remains. It includes displaying an error message on the prompt line and logging it if “debug ephone error” is enabled, playing a fast-busy or busy tone, and stopping the consultative transfer call leg.

Requires no new configuration to support these enhancements.

Configuring Transfer Digit Collection Method

By default, collects transfer digits from the incoming call leg. To change the transfer digit collection method, perform the following steps.

Before you begin

- Cisco Unified SRST 4.3
- Cisco Unified Communications Manager 6.0
- Cisco IOS Release 12.4(15)XZ

  The Cisco 3200 Series Mobile Access Router does not support SRST.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **call-manager-fallback**
4. **transfer-digit-collect {new-call | orig-call}**
5. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router# enable`	Enables privileged EXEC mode. • Enter your password if prompted.

	Command or Action	Purpose
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**call-manager-fallback** **Example:** `Router(config)# call-manager-fallback`	Enters call-manager-fallback configuration mode.

Step 4	**transfer-digit-collect {new-call | orig-call}**
	Example:
	`Router(config-cm-fallback)# transfer-digit-collect orig-call`
		Selects the digit-collection method used for consultative Call Transfers.
		• **new-call** : Digits are collected from the incoming call leg.
		• **orig-call** : Digits are collected from the original call-leg. It was the default behavior in versions before Cisco Unified SRST 4.3.

Step 5	**end**
	Example:
	`Router(config)# end`
		Returns to privileged EXEC mode.

Example

The following example shows the **transfer-digit-collect** method set to the legacy value of orig-call:

```
!
call-manager-fallback
transfer-digit collect orig-call
!
```

Configuring Global Prefixes

The **dialplan-pattern** command creates a dial-plan pattern that specifies a global prefix for the expansion of abbreviated extension numbers into fully qualified E.164 numbers.

The **extension-pattern** keyword allows extra manipulation of abbreviated extension-number prefix digits. When this keyword and its argument are used, the leading digits of an extension pattern are stripped and replaced by the corresponding leading digits of the dial-plan pattern. This command can be used to avoid Direct Inward Dialing (DID) numbers like 408 555-0101 resulting in 4-digit extensions such as 0101.

Global prefixes are set with the **dialplan-pattern** command. Up to five dial-plan patterns can be created. The **no-reg** keyword provides dialing flexibility and prevents the E.164 numbers in the dial peer from registering to the gatekeeper. You have the option not to register numbers to the gatekeeper so that those numbers can be used for other telephony services.

SUMMARY STEPS

1. **call-manager-fallback**

2. **dialplan-pattern** *tag pattern* **extension-length** *length* [**extension-pattern** *extension-pattern*] [**no-reg**]
3. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**call-manager-fallback** **Example:** `Router(config)# call-manager-fallback`	Enters call-manager-fallback configuration mode.
Step 2	**dialplan-pattern** *tag pattern* **extension-length** *length* [**extension-pattern** *extension-pattern*] [**no-reg**] **Example:** `Router(config-cm-fallback)# dialplan-pattern 1 4085550100 extension-length 3 extension-pattern 4..`	**Note** This example maps all extension numbers 4xx to the PSTN number 40855501xx, so that extension 412 corresponds to 4085550112. Creates a global prefix that can be used to expand the abbreviated extension numbers into fully qualified E.164 numbers. • *tag* : Dial-plan string tag used before a 10-digit phone number. The tag number is 1–5. • *pattern* : Dial-plan pattern, such as the area code, the prefix, and the first one or two digits of the extension number, plus wildcard markers or dots (.) for the remainder of the extension number digits. • **extension-length** : Sets the number of extension digits. • *length* : The number of extension digits. The range is 1–32. • **extension-pattern** : Sets an extension number's leading digit pattern when it is different from the E.164 phone number's leading digits defined in the pattern argument. • *extension-pattern* : The extension number's leading digit pattern. Consists of one or more digits and wildcard markers or dots (.). For example, 5..would include extension 500–599; 5... would include 5000–5999. • **no-reg** : Prevents the E.164 numbers in the dial peer from registering with the gatekeeper.
Step 3	**exit** **Example:** `Router(config-cm-fallback)# exit`	Exits call-manager-fallback configuration mode.

Example

The following example shows how to create dial-plan pattern 1 for extension numbers 101–199 with the phone prefix starting with 4085550. If the following example is set, the router recognizes that 4085550144 matches dial-plan pattern 1. It uses the **extension-length** keyword to extract the last three digits of the number 144 and present this as the caller ID for the incoming call.

```
call-manager-fallback
dialplan-pattern 1 40855501.. extension-length 3 no-reg
```

In the following example, the leading prefix digit for the 3-digit extension numbers is transformed 0–4, so that the extension-number range becomes 400–499:

```
call-manager-fallback
dialplan-pattern 1 40855500.. extension-length 3 extension-pattern 4..
```

In the following example, the **dialplan-pattern** command creates dial-plan pattern 2 for extensions 801–899 with the phone prefix starting with 4085559. As each number in the extension pattern is declared with the number command, two POTS dial peers are created. In the example, they are 801 (an internal office number) and 4085559001 (an external number).

```
call-manager-fallback
dialplan-pattern 2 40855590.. extension-length 3 extension-pattern 8..
```

Enabling Digit Translation Rules

Digit translation rules can be enabled during Cisco Unified Communications Manager fallback. Translation rules are a number-manipulation mechanism that performs operations such as automatically adding phone area codes and prefix codes to dialed numbers.

Note Digit translation rules have many applications and variations. For further information about them, see Cisco IOS Voice Configuration Library.

If you are running Cisco Unified SRST 3.2 and later or Cisco Unified SRST 4.0 and later, use the configuration described in the Enable Translation Profiles section instead of using the **translate** command as described below. Translation Profiles are new to Cisco Unified SRST 3.2 and provide added capabilities.

Translation rules can be used as follows:

- To manipulate the answer number indication (ANI) (calling number) or Dialed Number Identification Service (DNIS) (called number) digits for a voice call.
- To convert a phone number into a different number before the call is matched to an inbound dial peer or before the call is forwarded by the outbound dial peer.

To view the translation rules configured for your system, use the show translation-rule command.

SUMMARY STEPS

1. **call-manager-fallback**
2. **translate {called | calling}** *translation-rule-tag*
3. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**call-manager-fallback** **Example:** `Router(config)# call-manager-fallback`	Enters call-manager-fallback configuration mode.
Step 2	**translate {called | calling}** *translation-rule-tag* **Example:** `Router(config-cm-fallback)# translate called 20`	Applies a translation rule to modify the phone number dialed or received by any Cisco Unified IP Phone user while Cisco Unified Communications Manager fallback is active. • **called** : Applies the translation rule to an outbound call number. • **calling** : Applies the translation rule to an inbound call number. • *translation-rule-tag* : The reference number of the translation rule 1–2147483647.
Step 3	**exit** **Example:** `Router(config-cm-fallback)# exit`	Exits call-manager-fallback configuration mode.

Example

The following example applies translation rule 10 to the calls coming into extension 1111. All inbound calls to 1111 will go to 2222 during Cisco Unified Communications Manager fallback.

```
translation-rule 10
rule 1 1111 2222 abbreviated
exit
call-manager-fallback
translate calling 10
```

The following is a sample configuration of digit translation rule 20, where the priority of the translation rule is 1 (the range is 1–15) and the abbreviated representation of a complete number (1234) is replaced with the number 2345:

```
translation-rule 20
rule 1 1234 2345 abbreviated
exit
```

Enable Translation Profiles

Cisco Unified SRST 3.2 and later and Cisco Unified SRST 4.0 and later support translation profiles. Translation profiles are the suggested way to allow you to group translation rules and provide instructions on how to apply the translation rules to the following:

- Called numbers
- Calling numbers

- Redirected called numbers

In the configuration below, the **voice translation-rule** and the **rule** command allow you to set and define how a number is to be manipulated. The translate command in voice translation-profile mode defines the type of number you are going to manipulate, such as a called, calling, or a redirecting number. Once you have defined your translation profiles, you can then apply the translation profiles in various places, such as dial peers and voice ports. For SCCP SRST, you apply your profiles in Cisco Unified Communications Manager fallback mode.

Cisco IP phones support one incoming and one outgoing translation profile when in SRST mode.

Note For Cisco Unified SRST 3.2 and later versions and Cisco Unified SRST 4.0 and later versions, use the **voice translation-rule** and **translation-profile** commands shown below instead of the translation rule configuration described in the Enabling Digit Translation Rules section. Voice translation rules are a separate feature from translation rules. See the voice translation-rule command in Cisco IOS Voice Command Reference for more information and the VoIP Gateway Trunk and Carrier Based Routing Enhancements documentation for more general information on translation rules and profiles.

SUMMARY STEPS

1. **voice translation-rule** *number*
2. **rule** *precedence/match-pattern/ /replace-pattern/*
3. **exit**
4. **voice translation-profile** *name*
5. **translate {called | calling | redirect-called}** *translation-rule-number*
6. **exit**
7. **call-manager-fallback**
8. **translation-profile {incoming | outgoing}** *name*
9. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**voice translation-rule** *number* **Example:** `Router(config)# voice translation-rule 1`	Defines a translation rule for voice calls and enters voice translation-rule configuration mode. *number*: Number that identifies the translation rule. Range is 1–2147483647.
Step 2	**rule** *precedence/match-pattern/ /replace-pattern/* **Example:** `Router(cfg-translation-rule)# rule 1/^9/ //`	Defines a translation rule. • *precedence*: Priority of the translation rule. Range is 1–15. • *match-pattern*: Stream editor (SED) expression used to match incoming call information. The slash (/) is a delimiter in the pattern.

	Command or Action	Purpose
		• *replace-pattern*: SED expression used to replace the match pattern in the call information. The slash (/) is a delimiter in the pattern.
Step 3	**exit** **Example:** `Router(cfg-translation-rule)# exit`	Exits voice translation-rule configuration mode.
Step 4	**voice translation-profile** *name* **Example:** `Router(config)# voice translation-profile name1`	Defines a translation profile for voice calls. *name*: Name of the translation profile. Maximum length of the voice translation profile name is 31 alphanumeric characters.
Step 5	**translate {called | calling | redirect-called}** *translation-rule-number* **Example:** `Router(cfg-translation-profile)# translate called 1`	Associates a voice translation rule with a voice translation profile. • **called**: Associates the translation rule with called numbers. • **calling**: Associates the translation rule with calling numbers. • **redirect-called**: Associates the translation rule with redirected called numbers. • *translation-rule-number*: The reference number of the translation rule 1–2147483647.
Step 6	**exit** **Example:** `Router(cfg-translation-profile)# exit`	Exits translation-profile configuration mode.
Step 7	**call-manager-fallback** **Example:** `Router(config)# call-manager-fallback`	Enters call-manager-fallback configuration mode.
Step 8	**translation-profile {incoming | outgoing}** *name* **Example:** `Router(config-cm-fallback)# translation-profile outgoing name1`	Assigns a translation profile for incoming or outgoing call legs on a Cisco IP phone. • **incoming**: Applies the translation profile to incoming calls. • **outgoing**: Applies the translation profile to outgoing calls. • *name*: The name of the translation profile.
Step 9	**exit** **Example:**	Exits from call-manager-fallback configuration mode.

	Command or Action	Purpose
	`Router(config-cm-fallback)# exit`	

Example

The following example shows the configuration where a translation profile called name1 is created with two voice translation rules. Rule1 consists of associated calling numbers, and rule2 consists of redirected called numbers. The Cisco Unified IP Phones in SCCP SRST mode are configured with name1.

```
voice translation-profile name1
 translate calling 1
 translate called redirect-called 2

call-manager-fallback
 translation-profile incoming name1
```

Verifying Translation Profiles

Before you begin

To verify translation profiles, perform the following steps.

SUMMARY STEPS

1. **show voice translation-rule** *number*
2. **test voice translation-rule** *number input-test-string* [**test***match-type* [**plan** *match-type*]]

DETAILED STEPS

	Command or Action	Purpose

Step 1	**show voice translation-rule** *number*
	Example:
	`Router# show voice translation-rule 6`
	`Translation-rule tag: 6`
	`Rule 1:`
	`Match pattern: 65088801..`
	`Replace pattern: 6508880101`
	`Match type: none Replace type: none`
	`Match plan: none Replace plan: none`
		Use this command to verify the translation rules that you have defined for your translation profiles.

Step 2	**test voice translation-rule** *number input-test-string* [**test***match-type* [**plan** *match-type*]]
	Example:
	`Router(config)# voice translation-rule 5`
	`Router(cfg-translation-rule)# rule 1 /201/ /102/`
	`Router(cfg-translation-rule)# end`
	`Router# test voice translation-rule 5 2015550101`
	`Matched with rule 5`
	`Original number:2015550101 Translated`
		Use this command to test your translation profiles. See the test voice translation-rule command in Cisco IOS Voice Command Reference for more information.

	Command or Action	Purpose

	`number:1025550101`
	`Original number type: none Translated number`
	`type: none`
	`Original number plan: none Translated number`
	`plan: none`	

Configuring Dial-Peer and Channel Hunting

Dial-peer hunting, the search through a group of dial peers for an available phone line, is disabled during Cisco Unified Communications Manager fallback by default. To enable dial-peer hunting, use the no huntstop command. For more information about dial-peer hunting, see Cisco IOS Voice Configuration Library.

If you have a dual-line phone configuration, see the Configuring Dual-Line Phones section. Keep incoming calls from hunting to the second channel if the first channel is busy or does not answer by using the channel keyword in the huntstop command.

Channel huntstop also prevents situations in which a call can ring for 30 seconds on the first channel of a line with no person available to answer and then ring for another 30 seconds on the second channel before rolling over to another line.

SUMMARY STEPS

1. **call-manager-fallback**
2. **huntstop [channel]**
3. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**call-manager-fallback** **Example:** `Router(config)# call-manager-fallback`	Enters call-manager-fallback configuration mode.
Step 2	**huntstop [channel]** **Example:** `Router(config-cm-fallback)# huntstop channel`	Sets the huntstop attribute for the dial peers associated with the Cisco Unified IP Phone dial peers created during Cisco Unified Communications Manager fallback. • For dual-line configurations, the **channel** keyword keeps incoming calls from hunting to the second channel if the first channel is busy or does not answer.
Step 3	**exit** **Example:** `Router(config-cm-fallback)# exit`	Exits call-manager-fallback configuration mode.

Example

The following example disables dial-peer hunting during Cisco Unified Communications Manager fallback and hunting to the secondary channels in dual-line phone configurations:

```
call-manager-fallback
no huntstop channel
```

Configuring Busy Timeout

This task sets the timeout value for Call Transfers to busy destinations. The busy timeout value is the amount of time that can elapse after a transferred call reaches a busy signal before the call is disconnected.

SUMMARY STEPS

1. **call-manager-fallback**
2. **timeouts busy** *seconds*
3. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**call-manager-fallback** **Example:** `Router(config)# call-manager-fallback`	Enters call-manager-fallback configuration mode.
Step 2	**timeouts busy** *seconds* **Example:** `Router(config-cm-fallback)# timeouts busy 20`	Sets the amount of time for disconnecting the calls before transferring to busy destinations. *seconds* : Number of seconds. Range is 0–30. Default is 10. **Note** This command sets the busy timeout only for calls before transferring to busy destinations and does not affect the timeout for calls that directly dial busy destinations.
Step 3	**exit** **Example:** `Router(config-cm-fallback)# exit`	Exits call-manager-fallback configuration mode.

Example

The following example sets a timeout of 20 seconds for transferring calls to busy destinations:

```
call-manager-fallback
timeouts busy 20
```

Configuring the Ringing Timeout Default

The ringing timeout default is the length of time for which a phone can ring with no answer before returning a disconnect code to the caller. This timeout prevents hung calls received over interfaces such as Foreign Exchange Office (FXO) that do not have forward-disconnect supervision. It is used only for extensions that do not have no-answer call forwarding enabled.

SUMMARY STEPS

1. **call-manager-fallback**
2. **timeouts ringing** *seconds*
3. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**call-manager-fallback** **Example:** `Router(config)# call-manager-fallback`	Enters call-manager-fallback configuration mode.
Step 2	**timeouts ringing** *seconds* **Example:** `Router(config-cm-fallback)# timeouts ringing 30`	Sets the ringing timeout default, in seconds. The range is from 5 to 60000. There is no default value.
Step 3	**exit** **Example:** `Router(config-cm-fallback)# exit`	Exits call-manager-fallback configuration mode.

Example

The following example sets the ringing timeout default to 30 seconds:

```
call-manager-fallback
timeouts ringing 30
```

Configuring Outgoing Calls

Configuring Local and Remote Call Transfer

Configure the Cisco Unified SRST to allow Cisco Unified IP Phones to transfer phone calls from outside the local IP network to another Cisco Unified IP Phone. By default, all Cisco Unified IP Phone directory numbers or virtual voice ports are allowed as transfer targets. A maximum of 32 transfer patterns can be entered.

Call Transfer configuration is performed using the **transfer-pattern** command.

SUMMARY STEPS

1. **call-manager-fallback**
2. **transfer-pattern** *transfer-pattern*
3. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**call-manager-fallback** **Example:** `Router(config)# call-manager-fallback`	Enters call-manager-fallback configuration mode.
Step 2	**transfer-pattern** *transfer-pattern* **Example:** `Router(config-cm-fallback)# transfer-pattern 52540..`	Enables the transfer of a call from a non-IP phone number to another Cisco Unified IP Phone on the same IP network using the specified transfer pattern. *transfer-pattern* : String of digits for permitted call Transfers. Wildcards are permitted.
Step 3	**exit** **Example:** `Router(config-cm-fallback)# exit`	Exits call-manager-fallback configuration mode.

Example

In the following example, the transfer-pattern command permits transfers from a non-IP phone number to any Cisco Unified IP Phone on the same IP network with a number in range 5550100–5550199:

```
call-manager-fallback
transfer-pattern 55501..
```

Enabling Consultative Call Transfer and Forward Using H.450.2 and H.450.3 with Cisco Unified SRST 3.0

Consultative Call Transfer using H.450.2 adds support for initiating Call Transfers and call forwarding on a call leg using the ITU-T H.450.2 and ITU-T H.450.3 standards. Call Transfers and call forwarding using H.450.2 and H.450.3 can be blind or consultative. A blind Call Transfer or blind call forward is one in which the transferring or forwarding phone connects the caller to a destination line before a ringing tone begins. Consultative transfer is one in which the transferring or forwarding party either connects the caller to a ringing phone (ringback heard) or speaks with the third party before connecting the caller to the third party.

Note For Cisco Unified SRST 3.1 and later versions and Cisco Unified SRST 4.0 and later versions, Call Transfer and call forward using H.450.2 is supported automatically with the default session application.

Before you begin

Call Transfer with consultation is available only when an IP phone supports a second line or call instance. Please see the dual-line keyword in the max-dn command.

All voice gateway routers in the VoIP network must support the H.450 standard.

All voice gateway routers in the VoIP network must be running the following software:

- Cisco IOS Release 12.3(2)T or a later release
- Cisco Unified SRST 3.0

Restrictions

Does not implement a H.450.12 Supplementary Services Capabilities Exchange among routers.

SUMMARY STEPS

1. **call-manager-fallback**
2. **call-forward pattern** *pattern*
3. **transfer-system {blind | full-blind | full-consult | local-consult}**
4. **transfer-pattern** *transfer-pattern*
5. **exit**
6. (Optional) **voice service voip**
7. (Optional) **h323**
8. (Optional) **h450 h450-2 timeout {T1 | T2 | T3 | T4}***milliseconds*
9. (Optional) **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**call-manager-fallback** **Example:** `Router(config)# call-manager-fallback`	Enters call-manager-fallback configuration mode.
Step 2	**call-forward pattern** *pattern* **Example:** `Router(config-cm-fallback)# call-forward pattern 4...`	Specifies the H.450.3 standard for a call forwarding. *pattern* : Digits to match for a call forwarding using the H.450.3 standard. If an incoming calling-party number matches the pattern, it can be forwarded using the H.450.3 standard. A pattern of .T forwards all calling parties using the H.450.3 standard.

Step 3	**transfer-system {blind | full-blind | full-consult | local-consult}**
	Example:
	`Router(config-cm-fallback)# transfer-system full-consult`
		Not supported if the transfer-to destination is on the Cisco ATA, Cisco VG224, or an SCCP-controlled FXS port.
		Defines the call-transfer method for all lines served by the Cisco Unified SRST router.
		• **blind** : Calls are transferred without consultation with a single phone line using the Cisco proprietary method.

	Command or Action	Purpose
		Note We do not recommend the blind keyword. Use either the full-blind or full-consult keyword instead. • **full-blind** : Calls are transferred without consultation using H.450.2 standard methods. • **full-consult** : Calls are transferred with consultation using a second phone line if available. The calls fall back to full-blind if the second line is unavailable. • **local-consult** : Calls are transferred with local consultation using a second phone line if available. The calls fall back to blind for nonlocal consultation or nonlocal transfer target.
Step 4	**transfer-pattern** *transfer-pattern* **Example:** `Router(config-cm-fallback)# transfer-pattern 52540..`	Allows transfer of the phone calls by Cisco Unified IP Phones to specified phone number patterns. *transfer-pattern* : String of digits for permitted Call Transfers. Wildcards are allowed.
Step 5	**exit** **Example:** `Router(config-cm-fallback)# exit`	Exits call-manager-fallback configuration mode. **Timesaver** : Before exiting call-manager-fallback configuration mode, configure any other parameters that you must set for the entire Cisco Unified SRST phone network.
Step 6	(Optional) **voice service voip** **Example:** `Router(config)# voice service voip`	Enters voice service configuration mode.
Step 7	(Optional) **h323** **Example:** `Router(conf-voi-serv)# h323`	Enters H.323 voice service configuration mode.
Step 8	(Optional) **h450 h450-2 timeout {T1 | T2 | T3 | T4}***milliseconds* **Example:** `Router(conf-serv-h323)# h450 h450-2 timeout T1 750`	Sets timeouts for supplementary service timers, in milliseconds. This command is used primarily when the default settings for these timers do not match your network delay parameters. See the ITU-T H.450.2 specification for more information on these timers. • **T1** : Timeout value to wait to identify response. Default is 2000. • **T2** : Timeout value to wait for a call setup. Default is 5000. • **T3** : Timeout value to wait to initiate response. Default is 5000.

	Command or Action	Purpose
		• **T4** : Timeout value to wait for setup of response. Default is 5000. • *milliseconds* : Number of milliseconds. Range is 500–60000.
Step 9	(Optional) **end** **Example:** `Router(conf-serv-h323)# end`	Returns to privileged EXEC mode.

Example

The following example specifies transfer with consultation using the H.450.2 standard for all IP phones serviced by the Cisco Unified SRST router:

```
dial-peer voice 100 pots
destination-pattern 9.T
port 1/0/0
dial-peer voice 4000 voip
destination-pattern 4…
session-target ipv4:10.1.1.1
call-manager-fallback
transfer-pattern 4…
transfer-system full-consult
```

The following example enables call forwarding using the H.450.3 standard:

```
dial-peer voice 100 pots
 destination-pattern 9.T
 port 1/0/0
!
dial-peer voice 4000 voip
  destination-pattern 4
  session-target ipv4:10.1.1.1
!
call-manager-fallback
  call-forward pattern 4
```

Enabling Analog Transfer Using Hookflash and the H.450.2 Standard with Cisco Unified SRST 3.0 or Earlier

Analog Call Transfer using hookflash and the H.450.2 standard allows analog phones to transfer calls with consultation by using the hookflash to initiate transfer. Hookflash refers to the short on-hook period generated by a telephone-like device during a call to indicate that the phone is attempting to perform the dial-tone recall from a PBX. Uses Hookflash to perform Call Transfer. For example, a hookflash occurs when a caller quickly taps once on the button in the cradle of an analog phone's handset.

This feature requires installation of a Tool Command Language (Tcl) script. Download the script app-h450-transfer.tcl from the Cisco Software Center at http://www.cisco.com/cgi-bin/tablebuild.pl/ip-iostsp and copied to a TFTP server that is available to the Cisco Unified SRST router or copied to the flash memory on the Cisco Unified SRST router. To apply this script globally to all dial peers, use the **call application**

global command in global configuration mode. The Tcl script has parameters to which you can pass values using attribute-value (AV) pairs in the **call application voice** command. The parameter that applies to this feature is as follows:

- **delay-time** : Speeds up or delays the setting up of the consultation call during a Call Transfer from an analog phone using a delay timer. On collecting all digits, the delay timer starts. The call setup to the receiving party does not begin until the delay timer expires. If the transferring party goes on-hook before the delay timer expires, the transfer is considered blind transfer rather than consultative transfer. If the transferring party goes on-hook after the delay timer expires, either while the destination phone is ringing or after the destination party answers, the transfer is considered consultative transfer.

In addition to the Tcl script, a ReadMe file describes the script and the configurable attribute-value pairs. Read this file whenever you download a new version of the script because it may contain more script-specific information, such as configuration parameters and user interface descriptions.

Note For Cisco Unified SRST 3.1 and later versions and Cisco Unified SRST 4.0 and later versions, Call Transfer using H.450.2 is supported automatically with the default session application.

Restrictions

- When consultative transfer is made by an analog FXS phone using hookflash, the consultation call itself cannot be further transferred (that is, it cannot become a recursive or chained transfer) until after the initial transfer operation is completed and the transferee and transfer-to parties are connected. After the initial Call Transfer operation is completed and the transferee and transfer-to parties are now the only parties in the call, the transfer-to party may further transfer the call.
- Call Transfer with consultation is not supported for Cisco ATA-186, Cisco ATA-188, and Cisco IP Conference Station 7935. Transfer attempts from these devices are executed as blind transfers.

Before you begin

Download the H.450 Tcl script named app-h450-transfer.Tcl from the Cisco Software Center. The following versions of the script are available:

- app-h450-transfer.2.0.0.2.tcl for Cisco IOS Release 12.2(11)YT1 and later releases
- app-h450-transfer.2.0.0.1.tcl for Cisco IOS Release 12.2(11)YT

All voice gateway routers in the VoIP network must support H.450 and be running the following software:

- Cisco IOS Release 12.2(11)YT or a later release
- Cisco Unified SRST V3.0 or a lower version
- Tcl IVR 2.0
- H.450 Tcl script (app-h450-transfer.Tcl)

Note You can continue to use the app-h450-transfer.2.0.0.1.tcl script if you install Cisco IOS Release 12.2(11)YT1 or later, but you cannot use the app-h450-transfer.2.0.0.2.tcl script with a release of Cisco IOS Software that is earlier than Cisco IOS Release 12.2(11)YT1.

SUMMARY STEPS

1. **call application voice** *application-name location*
2. (Optional) **call application voice** *application-name* **language** *number language*
3. **call application voice** *application-name* **set-location** *language category location*
4. (Optional) **call application voice** *application-name* **delay-time** *seconds*
5. **dial-peer voice** *number* **pots**
6. **application** *application-name*
7. **exit**
8. **dial-peer voice** *number* **voip**
9. **application** *application-name*
10. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**call application voice** *application-name location* **Example:** `Router(config)# call application voice transfer_app flash:app-h450-transfer.tcl`	Loads the Tcl script and specifies its application name. • *application-name* : User-defined name for the IVR application. This name does not have to match the script filename. • *location* : Script directory and filename in URL format. For example, flash memory (flash:filename), a TFTP (tftp://../filename), or an HTTP server (http://../filename) are valid locations.
Step 2	(Optional) **call application voice** *application-name* **language** *number language* **Example:** `Router(config)# call application voice transfer_app language 1 en`	Sets the language for dynamic prompts by the application. • *application-name* : IVR application name that was assigned in Step 1. • *number* : Specify the number of languages for the audio files for the IVR application. • *language* : Two-character code that specifies the language of the prompts. Valid entries are en (English:default), sp (Spanish), ch (Chinese), or aa (all).
Step 3	**call application voice** *application-name* **set-location** *language category location* **Example:** `Router(config)# call application voice transfer_app set-location en 0 flash:/prompts`	Defines the location and category of the audio files that are used by the application for dynamic prompts. • *application-name* : Name of the Tcl IVR application. • *language* : Two-character code to specify the language of the prompts. Valid entries are en (English: default), sp (Spanish), ch (Chinese), or aa (all). • *category* : Category group (0–4) for the audio files from this location. The value 0 means all categories. .

	Command or Action	Purpose
		• *location* : URL of the directory that contains the language audio files in the application, without filenames. Flash memory (flash) or a directory on a server (TFTP, HTTP, or RTSP) are all valid. Prompts are required for Call Transfer from analog FXS phones. No prompts are needed for Call Transfer from IP phones.
Step 4	(Optional) **call application voice** *application-name* **delay-time** *seconds* **Example:** `Router(config)# call application voice transfer_app delay-time 1`	Sets the delay time for consultation call setup for an analog phone that is making a Call Transfer using the H.450 application. This command passes a value to the Tcl script by using an attribute-value (AV) pair. • *seconds* : Number of seconds to delay call setup. Range is 1–10. Default is 2. Delay of more than 2 seconds is noticeable to users. For more information about attribute-value pairs and the Tcl script for H.450 Call Transfer and forwarding, see the ReadMe file that accompanies the script.
Step 5	**dial-peer voice** *number* **pots** **Example:** `Router(config)# dial-peer voice 25 pots`	Enters dial-peer configuration mode to configure a POTS dial peer.
Step 6	**application** *application-name* **Example:** `Router(config-dial-peer)# application transfer_app`	Loads the application named in Step 1 onto the dial peer.
Step 7	**exit** **Example:** `Router(config-dial-peer)# exit`	Exits dial-peer configuration mode. **Timesaver** : Before exiting dial-peer configuration mode, configure any other dial-peer parameters that you must set for this dial peer.
Step 8	**dial-peer voice** *number* **voip** **Example:** `Router(config)# dial-peer voice 29 voip`	Enters dial-peer configuration mode to configure a VoIP dial peer.
Step 9	**application** *application-name* **Example:** `Router(config-dial-peer)# application transfer_app`	Loads the application named in Step 1 onto the dial peer.
Step 10	**exit** **Example:**	Exits dial-peer configuration mode.

	Command or Action	Purpose
	`Router(config-dial-peer)# exit`	**Timesaver** : Before exiting dial-peer configuration mode, configure any other dial-peer parameters that you must set for this dial peer.

Example

The following example enables the H.450 Tcl script for analog transfer using hookflash and sets delay time of 1 second:

```
call application voice transfer_app flash:app-h450-transfer.tcl
call application voice transfer_app language 1 en
call application voice transfer_app set-location en 0 flash:/prompts
call application voice transfer_app delay-time 1
!
dial-peer voice 25 pots
destination-pattern 9.T
port 1/0/0
application transfer_app
!
dial-peer voice 29 voip
destination-pattern 4…
session-target ipv4:10.1.10.1
application transfer_app
```

Configuring Trunk Access Codes

Note Configure trunk access codes only if your normal network dial-plan configuration prevents you from configuring a permanent POTS voice dial peer to provide trunk access for use during fallback. If you already have local PSTN ports configured with the appropriate access codes provided by dial peers (for example, dial 9 to select an FXO PSTN line), this configuration is not needed.

Trunk access codes provide IP phones with access to the PSTN during Cisco Unified Communications Manager fallback by creating POTS voice dial peers that are active during Cisco Unified Communications Manager fallback only. These temporary dial peers, which can be matched to voice ports (BRI, E&M, FXO, and PRI), allow Cisco Unified IP Phones access to trunk lines during Cisco Unified Communications Manager mode. When Cisco Unified SRST is active, all PSTN interfaces of the same type are treated as equivalent, and any port may be selected to place the outgoing PSTN call.

Trunk access codes are created using the **access-code** command.

SUMMARY STEPS

1. **call-manager-fallback**
2. **access-code** { { **fxo** | **e&m** } *dial-string* | { **bri** | **pri** } *dial-string* [**direct-inward-dial**] }
3. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**call-manager-fallback** **Example:** `Router(config)# call-manager-fallback`	Enters call-manager-fallback configuration mode.

Step 2	**access-code** { { **fxo** | **e&m** } *dial-string* | { **bri** | **pri** } *dial-string* [**direct-inward-dial**] }
	Example:
	`Router(config-cm-fallback)# access-code e&m 8`
		Configures trunk access codes for each type of line so that the Cisco Unified IP Phones can access the trunk lines only in Cisco Unified Communications Manager fallback mode when the Cisco Unified SRST is enabled.
		• **fxo** : Enables a Foreign Exchange Office (FXO) interface.
		• **e&m** : Enables an analog Ear and Mouth (E&M) interface.
		• *dial-string* : String of characters that sets up dial access codes for each specified line type by creating dial peers. The dial-string argument is used to set up temporary dial peers for each specified line type.
		• **bri** : Enables a BRI interface.
		• **pri** : Enables a PRI interface.
		• **direct-inward-dial** : Enables Direct Inward Dialing (DID) on the POTS dial peer.

Step 3	**exit** **Example:** `Router(config-cm-fallback)# exit`	Exits call-manager-fallback configuration mode.

Example

The following example creates access code number 8 for BRI and enables DID on the POTS dial peer:

```
call-manager-fallback
access-code bri 8 direct-inward-dial
```

Configuring Interdigit Timeout Values

Configuring interdigit timeout values involves specifying how long, in seconds, all Cisco Unified IP Phones attached to a Cisco Unified SRST router are to wait after an initial digit or a subsequent digit is dialed. The **timeouts interdigit** timer is enabled when a caller enters a digit and is restarted each time the caller enters subsequent digits until the destination address is identified. If the configured timeout value is exceeded before the destination address is identified, a tone sounds and the call is stopped.

SUMMARY STEPS

1. **call-manager-fallback**
2. (Optional) **timeouts interdigit** *seconds*
3. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**call-manager-fallback** **Example:** `Router(config)# call-manager-fallback`	Enters call-manager-fallback configuration mode.
Step 2	(Optional) **timeouts interdigit** *seconds* **Example:** `Router(config-cm-fallback)# timeouts interdigit 5`	Configures the interdigit timeout value for all Cisco IP phones that are attached to the router. *seconds* : Interdigit timeout duration, in seconds, for all Cisco Unified IP Phones. Valid entries are integers from 2 to 120.
Step 3	**exit** **Example:** `Router(config-cm-fallback)# exit`	Exits call-manager-fallback configuration mode.

Example

The following example sets the interdigit timeout value to 5 seconds for all Cisco Unified IP Phones. In this example, 5 seconds are the elapsed time after which an incompletely dialed number times out. For example, a caller who dials nine digits (408555010) instead of the required ten digits (4085550100) will hear a busy tone after the second timeout elapses.

```
call-manager-fallback
timeouts interdigit 5
```

Configuring Class of Restriction

The class of restriction (COR) functionality provides the ability to deny a certain call attempt on the basis of the incoming and outgoing class of restrictions that are provisioned on the dial peers. This functionality provides flexibility in the network design, allows you to block calls (for example, calls to 900 numbers), and applies different restrictions to call attempts from different originators. The **cor** command sets the dial-peer COR parameter for the dial peers associated with the directory numbers that are created during Cisco Unified Communications Manager fallback.

You can have up to 20 COR lists for each incoming and outgoing call. A default COR is assigned to directory numbers that do not match the COR list numbers or number ranges. An assigned COR is invoked for the dial peers and created for each directory number automatically during Cisco Unified Communications Manager fallback registration.

If a COR is applied on an incoming dial peer (for incoming calls) and it is a superset of or is equal to the COR applied to the outgoing dial peer (for outgoing calls), the call goes through. Voice ports determine whether a call is considered incoming or outgoing. If you hook up a phone to an FXS port on a Cisco Unified SRST router and try to call from that phone, the call will be considered an incoming call to the router and voice port. If you call the FXS phone, consider it as an outgoing call.

By default, an incoming call leg has the highest COR priority; the outgoing call leg has the lowest priority. If there is no COR configuration for incoming calls on a dial peer, you can call from a phone that is attached to the dial peer, so that the call goes out of any dial peer regardless of the COR configuration on that dial peer. The following table describes the call functionality that is based on your COR lists configuration.

COR List on Incoming Dial Peer	COR List on Outgoing Dial Peer	Result
No COR	No COR	The call succeeds.
No COR	COR list applied for outgoing calls	The call succeeds. By default, the incoming dial peer has the highest COR priority when no COR is applied. If you apply no COR for an incoming call leg to a dial peer, the dial peer can call of any other dial peer regardless of the COR configuration on the outgoing dial peer.
COR list applied for incoming calls	No COR	The call succeeds. By default, the outgoing dial peer has the lowest priority. Because there are some COR configurations for incoming calls on the incoming or originating dial peer, it is a superset of the outgoing call’s COR configuration for the outgoing or stopping dial peer.
COR list applied for incoming calls (superset of a COR list applied for outgoing calls on the outgoing dial peer)	COR list applied for outgoing calls (subsets of a COR list applied for incoming calls on the incoming dial peer)	The call succeeds. The COR list for incoming calls on the incoming dial peer is a superset of the COR list for outgoing calls on the outgoing dial peer.
COR list applied for incoming calls (subset of a COR list applied for outgoing calls on the outgoing dial peer)	COR list applied for outgoing calls (supersets of a COR list applied for incoming calls on the incoming dial peer)	The call does not succeed. The COR list for incoming calls on the incoming dial peer is not a superset of the COR list for outgoing calls on the outgoing dial peer.

SUMMARY STEPS

1. **call-manager-fallback**
2. **cor {incoming | outgoing}** *cor-list-name [cor-list-number starting-number - ending-number* | **default**]
3. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**call-manager-fallback** **Example:** `Router(config)# call-manager-fallback`	Enters call-manager-fallback configuration mode.
Step 2	**cor {incoming | outgoing}** *cor-list-name [cor-list-number starting-number - ending-number* | **default**] **Example:** `Router(config-cm-fallback)# cor outgoing LockforPhoneC 1 5010 - 5020`	Configures a COR on dial peers that are associated with directory numbers. • **incoming** : COR list to be used by incoming dial peers. • **outgoing** : COR list to be used by outgoing dial peers. • *cor-list-name* : COR list name. • *cor-list-number* : COR list identifier. You can create maximum 20 COR lists, comprising of incoming or outgoing dial peers. The first six COR lists are applied to range of directory numbers. Assign the COR configuration to the default COR list if the directory numbers do not have a COR configuration. • *starting-number- ending-number:* : Directory number range; for example, 2000–2025. • **default** : Instructs the router to use an existing default COR list.
Step 3	**exit** **Example:** `Router(config-cm-fallback)# exit`	Exits call-manager-fallback configuration mode.

Example

The following example shows how to set a dial-peer COR parameter for outgoing calls to the Cisco Unified IP Phone dial peers and directory numbers that are created during fallback:

```
call-manager-fallback
cor outgoing LockforPhoneC 1 5010 - 5020
```

The following example shows how to set the dial-peer COR parameter for incoming calls to the Cisco IP phone dial peers and directory numbers in the default COR list:

```
call-manager-fallback
cor incoming LockforPhoneC default
```

The following example shows creation of a sub- and super-COR sets. First, create a custom dial-peer COR with declared names under it:

```
dial-peer cor custom
name 911
name 1800
name 1900
name local_call
```

The following configuration example creates the COR lists and applies to the dial peer:

```
dial-peer cor list call911
member 911
dial-peer cor list call1800
member 1800
dial-peer cor list call1900
member 1900
dial-peer cor list calllocal
member local_call
dial-peer cor list engineering
member 911
member local_call
dial-peer cor list manager
member 911
member 1800
member 1900
member local_call
dial-peer cor list hr
member 911
member 1800
member local_call
```

The following example configures five dial peers for destination numbers 734…., 1800…….,1900……., 316…., and 911. A COR list is applied to each of the dial peers.

```
dial-peer voice 1 voip
destination pattern 734....
session target ipv4:10.1.1.1
cor outgoing calllocal
dial-peer voice 2 voip
destination pattern 1800.......
session target ipv4:10.1.1.1
cor outgoing call1800
dial-peer voice 3 pots
destination pattern 1900.......
port 1/0/0
cor outgoing call1900
dial-peer voice 5 pots
destination pattern 316....
port 1/1/0
! No COR is applied.
dial-peer voice 4 pots
destination pattern 911
port 1/0/1
cor outgoing call911
```

Finally, the COR list is applied to the individual phone numbers.

```
call-manager-fallback
max-conferences 8
cor incoming engineering 1 1001 - 1001
cor incoming hr 2 1002 - 1002
cor incoming manager 3 1003 - 1008
```

The sample configuration allows for the following:

- Extension 1001 to call 734... numbers, 911, and 316....
- Extension 1002 to call 734..., toll-free numbers, 911, and 316....
- Extension 1003–1008 to call all the possible Cisco Unified SRST router numbers.

- All extensions to call 316....

Call Blocking (Toll Bar) Based on Time of Day and Day of Week or Date

Call blocking to prevent unauthorized use of phones is implemented by matching a pattern of specified digits during specified time of day and day of the week or date. Specify up to 32 patterns of digits. Supports call blocking on IP phones only and not on analog Foreign Exchange Station (FXS) phones.

When you call to digits that match a pattern for call blocking during a defined time period for a call blocking, fast busy signal plays for approximately 10 seconds. The call stops and places the line back in on-hook status.

In SRST (call-manager-fallback configuration) mode, there is no phone- or pin-based exemption to after-hours call blocking.

SUMMARY STEPS

1. **call-manager-fallback**
2. **after-hours block pattern** *tag pattern* [**7-24**]
3. **after-hours day** *day start-time stop-time*
4. **after-hours date** *month date start-time stop-time*
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**call-manager-fallback** **Example:** `Router(config)# call-manager-fallback`	Enters call-manager-fallback configuration mode.
Step 2	**after-hours block pattern** *tag pattern* [**7-24**] **Example:** `Router(config-cm-fallback)# after-hours block pattern 1 91900`	For blocking, defines a pattern of outgoing digits. Define up to 32 patterns, using individual commands. • If you specify the 7–24 keyword, always blocks the pattern, 7 days a week, 24 hours a day. • If you do not specify the 7–24 keyword, blocks the pattern during the days and dates as defined in the after-hours day and after-hours date commands.
Step 3	**after-hours day** *day start-time stop-time* **Example:** `Router(config-cm-fallback)# after-hours day mon 19:00 7:00`	Defines a recurring time period for the day of the week during which calls are blocked to outgoing dial patterns that are defined using the **after-hours block pattern** command. • *day* : Day of the week abbreviation. The following are valid day abbreviations: sun, mon, tue, wed, thu, fri, sat. • *start-time stop-time* : Beginning and ending times for call blocking, in an HH:MM format using a 24-hour clock. If the stop time is smaller value than the start time, the stop time occurs on the day following the

	Command or Action	Purpose
		start time. For example, “mon 19:00 07:00” means “from Monday at 7 p.m. until Tuesday at 7 a.m.”.
Step 4	**after-hours date** *month date start-time stop-time* **Example:** `Router(config-cm-fallback)# after-hours date jan 1 0:00 0:00`	Defines a recurring time period for the month and date for blocking calls to outgoing dial patterns defined in the after-hours block pattern command. • *month* : Month abbreviation. The following are valid month abbreviations: jan, feb, mar, apr, may, jun, jul, aug, sep, oct, nov, dec. • *date* : Date of the month. Range is 1–31. • *start-time stop time* : Beginning and ending times for call blocking, in an HH:MM format using a 24-hour clock. The stop time must be larger than the start time. Value 24:00 is not valid. If you enter 00:00 as stop time, it changes to 23:59. If you enter 00:00 for both start time and stop time, blocks call for the entire 24-hour period on the specified date.
Step 5	**exit** **Example:** `Router(config-cm-fallback)# exit`	Exits call-manager-fallback configuration mode.

Example

The following example defines several patterns of digits for which blocks outgoing calls. Patterns 1 and 2, blocks call to external numbers that begin with “1” and “011”:

- On Monday through Friday before 7 a.m. and after 7 p.m.
- On Saturday before 7 a.m. and after 1 p.m.
- All day Sunday.

Pattern 3 blocks call to 900 numbers 7 days a week, 24 hours a day.

```
call-manager-fallback
after-hours block pattern 1 91
after-hours block pattern 2 9011
after-hours block pattern 3 91900 7-24
after-hours block day mon 19:00 07:00
after-hours block day tue 19:00 07:00
after-hours block day wed 19:00 07:00
after-hours block day thu 19:00 07:00
after-hours block day fri 19:00 07:00
after-hours block day sat 13:00 12:00
after-hours block day sun 12:00 07:00
```

How to Configure Cisco Unified SIP SRST

Configuring SIP Phone Features

After setting the voice register Pool, the procedure adds optional features to increase functionality. Some features are per Pool or globally.

In **voice register pool** configuration, you can now configure several new options per Pool (a Pool can be one phone or a group of phones). There is also a new voice register global configuration mode for Cisco Unified SIP SRST. In the voice register global mode, you can globally assign characteristics to phones.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register global** *tag*
4. **max-pool** *max-voice-register-pools*
5. **application** *application-name*
6. **external-ring {bellcore-dr1 | bellcore-dr2 |bellcore-dr3 | bellcore-dr4 | bellcore-dr5}**
7. **exit**
8. **voice register pool** *tag*
9. **no vad**
10. **codec** *codec-type [bytes]*
11. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice register global** *tag* **Example:** `Router(config)# voice register global 12`	Enters voice register global configuration mode to set global parameters for all supported Cisco SIP IP phones in a Cisco Unified SIP SRST environment.
Step 4	**max-pool** *max-voice-register-pools* **Example:** `Router(config-register-global)# max-pool 10`	Set the maximum number of supported SIP voice register Pools in a Cisco Unified SIP SRST environment. The max-voice-register-pools argument represents the maximum number of SIP voice register Pools supported by the Cisco Unified SIP SRST router. The upper limit of

	Command or Action	Purpose
		voice register Pools is version- and platform-dependent; see Cisco IOS command-line interface (CLI) help. Default is 0.
Step 5	**application** *application-name* **Example:** `Router(config-register-global)# application global_app`	Selects the session-level application for all dial peers associated with SIP phones. Use the application-name argument to define specific interactive voice response (IVR) application.

Step 6	**external-ring {bellcore-dr1 | bellcore-dr2 |bellcore-dr3 | bellcore-dr4 | bellcore-dr5}**
	Example:
	`Router(config-register-global)# external-ring bellcore-dr1`
		Specifies the type of ring sound on Cisco SIP or Cisco SCCP IP phones for external calls. Each bellcore-dr 1–5 keyword supports standard distinctive ringing patterns as defined in the standard GR-506-CORE, LSSGR: Signaling for Analog Interfaces.

Step 7	**exit** **Example:** `Router(config-register-global)# exit`	Exits voice register global configuration mode.
Step 8	**voice register pool** *tag* **Example:** `Router(config)# voice register pool 20`	Enters voice register Pool configuration mode for SIP phones. Use this command to control which phone registrations are accepted or rejected by a Cisco Unified SIP SRST device.
Step 9	**no vad** **Example:** `Router(config-register-pool)# no vad`	Disables voice activity detection (VAD) on the VoIP dial peer. VAD is enabled by default. Because there is no comfort noise during periods of silence, the call may disconnect. You may prefer to set no VAD on the SIP phone pool.
Step 10	**codec** *codec-type [bytes]* **Example:** `Router(config-register-pool)# codec g729r8`	Specifies the supported codec by a single SIP phone or a VoIP dial peer in a Cisco Unified SIP SRST environment. The codec-type argument specifies the preferred codec and can be one of the following: • **g711alaw** : G.711 a–law 64,000 bps. • **g711ulaw** : G.711 mu–law 64,000 bps. • **g729r8** : G.729 8000 bps (default). The bytes argument is optional and specifies the number of bytes in the voice payload of each frame.

Step 11	**end**
	Example:
	`Router(config-register-pool)# end`
		Returns to privileged EXEC mode.

Configuring SIP-to-SIP Call Forwarding

SIP-to-SIP call forwarding (call routing) is available. Call forwarding is provided either by the phone or by using a back-to-back user agent (B2BUA), which allows call forwarding on any dial peer. Calls into a SIP device may be forwarded to other SIP or SCCP devices (including Cisco Unity, third-party voicemail systems, or an auto attendant or IVR system such as Cisco Unified Contact Center and Cisco Unified Contact Center Express). In addition, SCCP IP phones may be forwarded to SIP phones.

Cisco Unity or other voice messaging systems connected by a SIP trunk or SIP user agent are able to pass a message-waiting indicator (MWI) when a message is left. The SIP phone then displays the MWI when indicated by the voice messaging system.

Note SIP-to-H.323 call forwarding is not supported.

To configure SIP-to-SIP call forwarding, you must first allow connections between specific types of endpoints in a Cisco IP-to-IP gateway. The **allow-connections** command grants this capability. Once the SIP-to-SIP connections are allowed, you can configure call forwarding under an individual SIP phone pool. Use any of the following commands to configure the call forwarding, according to your needs:

Under **voice register pool**

- Call-forward b2bua all directory-number
- Call-forward b2bua busy directory-number
- Call-forward b2bua mailbox directory-number
- Call-forward b2bua noan directory-number [timeout seconds]

A typical Cisco Unified SIP SRST setup does not use the call-forward b2bua mailbox command. However, Cisco Unified SIP Cisco Unified Communications Manager Express environment uses this command. You can find the detailed procedures for configuring the call-forward b2bua mailbox command in the Cisco Unified Communications Manager documentation on Cisco.com.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register pool** *tag* **voip**
4. **encall-forward b2bua alld** *directory-number*
5. **call-forward b2bua busy** *directory-number*
6. **call-forward b2bua mailbox** *directory-number*
7. **call-forward b2bua noan** *directory-number* **timeout** *seconds*
8. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:**	Enables privileged EXEC mode. • Enter your password if prompted.

	Command or Action	Purpose
	`Router> enable`	
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice register pool** *tag* **voip** **Example:** `Router(config)# voice register pool 15`	Enters voice register Pool configuration mode. Use this command to control the acceptance or rejections of the phone registrations on a Cisco Unified SIP SRST device.
Step 4	**encall-forward b2bua alld** *directory-number* **Example:** `Router(config-register-pool)# call-forward b2bua all 5005`	Enables call forwarding for a SIP back-to-back user agent (B2BUA) to forward all incoming calls to another non-SIP station extension. Namely to SIP trunk, H.323 trunk, SCCP device, and analog or digital trunk. *directory-number* : phone number to which calls are forwarded. Represents a fully qualified E.164 number. Maximum length of the phone number is 32.
Step 5	**call-forward b2bua busy** *directory-number* **Example:** `Router(config-register-pool)# call-forward b2bua busy 5006`	Enables call forwarding for a SIP B2BUA to forward incoming calls to a busy extension to another extension. *directory-number* : phone number to which calls are forwarded. Represents a fully qualified E.164 number. Maximum length of the phone number is 32.
Step 6	**call-forward b2bua mailbox** *directory-number* **Example:** `Router(config-register-pool)# call-forward b2bua mailbox 5007`	Controls the specific voicemail box in a voicemail system at the end of a call forwarding Exchange. *directory-number* : phone number to which calls are forwarded. Represents a fully qualified E.164 number. Maximum length of the phone number is 32.
Step 7	**call-forward b2bua noan** *directory-number* **timeout** *seconds* **Example:** `Router(config-register-pool)# call-forward b2bua noan 5010 timeout 10`	Enables call forwarding for a SIP B2BUA to forward incoming calls to an extension that does not answer after a configured amount of time to another extension. Use this command if a phone is registered with a Cisco Unified SIP SRST router, but the phone is not reachable because there is no IP connectivity (there is no response to Invite requests). • *directory-number* : phone number to which calls are forwarded. Represents a fully qualified E.164 number. Maximum length of the phone number is 32. • **timeout** *seconds* : Duration, in seconds, that a call can ring with no answer before the call is forwarded to another extension. Range is 3–60000. The default value is 20.

	Command or Action	Purpose

Step 8	**end**
	Example:
	`Router(config-register-pool)# end`
		Returns to privileged EXEC mode.

Configuring Call Blocking Based on Time of Day, Day of Week, or Date

This section applies to both SCCP and SIP SRST. Call blocking prevents the unauthorized use of phones. It is implemented by matching a pattern of up to 32 digits during specified time of day, day of the week, or date. Cisco Unified SIP SRST provides SIP endpoints the same time-based call blocking mechanism as provided for SCCP phones. The call blocking feature supports all incoming calls, including incoming SIP and analog FXS calls.

Note The Cisco Unified SIP SRST does not support the Pin-based exemptions and the “Login” toll-bar override.

Use the same commands for SIP phone call blocking and for SCCP phones on your Cisco Unified SRST system. The Cisco Unified SRST session application accesses the current after-hours configuration under call-manager-fallback mode. It applies to calls originated by Cisco SIP phones and registered to the Cisco Unified SRST router. The commands used in call-manager-fallback mode that set block criteria (time or date or block pattern) are the following:

- **after-hours block pattern** *pattern-tag pattern* [**7-24**]
- **after-hours day** *day start-time stop-time*
- **after-hours date** *month date start-time stop-time*

When you call to digits that match the specified patterns for call blocking during a defined time period for call blocking, the call stops and the caller hears a fast busy.

In SRST (call-manager-fallback configuration mode), there is no phone- or pin-based exemption to after-hours call blocking. However, in Cisco Unified SIP SRST (voice register Pool mode), individual IP phones can be exempted from all call blocking using the **after-hours exempt** command.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **call-manager-fallback**
4. **after-hours block pattern** *tag pattern* [**7-24**]
5. **after-hours day** *day start-time stop-time*
6. **after-hours date** *month date start-time stop-time*
7. **exit**
8. **voice register pool** *tag*
9. **after-hour exempt**
10. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**call-manager-fallback** **Example:** `Router(config)# call-manager-fallback`	Enters call-manager-fallback configuration mode.
Step 4	**after-hours block pattern** *tag pattern* [**7-24**] **Example:** `Router(config-cm-fallback)# after-hours block pattern 1 91900`	For blocking, defines a pattern of outgoing digits. Define up to 32 patterns, using individual commands. • If you specify the 7–24 keyword, always blocks the pattern, 7 days a week, 24 hours a day. • If you do not specify the 7–24 keyword, blocks the pattern during the days and dates as defined in the **after-hours day** and **after-hours date** commands.
Step 5	**after-hours day** *day start-time stop-time* **Example:** `Router(config-cm-fallback)# after-hours day mon 19:00 7:00`	Defines a recurring time period for the day of the week during which calls are blocked to outgoing dial patterns that are defined using the **after-hours block pattern** command. • *day* : Day of the week abbreviation. The following are valid day abbreviations: sun, mon, tue, wed, thu, fri, sat. • *start-time stop-time* : Beginning and ending times for call blocking, in an HH:MM format using a 24-hour clock. If the stop time is smaller value than the start time, the stop time occurs on the day following the start time. For example, "mon 19:00 07:00" means "from Monday at 7 p.m. until Tuesday at 7 a.m.". Value 24:00 is not valid. If you enter 00:00 as stop time, it changes to 23:59. If you enter 00:00 for both start time and stop time, blocks call for the entire 24-hour period on the specified date.
Step 6	**after-hours date** *month date start-time stop-time* **Example:** `Router(config-cm-fallback)# after-hours date jan 1 0:00 0:00`	Defines a recurring time period for the month and date for blocking calls to outgoing dial patterns defined in the after-hours block pattern command.

	Command or Action	Purpose
		• *month* : Month abbreviation. The following are valid month abbreviations: jan, feb, mar, apr, may, jun, jul, aug, sep, oct, nov, dec. • *date* : Date of the month. Range is 1–31. • *start-time stop time* : Beginning and ending times for call blocking, in an HH:MM format using a 24-hour clock. The stop time must be larger than the start time. Value 24:00 is not valid. If you enter 00:00 as stop time, it changes to 23:59. If you enter 00:00 for both start time and stop time, blocks call for the entire 24-hour period on the specified date.
Step 7	**exit** **Example:** `Router(config-cm-fallback)# exit`	Exits call-manager-fallback configuration mode.
Step 8	**voice register pool** *tag* **Example:** `Router(config)# voice register pool 12`	Enters voice register Pool configuration mode. Use this command to control the accepted or rejected registrations by a Cisco Unified SIP SRST device.
Step 9	**after-hour exempt** **Example:** `Router(config-register-pool)# after-hour exempt`	Specifies that for a particular voice register Pool, does not block the outgoing calls although call blocking is enabled.

Step 10	**end**
	Example:
	`Router(config-register-pool)# end`
		Returns to privileged EXEC mode.

Example

The following example defines several patterns of digits for which blocks outgoing calls. Patterns 1 and 2, blocks call to external numbers that begin with “1” and “011”:

- On Monday through Friday before 7 a.m. and after 7 p.m.
- On Saturday before 7 a.m. and after 1 p.m.
- All day Sunday.

Pattern 3 blocks call to 900 numbers 7 days a week, 24 hours a day.

```
call-manager-fallback
after-hours block pattern 1 91
after-hours block pattern 2 9011
after-hours block pattern 3 91900 7-24
after-hours day mon 19:00 07:00
after-hours day tue 19:00 07:00
```

```
after-hours day wed 19:00 07:00
after-hours day thu 19:00 07:00
after-hours day fri 19:00 07:00
```

The following example exempts a Cisco SIP phone pool from the configured blocking criteria:

```
voice register pool 1
after-hour exempt
```

Verification

To verify the feature's configuration, enter one of the following commands:

- **show voice register dial-peer** : Displays all the dial peers created dynamically by phones that have registered. This command also displays configurations for after hours blocking and call forwarding.
- **show voice register pool***tag* : Displays information about a specific Pool.
- **debug ccsip message** : Debugs basic B2BUA calls.

For more information about these commands, see Cisco Unified SRST and Cisco Unified SIP SRST Command Reference (All Versions).

SIP Call Hold and Resume

Cisco Unified SRST supports the ability for SIP phones to place calls on hold and to resume from calls placed on hold. It also includes support for consultative hold where A calls B, B place A on hold, B calls C, and B disconnects from C and then resumes with A. Support for a call hold is signaled by SIP phones using "re-INVITE c=0.0.0.0" and also by the receive-only mechanism.

No configuration is necessary.

```
Router# show running-config
Building configuration...
Current configuration : 1462 bytes
configuration mode exclusive manual
version 12.4
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
service internal
!
boot-start-marker
boot-end-marker
!
logging buffered 8000000 debugging
!
no aaa new-model
!
resource policy
!
clock timezone edt -5
clock summer-time edt recurring
ip subnet-zero
!
!
!
ip cef
!
!
```

```
!
voice-card 0
no dspfarm
!
!
voice service voip
allow-connections h323 to h323
allow-connections h323 to sip
allow-connections sip to h323
allow-connections sip to sip
sip
registrar server expires max 600 min 60
!
!
!
voice register global
max-dn 10
max-pool 10
!
! Define call forwarding under a voice register pool
voice register pool 1
id mac 0012.7F57.60AA
number 1 1000
call-forward b2bua busy 2413
call-forward b2bua noan 2414 timeout 30
codec g711ulaw
!
voice register pool 2
id mac 0012.7F3B.9025
number 1 2800
codec g711ulaw
!
voice register pool 3
id mac 0012.7F57.628F
number 1 2801
codec g711ulaw
!
!
!
interface GigabitEthernet0/0
ip address 10.0.2.99 255.255.255.0
duplex auto
speed auto
!
interface GigabitEthernet0/1
no ip address
shutdown
duplex auto
speed auto
!
ip classless
ip route 0.0.0.0 0.0.0.0 GigabitEthernet0/0
!
ip http server
!
!
!
control-plane
!
!
!
dial-peer voice 1000 voip
destination-pattern 24..
session protocol sipv2
session target ipv4:10.0.2.5
codec g711ulaw
!
! Define call blocking under call-manager-fallback mode
call-manager-fallback
max-conferences 4 gain -6
after-hours block pattern 1 2417
                  after-hours date Dec 25 12:01 20:00
                  !
                  !
                  line con 0
                  exec-timeout 0 0
                  line aux 0
                  line vty 0 4
                  login
                  !
                  scheduler allocate 20000 1000
                  ntp server 10.0.2.10
                  !
                  end
```

Enable Translation Profiles for SIP SRST

Cisco Unified SRST 3.2 and later and Cisco Unified SRST 4.0 and later support translation profiles. Translation profiles are the suggested way to allow you to group translation rules and provide instructions on how to apply the translation rules to the following:

- Called numbers
- Calling numbers
- Redirected called numbers

Note To enable translation-rule, see Enabling Digit Translation Rules section.

In the configuration below, the **voice translation-rule** and the **rule** command allow you to set and define how a number is to be manipulated. The translate command in voice translation-profile mode defines the type of number you are going to manipulate, such as a called, calling, or a redirecting number. Once you have defined your translation profiles, you can then apply the translation profiles in various places, such as dial peers and voice ports. For SRST, you apply your profiles in Cisco Unified Communications Manager fallback mode.

Cisco IP phones support one incoming and one outgoing translation profile when in SRST mode.

Note For Cisco Unified SRST 3.2 and later versions and Cisco Unified SRST 4.0 and later versions, use the **voice translation-rule** and **translation-profile** commands shown below instead of the translation rule configuration described in the Enabling Digit Translation Rules section. Voice translation rules are a separate feature from translation rules. See the voice translation-rule command in Cisco IOS Voice Command Reference for more information and the VoIP Gateway Trunk and Carrier Based Routing Enhancements documentation for more general information on translation rules and profiles.

SUMMARY STEPS

1. **voice translation-rule** *number*
2. **rule** *precedence/match-pattern/ /replace-pattern/*
3. **exit**
4. **voice translation-profile** *name*
5. **translate {called | calling | redirect-called}** *translation-rule-number*
6. **exit**
7. **voice register pool** *pool-tag*
8. **translation-profile {incoming | outgoing}** *name*
9. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**voice translation-rule** *number* **Example:** `Router(config)# voice translation-rule 1`	Defines a translation rule for voice calls and enters voice translation-rule configuration mode. *number*: Number that identifies the translation rule. Range is 1–2147483647.
Step 2	**rule** *precedence/match-pattern/ /replace-pattern/* **Example:** `Router(cfg-translation-rule)# rule 1/^9/ //`	Defines a translation rule. • *precedence*: Priority of the translation rule. Range is 1–15. • *match-pattern*: Stream editor (SED) expression used to match incoming call information. The slash (/) is a delimiter in the pattern. • *replace-pattern*: SED expression used to replace the match pattern in the call information. The slash (/) is a delimiter in the pattern.
Step 3	**exit** **Example:** `Router(cfg-translation-rule)# exit`	Exits voice translation-rule configuration mode.
Step 4	**voice translation-profile** *name* **Example:** `Router(config)# voice translation-profile name1`	Defines a translation profile for voice calls. *name*: Name of the translation profile. Maximum length of the voice translation profile name is 31 alphanumeric characters.
Step 5	**translate {called | calling | redirect-called}** *translation-rule-number* **Example:** `Router(cfg-translation-profile)# translate called 1`	Associates a voice translation rule with a voice translation profile. • **called**: Associates the translation rule with called numbers. • **calling**: Associates the translation rule with calling numbers.

	Command or Action	Purpose
		• **redirect-called**: Associates the translation rule with redirected called numbers. • *translation-rule-number*: The reference number of the translation rule 1–2147483647.
Step 6	**exit** **Example:** `Router(cfg-translation-profile)# exit`	Exits translation-profile configuration mode.
Step 7	**voice register pool** *pool-tag* **Example:** `Router(config)# voice register pool 10`	Enters voice register pool configuration mode for SIP phones. • *pool-tag*: Indicates an unique number assigned to the pool. Range is 1 to 100.
Step 8	**translation-profile {incoming | outgoing}** *name* **Example:** `Router(config-register-pool)# translation-profile outgoing name1`	Assigns a translation profile for incoming or outgoing call legs on a Cisco IP phone. • **incoming**: Applies the translation profile to incoming calls. • **outgoing**: Applies the translation profile to outgoing calls. • *name*: The name of the translation profile.
Step 9	**exit** **Example:** `Router(config-register-pool)# exit`	Exits from voice register pool configuration mode.

Example

The following example shows the configuration where a translation profile called name1 is created with two voice translation rules. Rule1 consists of associated calling numbers, and rule2 consists of redirected called numbers. The Cisco Unified IP Phones in SIP SRST mode are configured with name1.

```
voice translation-profile name1
 translate calling 1
 translate called redirect-called 2

voice register pool 10
 translation-profile incoming name1
```

Note For verification of translation profile configuration, see Verifying Translation Profiles.

How to Configure Optional Features

This section describes the following optional more call features:

- Three-party G.711 ad hoc conferencing—Cisco Unified Survivable Remote Site Telephony (SRST) support for simultaneous three-party conferences.
- XML application program interface (API)—This interface supplies data from Cisco Unified SRST to management software.

The following sections describe how to configure these optional features:

- Enabling Three-Party G.711 Ad Hoc Conferencing
- Defining XML API Schema

Enabling Three-Party G.711 Ad Hoc Conferencing

The enabling three-party G.711 ad hoc conferencing involves configuring the maximum number supported simultaneous three-party conferences by the Cisco Unified SRST router. For conferencing to be available, connect minimum of two lines to one or more buttons in an IP phone. See the Configuring a Secondary Dial Tone section.

SUMMARY STEPS

1. **call-manager-fallback**
2. **max-conferences** *max-conference-numbers*
3. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**call-manager-fallback** **Example:** `Router(config)# call-manager-fallback`	Enters call-manager-fallback configuration mode.
Step 2	**max-conferences** *max-conference-numbers* **Example:** `Router(config-cm-fallback)# max-conferences 16`	Sets the maximum number of supported simultaneous three-party conferences by the router. The maximum number possible is platform-dependent: • Cisco 1751 router: 8 • Cisco 1760 router: 8 • Cisco 2600 series routers: 8 • Cisco 2600-XM series routers: 8 • Cisco 2801 router: 8 • Cisco 2811, Cisco 2821, and Cisco 2851 routers: 16

	Command or Action	Purpose
		• Cisco 3640 and Cisco 3640A routers: 8 • Cisco 3660 router: 16 • Cisco 3725 router: 16 • Cisco 3745 router: 16 • Cisco 3800 Series router: 24
Step 3	**exit** **Example:** `Router(config-cm-fallback)# exit`	Exits call-manager-fallback configuration mode.

Example

The following example configures up to eight simultaneous three-way conferences on a router:

```
call-manager-fallback
max-conferences 8
```

Defining XML API Schema

The Cisco IOS commands in this section allow you to specify parameters associated with the XML API. For more information, see XML Provisioning Guide for Cisco CME/SRST. See the Enabling Consultative Call Transfer and Forward Using H.450.2 and H.450.3 with Cisco Unified SRST 3.0 section for configuration instructions.

SUMMARY STEPS

1. **call-manager-fallback**
2. **xmlschema** *schema-url*
3. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**call-manager-fallback** **Example:** `Router(config)# call-manager-fallback`	Enters call-manager-fallback configuration mode.
Step 2	**xmlschema** *schema-url* **Example:** `Router(config-cm-fallback)# xmlschema http://server2.example.com/ schema/schema1.xsd`	Specifies the URL for an XML API schema to be used with this Cisco Unified SRST system. *schema-url* : Local or remote URL as defined in RFC 2396.

	Command or Action	Purpose
Step 3	**exit** **Example:** `Router(config-cm-fallback)# exit`	Exits call-manager-fallback configuration mode.

Configuration Examples for Call Handling

Example: Monitoring the Status of Key Expansion Modules

Use the Show commands to monitor the status and other details of Key Expansion Modules (KEMs).

The following example demonstrates how the **show voice register all** command displays KEM details with all the Cisco Unified Communications Manager Express configurations and registration information:

```
show voice register all
VOICE REGISTER GLOBAL
=====================
CONFIG [Version=9.1]
========================
............
Pool Tag 5
Config:
Mac address is B4A4.E328.4698
Type is 9971 addon 1 CKEM
Number list 1 : DN 2
Number list 2 : DN 3
Proxy Ip address is 0.0.0.0
DTMF Relay is disabled
Call Waiting is enabled
DnD is disabled
Video is enabled
Camera is enabled
Busy trigger per button value is 0
keep-conference is enabled
registration expires timer max is 200 and min is 60
kpml signal is enabled
Lpcor Type is none
```

The following example demonstrates how the **show voice register pool type** command displays all the configured phones with add-on KEMs in Cisco Unified Communications Manager Express:

```
Router# show voice register pool type CKEM
Pool ID              IP Address      Ln DN Number                State
==== =============== =============== == === ==================== ============
4    B4A4.E328.4698  9.45.31.111     1  4   5589$                REGISTERED
```

Example: Configuring Voice Hunt Groups in Cisco Unified SIP SRST

The following example shows how to configure longest-idle hunt group 20 with pilot number 4701, final number 5000, and 6 numbers in the list. After directing a call six times (makes 6 hops), it is redirected to the final number 5000.

```
Router(config)# voice hunt-group 20 longest-idle
Router(config-voice-hunt-group)# pilot 4701
Router(config-voice-hunt-group)# list 4001, 4002, 4023, 4028, 4045, 4062
Router(config-voice-hunt-group)# final 5000
Router(config-voice-hunt-group)# hops 6
Router(config-voice-hunt-group)# timeout 20
Router(config-voice-hunt-group)# exit
```

Where to Go Next

If you must configure security, see the section, or if you must configure voicemail, see the Integrating Voice Mail with Cisco Unified SRST section. If you must configure video parameters, see the Setting Video Parameters section. If you do not need any of those features, go to the Monitoring and Maintaining Cisco Unified SRST section.

For additional information, see the Related Documents and References, on page 58 section in the Cisco Unified SRST Feature Overview chapter.

CHAPTER 10

Configure Secure SRST for SCCP and SIP

The Secure SRST adds security functionality to the Unified SRST.

Note Unified Secure SRST 12.6 on Cisco IOS XE Gibraltar 16.11.1a Release is not a recommended release version for Unified Secure SCCP SRST call flows and call flows that include stcapp configuration.

Prerequisites for Configuring Secure SRST

General

- Secure Cisco Unified IP phones supported in secure SCCP and SIP SRST must have the Certification Authority (CA) or third-party certificates installed, and encryption enabled. For more information on CA server authentication, see Autoenrolling and Authenticating the Secure Cisco Unified SRST Router to the CA Server.
- The SRST router must have a certificate; a certificate can be generated by a third party or by the Cisco IOS certificate authority (CA). The Cisco IOS CA can run on the same gateway as Cisco Unified SRST. Over the TLS channel (port 2445), automated certificate exchange happens between the Unified SRST router and the Cisco Unified Communications Manager. However, the phone certificate exchange to Unified SRST through Unified Communications Manager has to be downloaded manually on the Unified SRST router.
- Certificate trust lists (CTLs) on Cisco Unified Communications Manager must be enabled.
- It is mandatory to configure the command **supplementary-service media-renegotiate** under **voice service voip** configuration mode to enable the supplementary features supported on Unified Secure SRST.

Public Key Infrastructure on Secure SRST

- Set the clock, either manually or by using Network Time Protocol (NTP). Setting the clock ensures synchronicity with Cisco Unified Communications Manager.

- Enable the IP HTTP server (Cisco IOS processor) with the **ip http server** command, if not already enabled. For more information on public key infrastructure (PKI) deployment, see the Cisco IOS Certificate Server feature.
- If the certificate server is part of your startup configuration, you may see the following messages during the boot procedure:

```
% Failed to find Certificate Server's trustpoint at startup % Failed to find Certificate
Server's cert.
```

These messages are informational messages and indicate a temporary inability to configure the certificate server because the startup configuration has not been fully parsed yet. The messages are useful for debugging, in case the startup configuration is corrupted.

You can verify the status of the certificate server after the boot procedure using the **show crypto pki server** command.

Supported Cisco Unified IP Phones, Platforms, and Memory Requirements

- For a list of supported Cisco Unified IP Phones, routers, network modules, and codecs for secure SRST, see the Cisco Unified Survivable Remote Site Telephony Compatibility Information feature.
- For the most up-to-date information about the maximum number of Cisco Unified IP Phones, the maximum number of directory numbers (DNs) or virtual voice ports, and memory requirements, see the Cisco Unified SRST 12.3 Supported Firmware, Platforms, Memory, and Voice Products feature.

Restrictions for Configuring Secure SRST

General

- Cryptographic software features (“k9”) are under export controls. This product contains cryptographic features and is subject to United States and local country laws governing import, export, transfer, and use. Delivery of Cisco cryptographic products does not imply third-party authority to import, export, distribute or use encryption. Importers, exporters, distributors and, users are responsible for compliance with U.S. and local country laws. By using this product you agree to comply with applicable laws and regulations. If you are unable to comply with U.S. and local laws, return this product immediately

  A summary of U.S. laws governing Cisco cryptographic products may be found at the following URL: http://www.cisco.com/wwl/export/crypto/tool/

  If you require further assistance, please contact us by sending email to export@cisco.com.
- When a Secure Real-Time Transport Protocol (SRTP) encrypted call is made between Cisco Unified IP Phone endpoints or from a Cisco Unified IP Phone to a gateway endpoint, a lock icon is displayed on the IP phones. The lock indicates security only for the IP leg of the call. Security of the PSTN leg is not implied.

SCCP SRST

- Secure SCCP SRST is supported only within the scope of a single router.
- Cisco 4000 Series Integrated Services Routers support Secure SCCP SRST only on Unified SRST 12.3 and later releases. For Secure SCCP support on Unified SRST 12.3 Release:
  - Secure Cisco Jabber is not supported.

    - SRTP passthrough is not supported.
    - SDP Passthrough is not supported.
    - Video Calling is not supported.
    - Transcoding is not supported.
    - Hardware Conferencing is not supported (Only Software Conferencing is supported).
    - Secure Multicast MOH is not supported (Multicast MOH stays active, but non-secure).
    - Live MOH is not supported.
    - Secure H.323 is not supported.
    - Hot Standby Routing Protocol (HSRP) is not supported.
    - T.38 Fax Relay and Modem Relay is not supported for Unified Secure SRST.

- For call support on Voice Gateway introduced as part of Unified SRST 12.3 Release:
    - Speed Dial is not supported.
    - For a pure SCCP shared line, Hold and Remote Resume is not supported from an analog phone.
    - Full Blind Transfer mode (Configured with the CLI command transfer-system full-blind) is not supported.
    - Consider a call between two Analog Voice Gateways (VG A and VG B) registered on Unified Secure SRST as SCCP endpoints. If a call is already put on hold from the VG B endpoint (could be an SCCP phone too), then VG A (has to be an Analog Voice Gateway) cannot put the same call on hold (double hold). For more information, see CSCvi15203.
    - For three-way software conference related behavior and limitations, see Three-way Software Conferencing for Secure SCCP, Unified SRST Release 12.3.

SIP SRST

- Cisco 4000 Series Integrated Services Router supports Secure SIP SRST only on Unified SRST 12.1 and later releases.
- SRTP passthrough is not supported.
- SDP Passthrough is not supported.
- Video Calling is not supported.
- Transcoding is not supported.
- Hardware Conferencing is not supported (Only BIB Conferencing is supported).
- It is mandatory to configure security-policy secure under voice register global configuration mode. Non-Secure endpoints cannot register when security-policy secure is configured. As such, mixed deployments of secure and non-secure endpoints is not possible.

Information About Configuring Secure SRST

Benefits of Secure SRST

Secure Cisco Unified IP phones that are located at remote sites and that are attached to gateway routers can communicate securely with Cisco Unified Communications Manager using the WAN. But if the WAN link or Cisco Unified Communications Manager goes down, all communication through the remote phones becomes non-secure. To overcome this situation, gateway routers can now function in secure SRST mode, which activates when the WAN link or Cisco Unified Communications Manager goes down. When the WAN link or Cisco Unified Communications Manager is restored, Cisco Unified Communications Manager resumes secure call-handling capabilities.

Secure SRST provides new Cisco Unified SRST security features such as authentication, integrity, and media encryption. Authentication provides assurance to one party that another party is whom it claims to be. Integrity provides assurance that the given data has not been altered between the entities. Encryption implies confidentiality; that is, that no one can read the data except the intended recipient. These security features allow privacy for Cisco Unified SRST voice calls and protect against voice security violations and identity theft.

SRST security is achieved when:

- End devices are authenticated using certificates.
- Signaling is authenticated and encrypted using Transport Layer Security (TLS) for TCP.
- A secure media path is encrypted using Secure Real-Time Transport Protocol (SRTP).
- Certificates are generated and distributed by a CA.

Secure SIP SRST Support

As a part of the Secure SIP SRST feature on Unified SRST Release 12.1, support is provided for calls with Transport Layer Security protocols (TLS) versions up to 1.2. Also, supports TLS 1.2 exclusivity as part of Unified SRST Release 12.1.

Starting from Cisco Unified SRST 14.4 Release (Cisco IOS XE 17.14.1a), SRST security feature supports TLS version 1.3 and associated ciphers.

Note It's recommended to use TLS version 1.2 or 1.3 wherever possible to ensure security or compliance.

SIP SRST supports the following three functionalities:

- The TLS exclusivity functionality enables only the configured version of TLS (1.0 or 1.1 or 1.2 or 1.3).
- The default form supports all the TLS versions 1.3, 1.2, and 1.1. However, to configure TLS version 1.0, you must explicitly specify the TLS version.
- In sip-ua configuration mode, SIP SRST supports minimum TLS version functionality. You can configure the minimum TLS version only with TLS version 1.2, which supports both TLS versions 1.2 and 1.3 cipher negotiations with the peers.

Note

For the functionality configurations, see Signaling Security on Unified SRST - TLS.

For the list of supported TLS cipher suites, see TLS Cipher Suites.

Note

The Cisco IP Phone 7800 Series and Cisco IP Phone 8800 Series is supported on the Unified Secure SIP SRST Release 12.1 configured on a Cisco 4000 Integrated Services Router.

For Secure SIP SRST to be supported on Cisco 4000 Series Integrated Services Router, you need to enable the following technology package licenses on the router:

- security
- uck9

For Unified SRST 12.2 and previous releases, only SIP phones are supported on the Cisco 4000 Integrated Services Router for Secure SIP SRST. Unified SRST 12.3 and later releases supports a mixed deployment of SIP and SCCP phones on the Cisco 4000 Integrated Services Router.

Secure Music On Hold for Unified Secure SRST (SIP)

From Unified SRST Release 12.1, support is introduced for Secure Music On Hold (MOH), as part of the Secure SIP SRST solution on Cisco 4000 Series Integrated Services Router. For a Secure SIP call that is put on hold, playback of Flash-based G.729 and G.711 codec format MOH files are supported. Live MOH and transcoded MOH are not supported as part of Secure MOH feature support.

Note

If the CLI command **srtp pass-thru** is configured under the dial peer voice configuration mode, Secure MOH does not work.

Secure SCCP SRST Support

As a part of the Secure SCCP SRST feature on Unified SRST Release 12.3, support is provided for calls with the Transport Layer Security protocols (TLS) versions up to 1.2. Also, TLS 1.2 exclusivity is supported as part of Unified SRST Release 12.3. For more information on the TLS protocol support introduced for Secure SCCP in Unified SRST Release 12.3, see SRST Routers and the TLS Protocol.

Starting from Cisco Unified SRST 14.4 Release (Cisco IOS XE 17.14.1a), the secure SRST security feature is enhanced to support TLS version 1.3 and associated ciphers. SCCP SRST supports TLS exclusivity and default form functionalities:

Note

It's recommended to use TLS version 1.2 or 1.3 wherever possible to ensure security or compliance.

SCCP SRST supports the following three functionalities:

- The TLS exclusive functionality enables only the configured version of TLS (1.0 or 1.1 or 1.2 or 1.3).
- The default form supports all the TLS versions 1.3, 1.2, and 1.1. However, to configure TLS version 1.0, you must explicitly specify the TLS version.
- Secure SCCP SRST supports SHA2 ciphers with TLS versions 1.2 and 1.3.

Note For the functionality configurations, see Signaling Security on Unified SRST - TLS.

For the list of supported TLS cipher suites, see TLS Cipher Suites.

Secure SCCP SRST for Analog Voice Gateways

For Unified SRST 12.3 and later releases on a Cisco 4000 series Integrated Services Router and Catalyst 8000 series. Secure SCCP support is introduced for the following Voice Gateways:

- Cisco VG202 Analog Voice Gateway
- Cisco VG202XM Analog Voice Gateway
- Cisco VG204 Analog Voice Gateway
- Cisco VG204XM Analog Voice Gateway
- Cisco VG224 Analog Voice Gateway
- Cisco VG300 Series Gateways (VG310, VG320, VG350)

As a part of the Secure SCCP SRST feature on Unified SRST Release 12.3, Transport Layer Security protocols (TLS) supports versions up to 1.2, and TLS 1.2 exclusivity is supported for Cisco VG202XM Analog Voice Gateway, Cisco VG204XM Analog Voice Gateway, Cisco VG310 Analog Voice Gateway, and Cisco VG320 Analog Voice Gateway.

Note The above listed Cisco 2xx, and 3xx Analog Voice Gateways are End-of-Life or End-of-Support.

For Unified SRST Release 14.4 (Cisco IOS XE 17.14.1a), on a Cisco 4461 Integrated Services Router series, and Catalyst 8000 series SCCP TLS v1.3 support is introduced for the following Voice Gateways using STCAPP:

- Cisco VG400 Analog Voice Gateway
- Cisco VG410 Analog Voice Gateway
- Cisco VG420 Analog Voice Gateway
- Cisco VG450 Analog Voice Gateway

SCCP TLS v1.3 is supported on Catalyst 8200, 8300, and Cisco 4461 Integrated Services Router series.

For more information on configuring the Voice Gateways, see Supplementary Services Features for FXS Ports on Cisco IOS Voice Gateways Configuration Guide.

Note Cisco VG202 Analog Voice Gateway, Cisco VG204 Analog Voice Gateway, and Cisco VG224 Analog Voice Gateway only supports TLS version 1.0.

Note For Secure SCCP SRST to be supported on Cisco 4000 Integrated Services Router, you need to enable the following technology package licenses on the router:

- security
- uck9

The Cisco Unified IP Phone 6961 and Cisco Unified IP Phone 7962G are supported on the Unified Secure SCCP SRST Release 12.3 configured on Cisco 4000 Integrated Services Router. Also, supports analog phones for Analog Voice Gateways as part of Unified Secure SCCP SRST Release 12.3. For more information on support introduced on Voice Gateways, see Secure SCCP SRST for Analog Voice Gateways.

Secure Music On Hold for Secure Unified SRST (SCCP)

From Unified SRST Release 12.3, support is introduced for Secure Music On Hold (MOH), as part of the Secure SCCP SRST functionality on Cisco 4000 Series Integrated Services Router. For a Secure SCCP call that is put on hold, playback of Flash-based G.729 and G.711 codec format MOH files are supported. Live MOH and transcoded MOH are not supported as part of Secure MOH feature support. Also, Multicast MOH is supported as non-secure on fallback from Cisco Unified Communications Manager to Unified Secure SRST.

Three-way Software Conferencing for Secure SCCP, Unified SRST Release 12.3

From Unified SRST Release 12.3, three-way software conferencing is supported for Secure SCCP endpoints on Cisco 4000 Series Integrated Services Routers. The audio codec supported as part of the three-way software conferencing for Unified SRST 12.3 Release is G.711. The support is introduced for Secure SCCP phones and Secure SCCP endpoints registered on Cisco Analog Voice Gateways.

Three-way software conferencing is supported for a pure SCCP deployment (only involving SCCP endpoints), and a mixed deployment of secure SCCP and SIP phones. The SCCP phones such as Cisco Unified IP Phone 7962, Cisco Unified IP Phone 6961, and Cisco Unified IP Phone 7975 are supported as part of this deployment. For the mixed deployment, the Cisco IP Phone 7800 Series and Cisco IP Phone 8800 Series SIP phones are supported. Three-way Software Conference is supported on TDM trunks, for SIP and SCCP endpoints on Unified Secure SRST.

You can set a limit for the maximum number of conferences that are supported. Configure the CLI command **max-conferences** under **call-manager-fallback** configuration mode to set the maximum number of conferences supported. If you do not set the maximum number of supported conferences using the command **max-conferences**, the limit is set to the default value of 8.

```
Router(config-cm-fallback)#max-conferences ?
<1-16> Maximum conferences to support
```

For a three-way software conference supported on Secure Unified SRST:

- When a secure SCCP endpoint initiates the conference or the SCCP endpoint is a conference host, the conference is created. The three-way software conference is hosted on a Unified Secure SRST router.

- When a secure SIP endpoint initiates the conference, the three-way software conference is hosted on the SIP phone.
- When the conference host puts the call on hold, the other participants in the three-way software conference will hear Music On Hold until the call is resumed by the host. Multicast MOH is played for an SCCP endpoint, whereas Unicast MOH is played for SIP endpoints.
- When the three-way software conference host is an Analog Voice Gateway endpoint, the host cannot place the conference on hold. The three-way software conference can be put on hold only by SCCP or SIP endpoints.
- When any of the conference participants (apart from the host) put the call on hold, the other participants in the three-way software conference can continue to talk.
- For a three-way software conference on Unified SRST for Secure SCCP endpoints, the conference participants can transfer the call. The conference host cannot transfer the conference call. During an alert transfer, the other two participants can continue to talk without media interruption.
- Conference Cascading is not supported for a three-way software conference on Unified Secure SRST.
- Consider a three-way software conference hosted by an Analog Voice Gateway endpoint, with SCCP A and SCCP B as the second and third conference participants, respectively. In a scenario where SCCP B places the call on hold and the conference host tries to commit the conference using hookflash (followed by FAC), the call with SCCP B is terminated and conference attempt fails.
- Consider a scenario where an Analog Phone (AP 1) registered to the Analog Voice Gateway places a call to SCCP Phone (SCCP 1) registered to Secure SCCP SRST. After placing SCCP 1 on hold, AP 1 places a call to the third participant, SCCP Phone (SCCP 2), that is registered to the same Secure SRST. Three-way Software Conferencing is established. When SCCP 2 tries to perform an alert transfer to a phone (SIP 3/ SCCP 3) and it goes unanswered, the three-way conference is lost and it becomes a one-to-one call between AP 1 and SCCP 1. Any further attempt by AP 1 to establish a three-way software conference with another phone (SCCP 4) is not supported in this scenario.

Note If the failed alert transfer is by SCCP 1, then any further attempt to establish a three-way software conference with another phone will be supported.

Feature Support for Secure SRST (SCCP), Unified SRST Release 12.3

The Secure SCCP SRST on Cisco 4000 Series Integrated Services Routers and the Analog Voice Gateways introduced as part of Unified SRST Release 12.3, offers the following basic and supplementary call processing support. For a list of restrictions for Unified SRST 12.3 and later releases on Cisco Integrated Services Router Generation 2, see Restrictions for Configuring Secure SRST.

- Call Forward (Busy, No-answer, All)
- Call Hold or Resume
- Redial
- Secure MOH (Flash Based)
- Speed Dial (Only for Secure SCCP phones on Cisco 4000 Series Integrated Services Router)
- Secure Three-party Software Conference

- SIP trunks (Secure and Non-secure)
- TDM trunks
- Call Transfer (Alert, Consult, and Blind)
- Shared Line (Only for a pure SCCP-to-SCCP shared line. Mixed shared line is not supported.)
- Caller ID
- Call Waiting
- Media Inactivity

The following features are supported for Analog Voice Gateways for Fax and Modem calls on analog FXS ports:

- Fax Passthrough
- Modem Passthrough

Cisco IP Phones Clear-Text Fallback During Non-Secure SRST

- Cisco Unified SRST versions before 12.3(14)T are not capable of supporting secure connections or have security enabled. If an SRST router is not capable of SRST as a fallback mode—that is, it is not capable of completing a TLS handshake with Cisco Unified Communications Manager—its certificate is not added to the configuration file of the Cisco IP phone. The absence of a Cisco Unified SRST router certificate causes the Cisco Unified IP phone to use nonsecure (clear-text) communication when in Cisco Unified SRST fallback mode. The capability to detect and fallback in clear-text mode is built into Cisco Unified IP phone firmware. See Media and Signaling Authentication and Encryption Feature for Cisco IOS MGCP Gateways for more information on clear-text mode.

Signaling Security on Unified SRST - TLS

SRST Routers and the TLS Protocol

Transport Layer Security (TLS) provides secure TCP channels between Cisco Unified IP phones, secure Cisco Unified SRST Routers, and Cisco Unified Communications Manager. The TLS process begins with the Cisco Unified IP Phone establishing a TLS connection when registering with a Cisco Unified Communications Manager. Assuming that a Cisco Unified Communications Manager is configured to fall back to Cisco Unified SRST, the TLS connection between the Cisco Unified IP Phones and the secure Cisco Unified SRST Router is also established. If the WAN link or Cisco Unified Communications Manager fails, call control reverts to the Cisco Unified SRST router.

Unified Secure SIP and SCCP SRST Earlier Release Versions

From Unified Secure SIP SRST Release 12.1, support is introduced for SIP-to-SIP calls with Transport Layer Security up to TLS version 1.2. For configuring TLS 1.2 exclusivity functionality, you need to configure the command **transport tcp tls v1.2** under **sip-ua** configuration mode. When you configure TLS 1.2 exclusivity on the Secure SIP SRST, any registration attempt by phones using lower versions of TLS (1.0, 1.1) are rejected.

Before Unified SCCP SRST Release 12.3, support is available only for TLS 1.0 version. From Unified Secure SCCP SRST Release 12.3 and later releases, support is introduced for Transport Layer Security up to TLS version 1.2. To configure a specific TLS version or TLS 1.2 exclusivity for Unified Secure SCCP SRST, you need to configure **transport-tcp-tls** under **call-manager-fallback**. When **transport-tcp-tls** is configured without specifying a version, it enables the default behavior of the CLI command. In the default form, all the TLS versions (except TLS 1.0) are supported for this CLI command.

For TLS 1.0 support on Cisco IOS XE Fuji Release 16.9.1 for SCCP endpoints, you need to specifically configure:

- **transport-tcp-tls** *v1.0* in **call-manager-fallback** configuration mode.

For TLS 1.0 support on Cisco IOS XE Fuji Release 16.9.1 for SIP and mixed deployment scenarios, you need to specifically configure:

- **transport-tcp-tls** *v1.0* in **sip-ua** configuration mode.

From Cisco IOS XE Fuji Release 16.9.1, the security certificate exchange between Unified Secure SRST Release 12.3 and Unified Communications Manager doesn't support TLS version 1.0.

For Secure SIP and Secure SCCP endpoints that don't support TLS version 1.2, you need to configure TLS 1.0 for the endpoints to register to Unified Secure SRST 12.3 (Cisco IOS XE Fuji Release 16.9.1). This also means that endpoints which support 1.2 should also use the 1.0 suites.

Note Unified Communications Manager Release 11.5.1SU3 is the minimum version required to support security certificate exchange with Unified Secure SRST Release 12.3 (Cisco IOS XE Fuji Release 16.9.1).

Note SCCP phones and the Analog Voice Gateways VG202, VG204, and VG224 support only TLS version 1.0. For Unified Secure SRST 12.3 Release and later, supports TLS versions 1.1 and 1.2 for Cisco Analog Voice Gateways VG202XM, VG204XM, VG310, and VG320.

For Unified Secure SCCP SRST Release 12.3 and later releases, Analog Voice Gateways can register their SCCP endpoints with TLS versions up to 1.2 (TLS 1.0, 1.1, and 1.2).

The VG2xx and VG3xx Analog Voice Gateway series are End-of-Life or End-of-Support.

Unified Secure SIP and SCCP SRST Release 14.4

From Unified Secure SIP SRST Release 14.4 (Cisco IOS XE 17.14.1a), support for TLS version 1.3 and associated ciphers is introduced. The TLS exclusivity functionality enables only the configured version of TLS (1.0 or 1.1 or 1.2 or 1.3). To configure exclusivity functionality, use the **transport tcp tls** *version* command in **sip-ua** configuration mode.

In sip-ua configuration mode, SIP SRST supports **minimum** TLS version functionality. You can configure the minimum TLS version only with TLS version 1.2, which supports both TLS versions 1.2 and 1.3 cipher negotiations with the peers.

For secure SCCP SRST support for TLS version 1.3 is introduced in addition to SHA2 cipher support with TLS version 1.3.

To configure TLS version exclusivity for Unified Secure SCCP SRST, use the **transport-tcp-tls** command in **call-manager-fallback** configuration mode. When **transport-tcp-tls** is configured without specifying a version, the default behavior of the CLI command is enabled. In the default form, all the TLS versions 1.3, 1.2, and 1.1 are supported. However, to configure TLS version 1.0, you must explicitly specify the TLS version.

For more information on the **transport-tcp-tls** command, see Cisco Unified SRST Command Reference (All Versions).

TLS version 1.3 is supported for Cisco VG400, VG410, VG420, and VG450 Analog Voice Gateways. However, TLS version 1.3 is not supported on SCCP IP phone endpoints.

For the support of a specific TLS version on the Analog Voice Gateways for Unified SRST releases, you need to configure **stcapp security tls-version** command:

```
enable
 configure terminal
 stcapp security tls-version v1.3
exit

--
VG(config)#stcapp security tls-version ?
  v1.0 Enable TLS Version 1.0
  v1.1 Enable TLS Version 1.1
  v1.2 Enable TLS Version 1.2
  v1.3 Enable TLS Version 1.3
```

Configure SIP SRST in sip-ua Configuration Mode

For Unified Secure SIP SRST, you can configure **transport tcp tls** command in **sip-ua** configuration mode as follows:

```
Device(config)# voice service voip
Device(conf-voi-serv)#sip-ua
Device(config-sip-ua)# transport tcp tls ?
v1.0 Enable TLS Version 1.0
v1.1 Enable TLS Version 1.1
v1.2 Enable TLS Version 1.2
v1.3 Enable TLS Version 1.3
<cr>              <cr>
```

You can configure **transport tcp tls v1.2 minimum** command to enable tls versions 1.2 and 1.3:

```
Device(config)# voice service voip
Device(conf-voi-serv)#sip-ua
Device(config-sip-ua)# transport tcp tls
Device(config-sip-ua)# transport tcp tls v1.2 ?
minimum Enable TLS versions 1.2 and 1.3
<cr>              <cr>
```

Configure SCCP SRST in call-manager-fallback Configuration Mode

For Unified Secure SCCP SRST, you can configure **transport-tcp-tls** command in **call-manager-fallback** configuration mode as follows:

```
Router(config)#call-manager-fallback
Router(config-cm-fallback)#transport-tcp-tls ?
v1.0 Enable TLS Version 1.0
v1.1 Enable TLS Version 1.1
v1.2 Enable TLS Version 1.2
v1.3 Enable TLS Version 1.3
```

You can configure **transport-tcp-tls v1.3 sha2** command to enable SHA2 ciphers for media:

```
Router(config)#call-manager-fallback
Router(config-cm-fallback)#transport-tcp-tls v1.3 ?
sha2 Allow SHA2 ciphers only
```

TLS Cipher Support for Secure SRST

From Unified Secure SRST 12.6 onwards, the TLS cipher support offered on Secure SRST is modified to enhance security.

Starting from Unified Secure SRST 14.4, TLS version 1.3 is supported in addition to 1.2, 1.1, 1.0 and all the associated cipher suites.

TLS Cipher Support for SCCP/TLS (Ports 2443 and 2445)

The following cipher suites are supported (offer and accept):

Note ECDSA cipher is not supported with Secure SRST.

Ciphers	Descriptions
TLS 1.3 and TLS1.3 SHA2:	
AES128_GCM_SHA256	supported in TLS 1.3
AES256_GCM_SHA384	supported in TLS 1.3
CHACHA20_POLY1305_SHA256	supported in TLS 1.3
TLS 1.2 SHA2:	
ECDHE_ECDSA_AES128_GCM_SHA256	supported in TLS 1.2 & above
ECDHE_ECDSA_AES256_GCM_SHA384	supported in TLS 1.2 & above
DHE_RSA_WITH_AES_128_CBC_SHA	supported in TLS 1.0 & above
DHE_RSA_WITH_AES_256_CBC_SHA	supported in TLS 1.0 & above
ECDHE_RSA_AES256_GCM_SHA384	supported in TLS 1.2 & above
ECDHE_RSA_AES128_GCM_SHA256	supported in TLS 1.2 & above
Default or 1.2 or 1.1:	
RSA_WITH_AES_128_CBC_SHA	supported in TLS 1.0 & above

TLS Cipher Support for SIP/TLS (Port 5061)

The following cipher suites are supported (offer and accept):

Ciphers	Descriptions
TLS 1.3:	

Ciphers	Descriptions
AES128_GCM_SHA256	Supported in TLS 1.3
AES256_GCM_SHA384	Supported in TLS 1.3
CHACHA20_POLY1305_SHA256	Supported in TLS 1.3
TLS 1.2 SHA2:	
ECDHE_ECDSA_AES128_GCM_SHA256	Supported in TLS 1.2 and lower versions.
ECDHE_ECDSA_AES256_GCM_SHA384	Supported in TLS 1.2 and lower versions.
ECDHE_RSA_AES256_GCM_SHA384	Supported in TLS 1.2 and lower versions.
ECDHE_RSA_AES128_GCM_SHA256	Supported in TLS 1.2 and lower versions.
Default or 1.2 or 1.1:	
RSA_WITH_AES_128_CBC_SHA	Supported in TLS 1.0 and higher versions.

Certificates Operation on Secure SRST

Cisco Unified SRST Routers and PKI

The transfer of certificates between a Cisco Unified SRST router and Cisco Unified Communications Manager is mandatory for secure SRST functionality. Public key infrastructure (PKI) commands are used to generate, import, and export the certificates for secure Cisco Unified SRST. The following table shows the secure SRST-supported Cisco Unified IP Phones and the appropriate certificate for each phone. The Additional References section contains information and configurations about generating, importing, and exporting certificates that use PKI commands.

Note Certificate text can vary depending on your configuration. You may also need CAP-RTP-00X or CAP-SJC-00X for older phones that support manufacturing installed certificate (MIC).

Note Cisco supports Cisco IP Phones 7900 series phone memory reclamation phones that use MIC or locally significant certificate (LSC) certificates.

Table 1: Supported Cisco Unified IP Phones and Certificates

Cisco Unified IP Phone 7940	Cisco Unified IP Phone 7960	Cisco Unified IP Phone 7970
The phone receives locally significant certificate (LSC) from Certificate Authority Proxy Function (CAPF) in Distinguished Encoding Rules (DER) format. 59fe77ccd.0 The filename may change based on the CAPF certificate subject name and the CAPF certificate issuer. If Cisco Unified Communications Manager is using a third-party certificate provider, there can be multiple .0 files (from two to ten). Each .0 certificate file must be imported individually during the configuration. Manual enrollment supported only	The phone receives locally significant certificate (LSC) from Certificate Authority Proxy Function (CAPF) in Distinguished Encoding Rules (DER) format. 59fe77ccd.0 The filename may change based on the CAPF certificate subject name and the CAPF certificate issuer. If Cisco Unified Communications Manager is using a third-party certificate provider, there can be multiple .0 files (from two to ten). Each .0 certificate file must be imported individually during the configuration. Manual enrollment supported only.	The phone contains a manufacturing installed certificate (MIC) used for device authentication. If the Cisco 7970 implements MIC, two public certificate files are needed: CiscoCA.pem (Cisco Root CA, used to authenticate the certificate.) **Note** The name of the manufacturing certificate can vary depending on your configuration. a69d2e04.0, in Privacy Enhanced Mail (PEM) format If Cisco Unified Communications Manager is using a third-party certificate provider, there can be multiple .0 files (from two to ten). Each .0 Certificate file must be imported individually during the configuration. Manual enrollment supported only.

Cisco IOS Credentials Server on Secure SRST Routers

Secure SRST introduces a credentials server that runs on a secure SRST router. When the client, Cisco Unified Communications Manager, requests a certificate through the TLS channel, the credentials server provides the SRST router certificate to Cisco Unified Communications Manager. Cisco Unified Communications Manager inserts the SRST router certificate in the Cisco Unified IP Phone configuration file and downloads the configuration files to the phones. The secure Cisco Unified IP Phone uses the certificate to authenticate the SRST router during fallback operations. The credentials service runs on default TCP port 2445.

Three Cisco IOS commands configure the credentials server in call-manager-fallback mode:

- **credentials**
- **ip source-address (credentials)**
- **trustpoint (credentials)**

Two Cisco IOS commands provide credential server debugging and verification capabilities:

- debug credentials
- show credentials

Generating a Certificate for the Credentials Server

In configuring the credentials server on the Unified Secure SRST, a certificate is required to complete the "trustpoint " configuration entry.

To generate the certificate for Credentials Server, perform the following procedures:

- Autoenrolling and Authenticating the Secure Cisco Unified SRST Router to the CA Server
- Enabling Credentials Service on the Secure Cisco Unified SRST Router
- Configuring SRST Fallback on Cisco Unified Communications Manager

Once the certificate is generated, fill in the name of the certificate (or the name of the trustpoint in IOS) in the "trustpoint" entry.

This certificate for the Credentials Server on the Secure SRST will be seamlessly exported to the Cisco Unified CM when requested in Adding an SRST Reference to Cisco Unified Communications Manager section.

Certificates Transport from CUCM to Secure SRST

For more information about Certificates Transport from CUCM to Secure SRST, see Importing Phone Certificate Files in PEM Format to the Secure SRST Router section.

SIP OAuth Client Registration for Unified Secure SRST

Unified Secure SIP SRST enables routers to provide secure call-handling for Unified IP phones during an outage. The support is for endpoints that lose connection to the remote primary, secondary, or tertiary Unified CM installations during a WAN outage. If SIP OAuth is configured, SIP clients can securely register to the SRST during WAN link failures. The SRST can provide secure call control for the following SIP clients:

- Cisco Jabber Client
- Cisco Webex Client
- Cisco IP Phone 78xx Series
- Cisco IP Phone 88xx Series

Dynamic, token-based authentication provides improved security for devices registering to Unified CM.

When registering to the SRST during an outage, UCM issues an authentication token that the client uses. A challenge is issued when a new registration request doesn't include a token. The SRST attempts to validate the token using keys previously received securely from UCM. If the validation is successful, the SRST allows the client to register and place calls locally. Clients presenting a token that can't be validated by the SRST aren't allowed to register.

Note Supports TLS version 1.3 to fetch SIP OAuth keys from the CUCM. The **http client secure-ciphersuite** command configuration supports TLS version 1.3 ciphers. For configuration details, see **http client secure-ciphersuite**.

Note Stores the key pairs in persistent memory, ensuring that clients can register if the SRST router reloads during a service outage.

To configure SIP OAuth for the Unified Secure SIP SRST, perform the following:

1. Configure a TLS listen port without client validation for use by SIP OAuth clients.

   Note The TLS listen port is open in addition to the default secure port that uses mTLS.

   ```
   voice service voip
     sip
       listen-port secure no-client-validation ?
         <1024-49151>   Port number
   ```

2. Perform **call service stop** before configuring the listen port and **no call service stop** after configuring the listen port.

3. Configure access to the UCM key server with appropriate authentication details. Stores clear text passwords using type 6 encryption.

   ```
   voice register global
    sip-oauth                SIP OAuth parameters for Unified SRST
        key-server key-server ipv4:10.5.10.50:8443 username administrator password 0
   abcd12345
   ```

4. Configure device pools for compatible clients to use SIP OAuth. Enables **SIP OAuth** for compatible clients using the voice register pool configuration.

   ```
   voice register pool <tag>
           sip-oauth
   ```

Feature Characteristics

- SRST is configured to use a TLS socket without mTLS validation for clients that use SIP OAuth.
- Registration using SIP OAuth is enabled for clients through their voice register pool configuration.
- Cisco Unified SRST accepts new registration from clients with a valid SIP OAuth token.
- Protocol mode should be either "IPV4 only " or "IPV6 only" for SIP OAuth.

Restrictions

ECDSA cipher suite is not supported on port 2445.

Configure SIP OAuth-based Listener Port

SUMMARY STEPS

1. **configure terminal**
2. **voice service voip**
3. **sip**

4. **call service stop**
5. **listen-port secure no-client-validation** *<1024-49151>*
6. **no call service stop**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**configure terminal** **Example:** `Router#conf t`	Enters global configuration mode.
Step 2	**voice service voip** **Example:** `Router(config)#voice service voip`	Enters voice service VoIP mode.
Step 3	**sip** **Example:** `Router(conf-voi-serv)#sip`	Enters voice service VoIP sip mode.
Step 4	**call service stop**	Shuts down VoIP call service on a gateway.
Step 5	**listen-port secure no-client-validation** *<1024-49151>* **Example:** `Router(conf-serv-sip)#listen-port secure no-client-validation 5090`	Configures a TLS listen port with mTLS disabled. **Note** Default port is 5090.
Step 6	**no call service stop**	Enables VoIP call service.

Retrieve SIP OAuth Keys from CUCM

Voice Register Global Configuration Mode

Note Execute **voice sip oauth get-keys** to retrieve sip-oauth keys anytime from CUCM.

SUMMARY STEPS

1. **voice register global**
2. **sip-oauth**
3. **key-server** *word* **username** *word* **password 0/6** *word*
4. **key-server source-interface <options>**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**voice register global** **Example:** `Router(config)#voice register global`	Enters voice register global configuration mode.

Step 2	**sip-oauth**
	Example:
	`Router(config-register-global)#sip-oauth`
	`Router(config-oauth)#`
		Enables SIP OAuth feature.

Step 3	**key-server** *word* **username** *word* **password 0/6** *word*
	Example:
	`voice register global`
	`   sip-oauth`
	`       key-server ipv4:10.5.10.50:8443 username administrator password 0 C1sco123=`
		Configures key-server details for SIP OAuth. The key server provides the keys in JSON format to authenticate the token sent by phones. The key-server address is usually the CUCM IP address. The <*word*> must be in one of the following formats:
		ipv4:X.X.X.X
		ipv4:X.X.X.X:port-number
		ipv6:[X:X:X:X:X:X]
		ipv6:[X:X:X:X:X:X]:port-number
		dns:hostname.com
		dns:hostname.com:port-number
		Note If the port is not configured, then 443 secure port is used for HTTPS communication.
		Note If dns hostname is configured with a port, then SRV query is performed.

Step 4	**key-server source-interface <options>**	(Optional) Configures interface specification of source address for OAuth server.

Global Configuration Mode

SUMMARY STEPS

1. **http client secure-ciphersuite**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**http client secure-ciphersuite** **Example:** `Device(config)# http client secure-ciphersuite tls13-aes128-gcm-sha256`	(Optional) Configures one or more encryption cipher suite for the HTTP client. If **http client secure-ciphersuite** command is not configured, then by default, all the ciphers are negotiated.

Enable SIP OAuth-based Registration

SUMMARY STEPS

1. **voice register pool** *tag*
2. **sip-oauth**
3. **end**

DETAILED STEPS

	Command or Action	Purpose

Step 1	**voice register pool** *tag*
	Example:
	`Router(config)#voice register pool 20`
	`Router(config-register-pool)#`
		Enters voice register pool configuration mode.

Step 2	**sip-oauth** **Example:** `Router(config-register-pool)#sip-oauth`	Enables SIP OAuth on Pool.

Step 3	**end**
	Example:
	`Router(config-register-pool)#end`
		Returns to privileged EXEC mode.

Verify SIP OAuth for Secure SRST

SUMMARY STEPS

1. **show running-config all | sec listen-port**
2. **show sip-ua connections tcp tls detail**
3. **show sip status registrar**
4. **show voice register pool <index>**
5. **show voice register statistics**
6. **show voip sip-oauth key-server status**

DETAILED STEPS

Step 1 **show running-config all | sec listen-port**

Show command output that displays information on the **listen-port** configuration in SIP OAuth.

Example:

```
Router#show running-config | section listen-port
listen-port secure no-client-validation 5090
Router#
```

Step 2 **show sip-ua connections tcp tls detail**

Show command to display the status, port details, and negotiated ciphers for SIP OAuth.

Note The Conn-Id suffixed with * are the client connections using SIP OAuth port.

Note The RSA or ECDSA key types in the detailed output are displayed only with TLS v1.3.

Example:

The following is a sample output for the **show sip-ua connections tcp tls detail** command displaying RSA key type along with TLS v1.3 ciphers:

```
Device# show sip-ua connections tcp tls detail
Total active connections      : 2
No. of send failures          : 0
No. of remote closures        : 0
No. of conn. failures         : 0
No. of inactive conn. ageouts : 0
Max. tls send msg queue size of 1, recorded for 10.64.100.152:5061
TLS client handshake failures : 0
TLS server handshake failures : 0

---------Printing Detailed Connection Report---------
Note:
 ** Tuples with no matching socket entry
    - Do 'clear sip <tcp[tls]/udp> conn t ipv4:<addr>:<port>'
      to overcome this error condition
 ++ Tuples with mismatched address/port entry
    - Do 'clear sip <tcp[tls]/udp> conn t ipv4:<addr>:<port> id <connid>'
      to overcome this error condition
 * Connections with SIP OAuth ports

Remote-Agent:10.64.100.150, Connections-Count:1
  Remote-Port Conn-Id Conn-State  WriteQ-Size            Local-Address               TLS-Version
             Cipher              Curve Tenant
  =========== ======= =========== =========== ====================================== ===========
================================== ===== ======
        22943       7 Established           0 10.64.100.151:5061                         TLSv1.3
      TLS_AES_256_GCM_SHA384:RSA P-521     0

Remote-Agent:10.64.100.152, Connections-Count:1
  Remote-Port Conn-Id Conn-State  WriteQ-Size            Local-Address               TLS-Version
             Cipher              Curve Tenant
  =========== ======= =========== =========== ====================================== ===========
================================== ===== ======
         5061       8 Established           0 10.64.100.151:47687                        TLSv1.3
      TLS_AES_256_GCM_SHA384:RSA P-521     0


-------------- SIP Transport Layer Listen Sockets ---------------
  Conn-Id             Local-Address                      Tenant
 ==========    ===========================              ========
  0             [0.0.0.0]:5061:                              0
  6             [10.64.100.151]:5061:                        0
```

Step 3 show sip status registrar

Show command to display the registration status of a SIP client.

Note Transport parameter (TLS) suffixed with * are the endpoints registered using SIP OAuth port.

Example:

```
Router#show sip status registrar
Line          destination                               expires(sec)  contact
```

```
transport     call-id
              peer
====================================================================================================
2999904       10.5.10.204                             76            10.5.10.204

TLS*           00451d86-f1520107-5b4fd894-7ab6c4ce@10.5.10.204
              40004

2999901       10.5.10.212                             74            10.5.10.212

TLS            00af1f9c-12dc037b-14a5f99d-09f10ac4@10.5.10.212
              40001

2999902       10.5.10.213                             75            10.5.10.213

TLS*           00af1f9c-48370020-2bf6ccd4-2423aff8@10.5.10.213
              40002

2999905       10.5.10.209                             76            10.5.10.209

TLS*           5006ab80-69ca0049-1ce700d8-12edb829@10.5.10.209
              40003
* TLS without client validation
```

Step 4 **show voice register pool <index>**

Show command to display whether **sip-oauth** is enabled in a Secure SIP SRST **voice register pool**.

Example:

```
Router#show voice register pool  1
 Pool Tag 1
Config:
  Proxy Ip address is 0.0.0.0
  DTMF Relay is disabled
  kpml signal is enabled
  Lpcor Type is none

 SIP OAuth is enabled
  Reason for unregistered state: reboot

  paging-dn: config 0 [multicast]  effective 0 [multicast]

VRF:
  NA

Dialpeers created:

Statistics:
  Active registrations  : 0

  Total SIP phones registered: 0
  Total Registration Statistics
    Registration requests  : 0
    Registration success   : 0
    Registration failed    : 0
    unRegister requests    : 0
    unRegister success     : 0
    unRegister failed      : 0
    Auto-Register requests : 0
    Attempts to register
           after last unregister : 0
    Last register request time   :
    Last unregister request time :
    Register success time        :
```

```
    Unregister success time      :
```

Step 5 **show voice register statistics**

Show command to display statistics and ouput for success and error registration flows.

Example:

```
gw1-2a#show voice register statistics
Global statistics
  Active registrations  : 0

  Total SIP phones registered: 0
  Total Registration Statistics
    Registration requests  : 244
    Registration success   : 125
    Registration failed    : 119
    unRegister requests    : 121
    unRegister success     : 121
    unRegister failed      : 0
    Auto-Register requests : 0
    Attempts to register
           after last unregister : 0
    Last register request time   : 22:04:30.574 clock Tue Dec 21 2021
    Last unregister request time : 22:08:38.146 clock Tue Dec 21 2021
    Register success time        : 22:04:30.577 clock Tue Dec 21 2021
    Unregister success time      : 22:08:38.147 clock Tue Dec 21 2021

Register pool 29 statistics
  Active registrations  : 0

  Total SIP phones registered: 0
  Total Registration Statistics
    Registration requests  : 12
    Registration success   : 12
    Registration failed    : 0
    unRegister requests    : 12
    unRegister success     : 12
    unRegister failed      : 0
    Auto-Register requests : 0
    Attempts to register
           after last unregister : 0
    Last register request time   : 13:07:53.523 clock Tue Dec 21 2021
    Last unregister request time : 13:12:01.716 clock Tue Dec 21 2021
    Register success time        : 13:07:53.523 clock Tue Dec 21 2021
    Unregister success time      : 13:12:01.716 clock Tue Dec 21 2021


  Reason for unregistered state:
       No registration request since last reboot/unregister
```

Step 6 **show voip sip-oauth key-server status**

Show command to display key retrieval details for SIP OAuth.

Note The output is the same for IPv6 except that the key-server address is an IPv6 address.

Example:

```
Router#show voip sip-oauth key-server status
Key-server:                 10.1.10.50
Last Request Time:          11:40:58.389 UTC Fri Nov 12 2021
Last Success response Time: 11:40:58.456 UTC Fri Nov 12 2021
Current Status:             SUCCESS
```

```
Next Request Time:          11:40:58.389 UTC Sat Nov 13 2021
Total requests sent:        13
Total success responses:    3
Total failure responses:    10
```

SHA2-Cipher-Only Mode for Unified Secure SRST

From SRST 14.2 onwards, the ciphers that secure SIP SRST offers only ciphers that meet your compliance requirements. Similarly, configure secure SCCP SRST to allow SHA2 TLS1.2 ciphers.

From SRST 14.4 onwards, Secure SCCP SRST also supports SHA2 ciphers with TLS v1.3.

SCCP Client Registration

When SCCP SRST is configured with SHA2 ciphers, SCCP clients must use one of the following SHA2 cipher suites to establish a TLS connection:

- **TLS 1.2 SHA2:**
  - `ECDHE_RSA_AES_256_GCM_SHA256`
  - `ECDHE_RSA_AES_256_GCM_SHA384`
  - `DHE_RSA_AES128_GCM_SHA256`
  - `DHE_RSA_AES256_GCM_SHA384`
  - `ECDHE_ECDSA_AES128_GCM_SHA256`
  - `ECDHE_ECDSA_AES256_GCM_SHA384`
- **TLS 1.3 and TLS1.3 SHA2:**
  - `AES128_GCM_SHA256`
  - `AES256_GCM_SHA384`
  - `CHACHA20_POLY1305_SHA256`

Media packets are encrypted and sent using the AEAD_AES_256_GCM SRTP cipher suite.

Note When Secure SCCP SRST is configured to require SHA2 ciphers, only clients using SCCP protocol version 23 or higher are allowed to register. If SHA2 isn't configured as a requirement for Secure SCCP SRST, then clients using SCCP protocol version 23 or lesser may be used.

SIP Client Registration

Secure SIP SRST may be configured to allow SIP clients to establish a TLS connection using single or multiple preferred cipher suites.

For example:

```
Device(config)#voice class tls-cipher 333
Device(config-class)#cipher 1 ?
AES128_GCM_SHA256              supported in TLS 1.3
AES256_GCM_SHA384              supported in TLS 1.3
CHACHA20_POLY1305_SHA256       supported in TLS 1.3
DHE_RSA_AES128_GCM_SHA256      supported in TLS 1.2
DHE_RSA_AES256_GCM_SHA384      supported in TLS 1.2
DHE_RSA_WITH_AES_128_CBC_SHA   supported in TLS 1.2 & below
DHE_RSA_WITH_AES_256_CBC_SHA   supported in TLS 1.2 & below
ECDHE_ECDSA_AES128_GCM_SHA256  supported in TLS 1.2
ECDHE_ECDSA_AES256_GCM_SHA384  supported in TLS 1.2
ECDHE_RSA_AES128_GCM_SHA256    supported in TLS 1.2
ECDHE_RSA_AES256_GCM_SHA384    supported in TLS 1.2
RSA_WITH_AES_128_CBC_SHA       supported in TLS 1.2 & below
RSA_WITH_AES_256_CBC_SHA       supported in TLS 1.2 & below

Device(config-class)#cipher 1 DHE_RSA_AES128_GCM_SHA256
Device(config-class)#end
```

Note Configure SRST TLS cipher policy before a SIP client is allowed to connect and register.

After the successful signaling, media packets are encrypted based on the **srtp-crypto** configuration. Configure an SRTP cipher list first using the **voice class srtp-crypto** <*tag*> command. Associate the SRTP cipher list with the voice register pool.

```
Device(config)#voice class srtp-crypto 22
Device(config-class)#?
VOICECLASS configuration commands:
crypto      Configure preferred SRTP cipher-suite
exit        Exit from voice class configuration mode
help        Description of the interactive help system
no          Negate a command or set its defaults

Device(config-class)#crypto ?
<1-4>       Set the preference order for the cipher-suite (1 = Highest)

Device(config-class)#crypto 1 ?
 AEAD_AES_128_GCM         Allow secure calls with SRTP AEAD_AES_128_GCM cipher-suite
 AEAD_AES_256_GCM         Allow secure calls with SRTP AEAD_AES_256_GCM cipher-suite
 AES_CM_128_HMAC_SHA1_32  Allow secure calls with SRTP AES_CM_128_HMAC_SHA1_32 cipher-suite

 AES_CM_128_HMAC_SHA1_80  Allow secure calls with SRTP AES_CM_128_HMAC_SHA1_80 cipher-suite

Device(config-class)#crypto 1 AEAD_AES_256_GCM
Device(config-class)#do show run | sec srtp-cry
 voice class srtp-crypto 22
 crypto 1 AEAD_AES_256_GCM

Device(config)# voice register pool 17
Device(config-register-pool)# id network 10.1.10.217 mask 255.255.255.255
Device(config-register-pool)# dtmf-relay rtp-nte
Device(config-register-pool)# codec g711ulaw

Show run output for pool:
============================================
Device#show running-config | sec voice register pool 17
 voice register pool 17
 id network 10.1.10.217 mask 255.255.255.255
 dtmf-relay rtp-nte
 voice-class srtp-crypto 22
 codec g711ulaw
Device#show run
```

Configure **srtp-crytpto 23** command, which is present, you get the following error:

```
Device(config-register-pool)#voice-class srtp-crypto 23
ERROR: There is no voice-class srtp-crypto 23
```

Configure **srtp-crytpto 22** command, which is present, you get the following output:

```
Device(config-register-pool)#voice-class srtp-crypto 22
Device(config-register-pool)#
```

Note SRTP crypto policy must be configured before it can be used in a voice register pool configuration.

Benefits

When you configure SHA2 cipher suite with TLS version 1.2, you get the following benefits:

- Improved security as SHA2 cipher suites provides more reliable security certificates.
- Fast computation.
- Resistance to collision attacks.

Configure SHA2 Cipher Suite with TLS

SUMMARY STEPS

1. **configure terminal**
2. **call-manager-fallback**
3. **transport-tcp-tls {v1.2 | v1.3} [sha2]**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**configure terminal** **Example:** `Router#configure terminal`	Enters global configuration mode.

Step 2	**call-manager-fallback**
	Example:
	`Router(config)#call-manager-fallback`
	`Router(config-cm-fallback)`
		Enters config-cm-fallback mode.

Step 3	**transport-tcp-tls {v1.2 | v1.3} [sha2]** **Example:** `Router(config-cm-fallback)#transport-tcp-tls v1.2 sha2` **Example:**	Configures the SHA2 cipher suite for TLS version 1.2 or 1.3 on the router.

	Command or Action	Purpose
	`Router(config-cm-fallback)#transport-tcp-tls v1.3 sha2`	

Media Security on Unified SRST - SRTP

Media encryption, which uses Secure Real-Time Protocol (SRTP), ensures that only the intended recipient can interpret the media streams between supported devices. Support includes audio streams only.

If the devices support SRTP, the system uses a SRTP connection. If at least one device does not support SRTP, the system uses an RTP connection. SRTP-to-RTP fallback may occur for transfers from a secure device to a non-secure device for music-on-hold (MOH), and so on.

Note Secure SRST handles media encryption keys differently for different devices and protocols. All phones that are running SCCP get their media encryption keys from SRST, which secures the media encryption key downloads to phones with TLS encrypted signaling channels. Phones that are running SIP generate and store their own media encryption keys. Media encryption keys that are derived by SRST securely get sent through encrypted signaling paths to gateways over IPSec-protected links for H.323.

Warning Before you configure SRTP or signaling encryption for gateways and trunks, Cisco strongly recommends that you configure IPSec because Cisco H.323 gateways, and H.323/H.245/H.225 trunks rely on IPSec configuration to ensure that security-related information does not get sent in the clear. Cisco Unified SRST does not verify that you configured IPSec correctly. If you do not configure IPSec correctly, security-related information may get exposed.

Establishment of Secure Cisco Unified SRST to the Cisco Unified IP Phone

The following figure shows the interworking of the credentials server on the SRST router, Cisco Unified Communications Manager, and the Cisco Unified IP Phone. The following table describes the establishment of secure SRST to the Cisco Unified IP Phone.

Figure 2: Interworking of Credentials Server on SRST Router, Cisco Unified Communications Manager, and Cisco Unified IP Phone

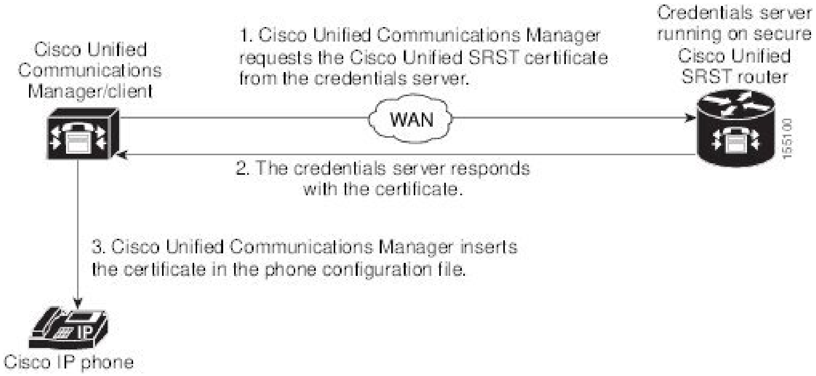

Cisco Unified Communications Manager/client
1. Cisco Unified Communications Manager requests the Cisco Unified SRST certificate from the credentials server.
Credentials server running on secure Cisco Unified SRST router
WAN
155100
2. The credentials server responds with the certificate.
3. Cisco Unified Communications Manager inserts the certificate in the phone configuration file.
IP
Cisco IP phone

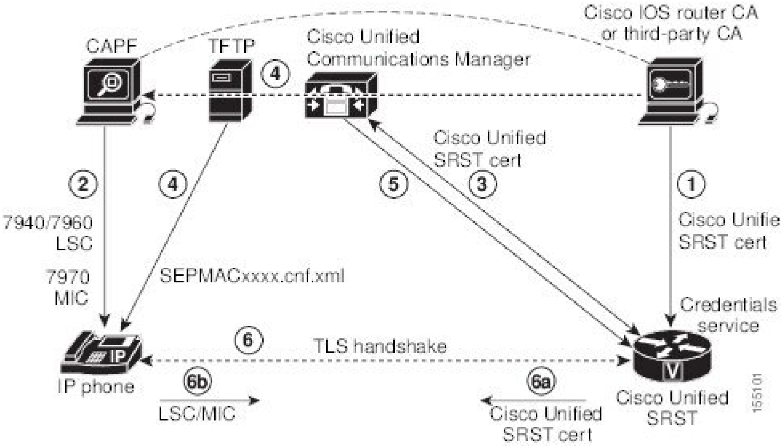
Table 2: Establishing Secure SRST

Mode	Process	Description or Detail
Regular Mode	The Cisco Unified IP Phone configures DHCP and gets the TFTP server address.	—
	The Cisco Unified IP Phone retrieves a CTL file from the TFTP server.	The CTL file contains the certificates that the phone should trust.
	The Cisco IP Phone opens a Transport Layer Security (TLS) protocol channel and registers to Cisco Unified Communications Manager.	Cisco Unified Communications Manager exports secure Cisco Unified SRST router information and the Cisco Unified SRST router certificate to the Cisco Unified IP phone. The phone places the certificate into its configuration. Once the phone has the Cisco Unified SRST certificate, the Cisco Unified SRST router is considered secure. See Figure Interworking of Credentials Server on SRST Router, Cisco Unified Communications Manager, and Cisco Unified IP Phone.
	If the Cisco Unified IP Phone is configured as “authenticated” or “encrypted” and Cisco Unified Communications Manager is configured in mixed mode, the phone looks for an SRST certificate in its configuration file. If it finds an SRST certificate, it opens a standby TLS connection to the default port. The default port is the Cisco Unified IP Phone TCP port plus 443; that is, port 2443 on a Cisco Unified SRST router.	The connection to the SRST router happens automatically, assuming there is not a secondary Cisco Unified Communications Manager and Cisco Unified SRST is configured as the backup device. See Figure Interworking of Credentials Server on SRST Router, Cisco Unified Communications Manager, and Cisco Unified IP Phone. Cisco Unified Communications Manager should be configured in mixed mode, which is its secure mode.
In case of WAN failure, the Cisco Unified IP Phone starts Cisco Unified SRST registration.		
SRST	The Cisco Unified IP Phone registers with the SRST router at the default port for secure communications.	—

Secure SRST Authentication and Encryption

The following figure illustrates the process of secure SRST authentication and encryption, and the following table describes the process.

Figure 3: Secure Cisco Unified SRST Authentication and Encryption



Process Steps	Description or Detail
1.	The CA server, whether it is a Cisco IOS router CA or a third-party CA, issues a device certificate to the SRST gateway, enabling credentials service. Optionally, the certificate can be self-generated by the SRST router using a Cisco IOS CA server. The CA router is the ultimate trustpoint for the Certificate Authority Proxy Function (CAPF). For more information on CAPF, see Cisco Communications Manager Security Guide.
2.	The CAPF is a process where supported devices can request a locally significant certificate (LSC). The CAPF utility generates a key pair and certificate that is specific for CAPF, copies this certificate to all Cisco Unified Communications Manager servers in the cluster, and provides the LSC to the Cisco Unified IP Phone. An LSC is required for Cisco Unified IP Phones that do not have a manufacturing installed certificate (MIC). The Cisco 7970 is equipped with a MIC and therefore does not need to go through the CAPF process.
3.	Cisco Unified Communications Manager requests the SRST certificate from credentials server, and the credentials server responds with the certificate.
4.	For each device, Cisco Unified CM uses the TFTP process and inserts the certificate into the SEPMACxxxx.cnf.xml configuration file of the Cisco Unified IP Phone.
5.	Cisco Unified CM provides the PEM format files that contain phone certificate information to the Cisco Unified SRST router. Providing the PEM files to the Cisco Unified SRST router is done manually. See Cisco IOS Credentials Server on Secure SRST Routers section. When the Cisco Unified SRST router has the PEM files, the Cisco Unified SRST Router can authenticate the IP phone and validate the issuer of the IP phones certificate during the TLS handshake.

Process Steps	Description or Detail
6.	The TLS handshake occurs, certificates are exchanged, and mutual authentication and registration occurs between the Cisco Unified IP Phone and the Cisco Unified SRST Router.
a.	The Cisco Unified SRST Router sends its certificate, and the phone validates the certificate to the certificate that it received from Cisco Unified CM in Step 4.
b.	The Cisco Unified IP Phone provides the Cisco Unified SRST Router the LSC or MIC, and the router validates the LSC or MIC using the PEM format files that it was provided in Step 5.

Note The media is encrypted automatically after the phone and router certificates are exchanged and the TLS connection is established with the SRST router.

How to Configure Secure Unified SRST

The following configuration sections ensure that the secure Cisco Unified SRST Router and the Cisco Unified IP Phones can request mutual authentication during the TLS handshake. The TLS handshake occurs when the phone registers with the Cisco Unified SRST Router, either before or after the WAN link fails.

This section contains the following procedures:

Preparing the Cisco Unified SRST Router for Secure Communication

The following tasks prepare the Cisco Unified SRST Router to process secure communications.

Configuring a Certificate Authority Server on a Cisco IOS Certificate Server

For Cisco Unified SRST Routers to provide secure communications, there must be a CA server that issues the device certificate in the network. The CA server can be a third-party CA or one generated from a Cisco IOS certificate server.

The Cisco IOS certificate server provides a certificate generation option to users who do not have a third-party CA in their network. The Cisco IOS certificate server can run on the SRST router or on a different Cisco IOS router.

If you do not have a third-party CA, full instructions on enabling and configuring a CA server can be found in the Cisco IOS Certificate Server documentation. A sample configuration is provided below.

SUMMARY STEPS

1. **crypto pki server** *cs-label*
2. **database level** {**minimal** | **names** |**complete**}
3. **database url** *root-url*
4. **issuer-name** *DN-string*
5. **grant auto**
6. **no shutdown**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**crypto pki server** *cs-label* **Example:** `Router (config)# crypto pki server srstcaserver`	Enables the certificate server and enters certificate server configuration mode. **Note** If you manually generated an RSA key pair, the *cs-label* argument must match the name of the key pair. For more information on the certificate server, see the Cisco IOS Certificate Server documentation.

Step 2	**database level** {**minimal** | **names** |**complete**}
	Example:
	`Router (cs-server)# database level complete`
		Controls what type of data is stored in the certificate enrollment database.
		• **minimal**: Enough information is stored only to continue issuing new certificates without conflict; this is the default.
		• **names**: In addition to the information given in the minimal level, the serial number and subject name of each certificate are stored.
		• **complete**: In addition to the information given in the minimal and names levels, each issued certificate is written to the database.
		Note The **complete** keyword produces a large amount of information; if it is issued, you should also specify an external TFTP server on which to store the data using the **database url** command.

Step 3	**database url** *root-url* **Example:** `Router (cs-server)# database url nvram`	Specifies the location where all database entries for the certificate server will be written. After you create a certificate server using the **crypto pki server** command, use this command to specify a combined list of all the certificates that have been issued. The *root-url* argument specifies the location where database entries are written. • The default location for the database entries to be written is flash; however, NVRAM is recommended for this task.
Step 4	**issuer-name** *DN-string* **Example:** `Router (cs-server)# issuer-name CN=srstcaserver`	Sets the CA issuer name to the specified distinguished name (DN-string). The default value is as follows: **issuer-name CN**= *cs-label* .
Step 5	**grant auto** **Example:**	Allows an automatic certificate to be issued to any requestor.

	Command or Action	Purpose
	`Router (cs-server)# grant auto`	• This command is used only during enrollment and will be removed in the Disabling Automatic Certificate Enrollment section.
Step 6	**no shutdown** **Example:** `Router (cs-server)# no shutdown`	Enables the Cisco IOS certificate server. • You should issue this command only after you have completely configured your certificate server.

Example

The following example reflects one way of generating a CA:

```
Router(config)# crypto pki server srstcaserver
Router(cs-server)# database level complete
Router(cs-server)# database url nvram
Router(cs-server)# issuer-name CN=srstcaserver
Router(cs-server)# grant auto
% This will cause all certificate requests to be automatically granted.
Are you sure you want to do this? [yes/no]: y
Router(cs-server)# no shutdown
% Once you start the server, you can no longer change some of
% the configuration.
Are you sure you want to do this? [yes/no]: y
% Generating 1024 bit RSA keys ...[OK]
% Certificate Server enabled.
```

Autoenrolling and Authenticating the Secure Cisco Unified SRST Router to the CA Server

The secure Cisco Unified SRST Router needs to define a trustpoint; that is, it must obtain a device certificate from the CA server. The procedure is called certificate enrollment. Once enrolled, the secure Cisco Unified SRST Router can be recognized by Cisco Unified Communications Manager as a secure SRST router.

There are three options to enroll the secure Cisco Unified SRST Router to a CA server: autoenrollment, cut and paste, and TFTP. When the CA server is a Cisco IOS certificate server, autoenrollment can be used. Otherwise, manual enrollment is required. Manual enrollment refers to cut and paste or TFTP.

Use the **enrollment url** command for autoenrollment and the **crypto pki authenticate** command to authenticate the SRST router. Full instructions for the commands can be found in the Certification Authority Interoperability Commands documentation. An example of autoenrollment is available in the Certificate Enrollment Enhancements feature. A sample configuration is provided in the .

SUMMARY STEPS

1. **crypto pki trustpoint***name*
2. **rsakeypair** *keypair-label*
3. **enrollment url** *url*
4. **revocation-check** *method1*
5. **exit**
6. **crypto pki authenticate** *name*
7. **crypto pki enroll** *name*

DETAILED STEPS

	Command or Action	Purpose
Step 1	**crypto pki trustpoint***name* **Example:** `Router(config)# crypto pki trustpoint srstca`	Declares the CA that your router should use and enters ca-trustpoint configuration mode. • The name provided will be the same as the trustpoint name that will be declared in the Enabling Credentials Service on the Secure Cisco Unified SRST Router section.
Step 2	**rsakeypair** *keypair-label* **Example:** `Router(config-trustp)# rsakeypair srstcakey 2048`	To specify a named Rivest, Shamir, and Adelman (RSA) key pair for this trustpoint, use the **rsakeypair** command in trustpoint configuration mode. • Configure the RSA key length to 2048 bits or above.
Step 3	**enrollment url** *url* **Example:** `Router(ca-trustpoint)# enrollment url http://10.1.1.22`	Specifies the enrollment parameters of your CA. • **url** *url*: Specifies the URL of the CA to which your router should send certificate requests. • If you are using Cisco proprietary SCEP for enrollment, url must be in the form http://*CA_name*, where *CA_name* is the host Domain Name System (DNS) name or IP address of the Cisco IOS CA. • If you used the procedure documented in the Configuring a Certificate Authority Server on a Cisco IOS Certificate Server section, the URL is the IP address of the certificate server router configured in Step 1. If a third-party CA was used, the IP address is to an external CA.
Step 4	**revocation-check** *method1* **Example:** `Router(ca-trustpoint)# revocation-check none`	Checks the revocation status of a certificate. The argument *method1* is the method used by the router to check the revocation status of the certificate. For this task, the only available method is **none**. The keyword **none** means that a revocation check will not be performed and the certificate will always be accepted. • Using the **none** keyword is mandatory for this task.
Step 5	**exit** **Example:** `Router(ca-trustpoint)# exit`	Exits ca-trustpoint configuration mode and returns to global configuration mode.
Step 6	**crypto pki authenticate** *name* **Example:** `Router(config)# crypto pki authenticate srstca`	Authenticates the CA (by getting the certificate from the CA). • Takes the name of the CA as the argument.

	Command or Action	Purpose
Step 7	**crypto pki enroll** *name* **Example:** `Router(config)# crypto pki enroll srstca`	Obtains the SRST router certificate from the CA. • Takes the name of the CA as the argument.

Example

The following example autoenrolls and authenticates the Cisco Unified SRST router:

```
Router(config)# crypto pki trustpoint srstca
Router(ca-trustpoint)# enrollment url http://10.1.1.22
Router(ca-trustpoint)# revocation-check none
Router(ca-trustpoint)# exit
Router(config)# crypto pki authenticate srstca
Certificate has the following attributes:
Fingerprint MD5: 4C894B7D 71DBA53F 50C65FD7 75DDBFCA
Fingerprint SHA1: 5C3B6B9E EFA40927 9DF6A826 58DA618A BF39F291
% Do you accept this certificate? [yes/no]: y
Trustpoint CA certificate accepted.
Router(config)# crypto pki enroll srstca
%
% Start certificate enrollment ..
% Create a challenge password. You will need to verbally provide this
password to the CA Administrator in order to revoke your certificate.
For security reasons your password will not be saved in the configuration.
Please make a note of it.
Password:
Re-enter password:
% The fully-qualified domain name in the certificate will be: router.cisco.com
% The subject name in the certificate will be: router.cisco.com
% Include the router serial number in the subject name? [yes/no]: y
% The serial number in the certificate will be: D0B9E79C
% Include an IP address in the subject name? [no]: n
Request certificate from CA? [yes/no]: y
% Certificate request sent to Certificate Authority
% The certificate request fingerprint will be displayed.
% The 'show crypto pki certificate' command will also show the fingerprint.
Sep 29 00:41:55.427: CRYPTO_PKI: Certificate Request Fingerprint MD5: D154FB75
2524A24D 3D1F5C2B 46A7B9E4
Sep 29 00:41:55.427: CRYPTO_PKI: Certificate Request Fingerprint SHA1: 0573FBB2
98CD1AD0 F37D591A C595252D A17523C1
Sep 29 00:41:57.339: %PKI-6-CERTRET: Certificate received from Certificate Authority
```

Disabling Automatic Certificate Enrollment

The command **grant auto** allows certificates to be issued and was activated in the optional task documented in the Configuring a Certificate Authority Server on a Cisco IOS Certificate Server section.

Note You should disable the **grant auto** command so that certificates cannot be continually granted.

SUMMARY STEPS

1. **crypto pki server***cs-label*

2. **shutdown**
3. **no grant auto**
4. **no shutdown**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**crypto pki server***cs-label* **Example:** `Router (config)# crypto pki server srstcaserver`	Enables the certificate server and enters certificate server configuration mode. **Note** If you manually generated an RSA key pair, the *cs-label* argument must match the name of the key pair.
Step 2	**shutdown** **Example:** `Router (cs-server)# shutdown`	Disables the Cisco IOS certificate server.
Step 3	**no grant auto** **Example:** `Router (cs-server)# no grant auto`	Disables automatic certificates to be issued to any requestor. • This command was for use during enrollment only and thus needs to be removed in this task.
Step 4	**no shutdown** **Example:** `Router (cs-server)# no shutdown`	Enables the Cisco IOS certificate server. • You should issue this command only after you have completely configured your certificate server.

What to do next

For manual enrollment instructions, see the Manual Certificate Enrollment (TFTP and Cut-and-Paste) feature.

Verifying Certificate Enrollment

If you used the Cisco IOS certificate server as your CA, use the **show running-config** command to verify certificate enrollment or the **show crypto pki server** command to verify the status of the CA server.

SUMMARY STEPS

1. **show running-config**
2. **show crypto pki server**

DETAILED STEPS

	Command or Action	Purpose

Step 1	**show running-config**
	Example:
	`Router# show running-config`
	`.`
	`.`
		Use the **show running-config** command to verify the creation of the CA server (01) and device (02) certificates. This example shows the enrolled certificates.

	Command or Action	Purpose
	. ! SRST router device certificate. crypto pki certificate chain srstca certificate 02 308201AD 30820116 A0030201 02020102 300D0609 2A864886 F70D0101 04050030 17311530 13060355 0403130C 73727374 63617365 72766572 301E170D 30343034 31323139 35323233 5A170D30 35303431 32313935 3232335A 30343132 300F0603 55040513 08443042 39453739 43301F06 092A8648 86F70D01 09021612 6A61736F 32363931 2E636973 636F2E63 6F6D305C 300D0609 2A864886 F70D0101 01050003 4B003048 024100D7 0CC354FB 5F7C1AE7 7A25C3F2 056E0485 22896D36 6CA70C19 C98F9BAE AE9D1F9B D4BB7A67 F3251174 193BB1A3 12946123 E5C1CCD7 A23E6155 FA2ED743 3FB8B902 03010001 A330302E 300B0603 551D0F04 04030205 A0301F06 03551D23 04183016 8014F829 CE97AD60 18D05467 FC293963 C2470691 F9BD300D 06092A86 4886F70D 01010405 00038181 007EB48E CAE9E1B3 D1E7A185 D7F0D565 CB84B17B 1151BD78 B3E39763 59EC650E 49371F6D 99CBD267 EB8ADF9D 9E43A5F2 FB2B18A0 34AF6564 11239473 41478AFC A86E6DA1 AC518E0B 8657CEBB ED2BDE8E B586FE67 00C358D4 EFDD8D44 3F423141 C2D331D3 1EE43B6E 6CB29EE7 0B8C2752 C3AF4A66 BD007348 D013000A EA3C206D CF quit certificate ca 01 30820207 30820170 A0030201 02020101 300D0609 2A864886 F70D0101 04050030 17311530 13060355 0403130C 73727374 63617365 72766572 301E170D 30343034 31323139 34353136 5A170D30 37303431 32313934 3531365A 30173115 30130603 55040313 0C737273 74636173 65727665 7230819F 300D0609 2A864886 F70D0101 01050003 818D0030 81890281 8100C3AF EE1E4BB1 9922A8DA 2BB9DC8E 5B1BD332 1051C9FE 32A971B3 3C336635 74691954 98E765B1 059E24B6 32154E99 105CA989 9619993F CC72C525 7357EBAC E6335A32 2AAF9391 99325BFD 9B8355EB C10F8963 9D8FC222 EE8AC831 71ACD3A7 4E918A8F D5775159 76FBF499 5AD0849D CAA41417 DD866902 21E5DD03 C37D4B28 0FAB0203 010001A3 63306130 0F060355 1D130101 FF040530 030101FF 300E0603 551D0F01 01FF0404 03020186 301D0603 551D0E04 160414F8 29CE97AD 6018D054 67FC2939 63C24706 91F9BD30 1F060355 1D230418 30168014 F829CE97 AD6018D0 5467FC29 3963C247 0691F9BD 300D0609 2A864886 F70D0101 04050003 8181007A F71B25F9 73D74552 25DFD03A D8D1338F 6792C805 47A81019 795B5AAE 035400BB F859DABF 21892B5B E71A8283 08950414 8633A8B2 C98565A6 C09CA641 88661402 ACC424FD 36F23360	

	Command or Action	Purpose
	ABFF4C55 BB23C66A C80A3A57 5EE85FF8 C1B1A540 E818CE6D 58131726 BB060974 4E1A2F4B E6195522 122457F3 DEDBAAD7 3780136E B112A6 quit	
Step 2	**show crypto pki server** **Example:** Router# show crypto pki server Certificate Server srstcaserver: Status: enabled Server's configuration is locked (enter "shut" to unlock it) Issuer name: CN=srstcaserver CA cert fingerprint: AC9919F5 CAFE0560 92B3478A CFF5EC00 Granting mode is: auto Last certificate issued serial number: 0x2 CA certificate expiration timer: 13:46:57 PST Dec 1 2007 CRL NextUpdate timer: 14:54:57 PST Jan 19 2005 Current storage dir: nvram Database Level: Complete - all issued certs written as <serialnum>.cer	Use the **show crypto pki server** command to verify the status of the CA server after a boot procedure.

Enabling Credentials Service on the Secure Cisco Unified SRST Router

Once the Cisco Unified SRST Router has its own certificate, you need to provide Cisco Unified Communications Manager the certificate. Enabling credentials service allows Cisco Unified Communications Manager to retrieve the secure SRST device certificate and place it in the configuration file of the Cisco Unified IP Phone.

Activate credentials service on all Cisco Unified SRST Routers.

Note TLS v1.3 support has been added for the TLS handshake between CUCM and SRST to fetch the certificate using the credentials service.

Note A security best practice is to protect the credentials service port using Control Plane Policing. Control Plane Policing protects the gateway and maintains packet forwarding and protocol states despite a heavy traffic load. For more information on control planes, see the Control Plane Policing documentation. In addition, a sample configuration is given in the Control Plane Policing: Example section.

SUMMARY STEPS

1. **credentials**
2. **ip source-address** *ip-address* [**port** *port*]
3. **trustpoint** *trustpoint-name*
4. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**credentials** **Example:** `Router(config)# credentials`	Provides the Cisco Unified SRST Router certificate to Cisco Unified Communications Manager and enters credentials configuration mode.
Step 2	**ip source-address** *ip-address* [**port** *port*] **Example:** `Router(config-credentials)# ip source-address 10.1.1.22 port 2445`	Enables the Cisco Unified SRST Router to receive messages from Cisco Unified Communications Manager through the specified IP address and port. • *ip-address*: The IP address is the pre-existing router IP address, typically one of the addresses of the Ethernet port of the router. • **port** *port*: (Optional) The port to which the gateway router connects to receive messages from Cisco Unified Communications Manager. The port number is from 2000 to 9999. The default port number is 2445.
Step 3	**trustpoint** *trustpoint-name* **Example:** `Router(config-credentials)# trustpoint srstca`	Specifies the name of the trustpoint that is to be associated with the Cisco Unified SRST Router certificate. The *trustpoint-name* argument is the name of the trustpoint and corresponds to the SRST device certificate. • The trustpoint name should be the same as the one declared in the Autoenrolling and Authenticating the Secure Cisco Unified SRST Router to the CA Server section.
Step 4	**exit** **Example:** `Router(config-credentials)# exit`	Exits credentials configuration mode.

Example

```
Router(config)# credentials
Router(config-credentials)# ip source-address 10.1.1.22 port 2445
Router(config-credentials)# trustpoint srstca
Router(config-credentials)# exit
```

Troubleshooting Credential Settings

The following steps display credential settings or set debugging on the credential settings of the Cisco Unified SRST Router.

SUMMARY STEPS

1. **show credentials**
2. **debug credentials**

DETAILED STEPS

	Command or Action	Purpose

Step 1	**show credentials**
	Example:
	`Router# show credentials`
	`Credentials IP: 10.1.1.22`
	`Credentials PORT: 2445`
	`Trustpoint: srstca`
		Use the **show credentials** command to display the credential settings on the Cisco Unified SRST Router that are supplied to Cisco Unified Communications Manager for use during secure Cisco Unified SRST fallback.

Step 2	**debug credentials**
	Example:
	`Router# debug credentials`
	`Credentials server debugging is enabled`
	`Router#`
	`Sep 29 01:01:50.903: Credentials service: Start`
	`TLS`
	`Handshake 1 10.1.1.13 2187`
	`Sep 29 01:01:50.903: Credentials service: TLS`
	`Handshake returns OPSSLReadWouldBlockErr`
	`Sep 29 01:01:51.903: Credentials service: TLS`
	`Handshake returns OPSSLReadWouldBlockErr`
	`Sep 29 01:01:52.907: Credentials service: TLS`
	`Handshake returns OPSSLReadWouldBlockErr`
	`Sep 29 01:01:53.927: Credentials service: TLS`
	`Handshake completes.`
		Use the **debug credentials** command to set debugging on the credential settings of the Cisco Unified SRST Router.

Importing Phone Certificate Files in PEM Format to the Secure SRST Router

This task completes the tasks required for Cisco IP Unified Phones to authenticate secure SRST.

Cisco Unified Communications Manager 4.X.X and Earlier Versions

For systems running Cisco Unified Communications Manager 4.X.X and earlier versions, the secure Cisco Unified SRST Router must retrieve phone certificates so that it can authenticate Cisco Unified IP phones during the TLS handshake. Different certificates are used for different Cisco Unified IP Phones. The Supported Cisco Unified IP Phones and Certificates table lists the certificates needed for each type of phone.

Certificates must be imported manually from Cisco Unified Communications Manager to the Cisco Unified SRST Router. The number of certificates depends on the Cisco Unified Communications Manager configuration. Manual enrollment refers to cut and paste or TFTP. For manual enrollment instructions, see the Manual Certificate Enrollment (TFTP and Cut-and-Paste) feature. Repeat the enrollment procedure for each phone or PEM file.

For Cisco Unified Communications Manager 4.X.X and earlier versions, certificates are found by going to the menu bar in Cisco Unified Communications Manager, choose **Program Files** > **Cisco** > **Certificates**.

Open the .0 files with Windows WordPad or Notepad, and copy and paste the contents to the SRST router console. Then, repeat the procedure with the .pem file. Copy all the contents that appear between "-----BEGIN CERTIFICATE-----" and "-----END CERTIFICATE-----".

For certification operation on Cisco Unified Communications Operating System Administration Guide, Release 6.1(1), see http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/cucos/6_1_1/cucos/iptpch6.html.

Cisco Unified Communications Manager 5.0 and Later Versions

Systems running Cisco Unified CM 5.0 and later versions require four certificates (CAPF, CiscoCA, CiscoManufactureCA, and CiscoRootCA2048) in addition to the requirements listed in the Supported Cisco Unified IP Phones and Certificates table, which must be copied and pasted to Cisco Unified SRST Routers.

Note CiscoRootCA is also called CiscoRoot2048CA.

For Cisco Unified CM 5.0 and later versions, perform the following steps:

Before you begin

You must have certificates available when the last configuration command (**crypto pki authenticate**) issues the following prompt:

```
Enter the base 64 encoded CA certificate.
End with a blank line or the word "quit" on a line by itself
```

SUMMARY STEPS

1. Login to Cisco Unified Communications Manager.
2. Go to **Security** > **Certificate Management** > **Download Certificate/CTL**.
3. Select **Download Trust Cert** and click **Next**.
4. Select **CAPF-trust** and click **Next**.
5. Select **CiscoCA** and click **Next**.
6. Click **Continue**.
7. Click the file name.
8. Copy all the contents that appear between “-----BEGIN CERTIFICATE-----” and “-----END CERTIFICATE-----” to a location where you can retrieve it later.
9. Repeat Steps 5 to 8 for CiscoManufactureCA, CiscoRootCA2048, and CAPF.

DETAILED STEPS

	Command or Action	Purpose
Step 1	Login to Cisco Unified Communications Manager.	
Step 2	Go to **Security** > **Certificate Management** > **Download Certificate/CTL**.	
Step 3	Select **Download Trust Cert** and click **Next**.	
Step 4	Select **CAPF-trust** and click **Next**.	
Step 5	Select **CiscoCA** and click **Next**.	
Step 6	Click **Continue**.	
Step 7	Click the file name.	
Step 8	Copy all the contents that appear between “-----BEGIN CERTIFICATE-----” and “-----END CERTIFICATE-----” to a location where you can retrieve it later.	

	Command or Action	Purpose
Step 9	Repeat Steps 5 to 8 for CiscoManufactureCA, CiscoRootCA2048, and CAPF.	

Cisco Unified Communications Manager 6.0 and Later Versions

From Cisco Unified Communications Operating System Administration, download all certificates listed under CAPF-trust, including Cisco_Manufacturing_CA, Cisco_Root_CA_2048, CAP-RTP-001, CAP-RTP-002, CAPF, and CAPF-*xxx*. Also download any CAPF-*xxx* certificates that are listed under CallManager-trust and not under CAPF-trust.

For instructions on downloading certificates, see the "Security" chapter in the appropriate version of Cisco Unified Communications Operating System Administration Guide.

Authenticating the Imported Certificates on the Cisco Unified SRST Router

To authenticate certificates on the Cisco Unified SRST router, perform these steps.

Restrictions

HTTP automatic enrollment from Cisco Unified Communications Manager through a virtual web server is not supported.

SUMMARY STEPS

1. **crypto pki trustpoint** *name*
2. **revocation-check none**
3. **enrollment terminal**
4. **exit**
5. **crypto pki authenticate** *name*

DETAILED STEPS

	Command or Action	Purpose
Step 1	**crypto pki trustpoint** *name* **Example:** `Router (config)# crypto pki trustpoint CAPF`	Declares the CA that your router should use and enters ca-trustpoint configuration mode. • *name*: Enter the name of each certificate individually (for example, CAPF, CiscoCA, CiscoManufactureCA, and CiscoRootCA2048).
Step 2	**revocation-check none** **Example:** `Router(ca-trustpoint)# revocation-check none`	Checks the revocation status of a certificate using the selected method. • Using the **none** keyword is mandatory for this task. The keyword **none** means that a revocation check is not performed and the certificate is always accepted.
Step 3	**enrollment terminal** **Example:** `Router(ca-trustpoint)# enrollment terminal`	Specifies manual cut-and-paste certificate enrollment.

	Command or Action	Purpose
Step 4	**exit** **Example:** `Router(ca-trustpoint)# exit`	Exits ca-trustpoint configuration mode and returns to global configuration.
Step 5	**crypto pki authenticate** *name* **Example:** `Router(config)# crypto pki authenticate CAPF`	Authenticates the CA (by getting the certificate from the CA). • Enter the same *name* argument used in the **crypto pki trustpoint** command in Step 1.

What to do next

Update the certificates in Cisco Unified CM. See the “Configuring a Secure Survivable Remote Site Telephony (SRST) Reference” chapter in the appropriate version of Cisco Unified Communications Manager Security Guide.

Examples

Cisco Unified Communications Manager 4.X.X and Earlier Versions: Example

The following example shows three certificates (Cisco 7970, 7960, PEM) imported to the Cisco Unified SRST Router:

```
Router(config)# crypto pki trustpoint 7970
Router(ca-trustpoint)# revocation-check none
Router(ca-trustpoint)# enrollment terminal
Router(ca-trustpoint)# exit
Router(config)# crypto pki authenticate 7970
Enter the base 64 encoded CA certificate.
End with a blank line or the word "quit" on a line by itself
MIIDqDCCApCgAwIBAgIQNT+yS9cPFKNGwfOprHJWdTANBgkqhkiG9w0BAQUFADAu
MRYwFAYDVQQKEw1DaXNjbyBTeXN0ZW1zMRQwEgYDVQQDEwtDQVAtUlRQLTAwMjAe
Fw0wMzEwMTAyMDE4NDlaFw0yMzEwMTAyMDI3MzdaMC4xFjAUBgNVBAoTDUNpc2Nv
IFN5c3RlbXMxFDASBgNVBAMTC0NBUC1SVFAtMDAyMIIBIDANBgkqhkiG9w0BAQEF
AAOCAQ0AMIIBCAKCAQEAxCZlBK19w/2NZVVvpjCPrpW1cCY7V1q9lhzI85RZZdnQ
2M4CufgIzNa3zYxGJIAYeFfcRECnMB3f5A+x7xNiEuzE87UPvK+7S80uWCY0Uhtl
AVVf5NQgZ3YDNoNXg5MmONb8lT86F55EZyVac0XGne77TSIbIdejrTgYQXGP2MJx
Qhg+ZQlGFDRzbHfM84Duv2Msez+l+SqmqO80kIckqE9Nr3/XCSj1hXZNNVg8D+mv
Hth2P6KZqAKXAAStGRLSZX3jNbS8tveJ3Gi5+sj9+F6KKK2PD0iDwHcRKkcUHb7g
lI++U/5nswjUDIAph715Ds2rn9ehkMGipGLF8kpuCwIBA6OBwzCBwDALBgNVHQ8E
BAMCAYYwDwYDVR0TAQH/BAUwAwEB/zAdBgNVHQ4EFgQUUpIr4ojuLgmKTn5wLFal
mrTUm5YwbwYDVR0fBGgwZjBkoGKgYIYtaHR0cDovL2NhcC1ydHAtMDAyL0NlcnRF
bnJvbGwvQ0FQLVJUUC0wMDIuY3Jshi9maWxlOi8vXFxjYXAtcnRwLTAwMlxDZXJ0
RW5yb2xsXENBUC1SVFAtMDAyLmNybDAQBgkrBgEEAYI3FQEEAwIBADANBgkqhkiG
9w0BAQUFAAOCAQEAVoOM78TaOtHqj7sVL/5u5VChlyvU168f0piJLNWip2vDRihm
E+DlXdwMS5JaqUtuaSd/m/xzxpcRJm4ZRRwPq6VeaiiQGkjFuZEe5jSKiSAK7eHg
tup4HP/ZfKSwPA40DlsGSYsKNMm3OmVOCQUMH02lPkS/eEQ9sIw6QS7uuHN4y4CJ
NPnRbpFRLw06hnStCZHtGpKEHnY213QOy3h/EWhbnp0MZ+hdr20FujSI6G1+L39l
aRjeD708f2fYoz9wnEpZbtn2Kzse3uhU1Ygq1D1x9yuPq388C18HWdmCj4OVTXux
V6Y47H1yv/GJM8FvdgvKlExbGTFnlHpPiaG9tQ==
quit
Certificate has the following attributes:
Fingerprint MD5: F7E150EA 5E6E3AC5 615FC696 66415C9F
Fingerprint SHA1: 1BE2B503 DC72EE28 0C0F6B18 798236D8 D3B18BE6
% Do you accept this certificate? [yes/no]: y
Trustpoint CA certificate accepted.
% Certificate successfully imported
```

```
Router(config)# crypto pki trustpoint 7960
Router(ca-trustpoint)# revocation-check none
Router(ca-trustpoint)# enrollment terminal
Router(ca-trustpoint)# exit
Router(config)# crypto pki authenticate 7960
Enter the base 64 encoded CA certificate.
End with a blank line or the word "quit" on a line by itself
MIICKDCCAZGgAwIBAgIC8wEwDQYJKoZIhvcNAQEFBQAwQDELMAkGA1UEBhMCVVMx
GjAYBgNVBAoTEUNpc2NvIFN5c3RlbXMgSW5jMRUwEwYDVQQDEwxDQVBGLTdEN0Qw
QzAwHhcNMDQwNzE1MjIzODMyWhcNMTkwNzEyMjIzODMxWjBAMQswCQYDVQQGEwJV
UzEaMBgGA1UEChMRQ2lzY28gU3lzdGVtcyBJbmMxFTATBgNVBAMTDENBUEYtN0Q3
RDBDMDCBnzANBgkqhkiG9w0BAQEFAAOBjQAwgYkCgYEA0hvMOZZ9ENYWme11YGY1
it2rvE3Nk/eqhnv8P9eqB1iqt+fFBeAG0WZ5bO5FetdU+BCmPnddvAeSpsfr3Z+h
x+r58fOEIBRHQLgnDZ+nwYH39uwXcRWWqWwlW147YHjV7M5c/R8T6daCx4B5NBo6
kdQdQNOrV3IP7kQaCShdM/kCAwEAAaMxMC8wDgYDVR0PAQH/BAQDAgKEMB0GA1Ud
JQQWMBQGCCsGAQUFBwMBBggrBgEFBQcDBTANBgkqhkiG9w0BAQUFAAOBgQCaNi6x
sL6M5NlDezpSBO3QmUVyXMfrONV2ysrSwcXzHu0gJ9MSJ8TwiQmVaJ47hSTlF5a8
YVYJ0IdifXbXRo+/EEO7kkmFE8MZta5rM7UWj8bAeR42iqA3RzQaDwuJgNWT9Fhh
GgfuNAlo5h1AikxsvxivmDlLdZyCMoqJJd7B2Q==
quit
Certificate has the following attributes:
Fingerprint MD5: 4B9636DF 0F3BA6B7 5F54BE72 24762DBC
Fingerprint SHA1: A9917775 F86BB37A 5C130ED2 3E528BB8 286E8C2D
% Do you accept this certificate? [yes/no]: y
Trustpoint CA certificate accepted.
% Certificate successfully imported
Router(config)# crypto pki trustpoint PEM
Router(ca-trustpoint)# revocation-check none
Router(ca-trustpoint)# enrollment terminal
Router(ca-trustpoint)# exit
Router(config)# crypto pki authenticate PEM
Enter the base 64 encoded CA certificate.
End with a blank line or the word "quit" on a line by itself
MIIDqDCCApCgAwIBAgIQdhL5YBU9b59OQiAgMrcjVjANBgkqhkiG9w0BAQUFADAu
MRYwFAYDVQQKEw1DaXNjbyBTeXN0ZW1zMRQwEgYDVQQDEwtDQVAtUlRQLTAwMTAe
Fw0wMzAyMDYyMzI3MTNaFw0yMzAyMDYyMzM2MzRaMC4xFjAUBgNVBAoTDUNpc2Nv
IFN5c3RlbXMxFDASBgNVBAMTC0NBUC1SVFAtMDAxMIIBIDANBgkqhkiG9w0BAQEF
AAOCAQ0AMIIBCAKCAQEArFW77Rjem4cJ/7yPLVCauDohwZZ/3qf0sJaWlLeAzBlq
Rj2lFlSij0ddkDtfEEo9VKmBOJsvx6xJlWJiuBwUMDhTRbsuJz+npkaGBXPOXJmN
Vd54qlpc/hQDfWlbrIFkCcYhHws7vwnPsLuy1Kw2L2cP0UXxYghSsx8H4vGqdPFQ
NnYy7aKJ43SvDFt4zn37n8jrvlRuz0x3mdbcBEdHbA825Yo7a8sk12tshMJ/YdMm
vny0pmDNZXmeHjqEgVO3UFUn6GVCO+K1y1dUU1qpYJNYtqLkqj7wgccGjsHdHr3a
U+bw1uLgSGsQnxMWeMaWo8+6hMxwlANPweufgZMaywIBA6OBwzCBwDALBgNVHQ8E
BAMCAYYwDwYDVR0TAQH/BAUwAwEB/zAdBgNVHQ4EFgQU6Rexgscfz6ypG270qSac
cK4FoJowbwYDVR0fBGgwZjBkoGKgYIYtaHR0cDovL2NhcC1ydHAtMDAxL0NlcnRF
bnJvbGwvQ0FQLVJUUC0wMDEuY3Jshi9maWxlOi8vXFxjYXAtcnRwLTAwMVxDZXJ0
RW5yb2xsXENBUC1SVFAtMDAxLmNybDAQBgkrBgEEAYI3FQEEAwIBADANBgkqhkiG
9w0BAQUFAAOCAQEAq2T96/YMMtw2Dw4QX+F1+g1XSrUCrNyjx7vtFaRDHyB+kobw
dwkpohfkzfTyYpJELzV1r+kMRoyuZ7oIqqccEroMDnnmeApc+BRGbDJqS1Zzk4OA
c6Ea7fm53nQRlcSPmUVLjDBzKYDNbnEjizptaIC5fgB/S9S6C1q0YpTZFn5tjUjy
WXzeYSXPrcxb0UH7IQJ1ogpONAAUKLoPaZU7tVDSH3hD4+VjmLyysaLUhksGFrrN
phzZrsVVilK17qpqCPllKLGAS4fSbkruq3r/6S/SpXS6/gAoljBKixP7ZW2PxgCU
1aU9cURLPO95NDOFN3jBk3Sips7cVidcogowPQ==
quit
Certificate has the following attributes:
Fingerprint MD5: 233C8E33 8632EA4E 76D79FEB FFB061C6
Fingerprint SHA1: F7B40B94 5831D2AB 447AB8F2 25990732 227631BE
% Do you accept this certificate? [yes/no]: y
Trustpoint CA certificate accepted.
% Certificate successfully imported
Use the show crypto pki trustpoint status command to show that enrollment has succeeded
and that five CA certificates were granted. The five certificates include the three
certificates just entered and the CA server certificate and the SRST router certificate.
Router# show crypto pki trustpoint status
```

```
Trustpoint 7970:
Issuing CA certificate configured:
Subject Name:
cn=CAP-RTP-002,o=Cisco Systems
Fingerprint MD5: F7E150EA 5E6E3AC5 615FC696 66415C9F
Fingerprint SHA1: 1BE2B503 DC72EE28 0C0F6B18 798236D8 D3B18BE6
State:
Keys generated ............ Yes (General Purpose)
Issuing CA authenticated ....... Yes
Certificate request(s) ..... None
Trustpoint 7960:
Issuing CA certificate configured:
Subject Name:
cn=CAPF-508A3754,o=Cisco Systems Inc,c=US
Fingerprint MD5: 6BAE18C2 0BCE391E DAE2FE4C 5810F576
Fingerprint SHA1: B7735A2E 3A5C274F C311D7F1 3BE89942 355102DE
State:
Keys generated ............ Yes (General Purpose)
Issuing CA authenticated ....... Yes
Certificate request(s) ..... None
Trustpoint PEM:
Issuing CA certificate configured:
Subject Name:
cn=CAP-RTP-001,o=Cisco Systems
Fingerprint MD5: 233C8E33 8632EA4E 76D79FEB FFB061C6
Fingerprint SHA1: F7B40B94 5831D2AB 447AB8F2 25990732 227631BE
State:
Keys generated ............ Yes (General Purpose)
Issuing CA authenticated ....... Yes
Certificate request(s) ..... None
Trustpoint srstcaserver:
Issuing CA certificate configured:
Subject Name:
cn=srstcaserver
Fingerprint MD5: 6AF5B084 79C93F2B 76CC8FE6 8781AF5E
Fingerprint SHA1: 47D30503 38FF1524 711448B4 9763FAF6 3A8E7DCF
State:
Keys generated ............ Yes (General Purpose)
Issuing CA authenticated ....... Yes
Certificate request(s) ..... None
Trustpoint srstca:
Issuing CA certificate configured:
Subject Name:
cn=srstcaserver
Fingerprint MD5: 6AF5B084 79C93F2B 76CC8FE6 8781AF5E
Fingerprint SHA1: 47D30503 38FF1524 711448B4 9763FAF6 3A8E7DCF
Router General Purpose certificate configured:
Subject Name:
serialNumber=F3246544+hostname=c2611XM-sSRST.cisco.com
Fingerprint: 35471295 1C907EC1 45B347BC 7A9C4B86
State:
Keys generated ............ Yes (General Purpose)
Issuing CA authenticated ....... Yes
Certificate request(s) ..... Yes
```

Cisco Unified Communications Manager 5.0 and Later Versions Example

The following example shows the configuration for the four certificates (CAPF, CiscoCA, CiscoManufactureCA, and CiscoRootCA2048) that are required for systems running Cisco Unified Communications Manager 5.0:

```
Router(config)# crypto pki trustpoint CAPF
Router(ca-trustpoint)# revocation-check none
Router(ca-trustpoint)# enrollment terminal
Router(ca-trustpoint)# exit
Router(config)# crypto pki authenticate CAPF

Enter the base 64 encoded CA certificate.
End with a blank line or the word "quit" on a line by itself
MIICKjCCAZOgAwIBAgIC8wEwDQYJKoZIhvcNAQEFBQAwQTELMAkGA1UEBhMCVVMx
GjAYBgNVBAoTEUNpc2NvIFN5c3RlbXMgSW5jMRYwFAYDVQQDEw1DQVBGLTU4RUFE
MkQyMB4XDTA2MDMwMTIxMjc1MloXDTIxMDIyNTIxMjc1MVowQTELMAkGA1UEBhMC
VVMxGjAYBgNVBAoTEUNpc2NvIFN5c3RlbXMgSW5jMRYwFAYDVQQDEw1DQVBGLTU4
RUFEMkQyMIGfMA0GCSqGSIb3DQEBAQUAA4GNADCBiQKBgQC99KgZT94qhozw4bOB
f8Z0tYwT2l4L++mC64O3s3AshDi8xe8Y8sN/f/ZKRRhNIxBlK4SWafXnHKJBqKZn
WtSgkRjJ3Dh0XtqcWYt8VS2sC69g8sX09lskKl3m+TpWsr2T/mDXv6CceaKN+mch
gcrrnNo8kamOOIG8OsQc4L6XzQIDAQABozEwLzAOBgNVHQ8BAf8EBAMCAoQwHQYD
quit
Certificate has the following attributes:
Fingerprint MD5: 1951DJ4E 76D79FEB FFB061C6 233C8E33
Fingerprint SHA1: 222891BE Z7B89B94 447AB8F2 5831D2AB 25990732
% Do you accept this certificate? [yes/no]: y
Trustpoint CA certificate accepted.
% Certificate successfully imported

Router(config)# crypto pki trustpoint CiscoCA
Router(ca-trustpoint)# revocation-check none
Router(ca-trustpoint)# enrollment terminal
Router(ca-trustpoint)# exit
Router(config)# crypto pki authenticate CiscoCA

Enter the base 64 encoded CA certificate.
End with a blank line or the word "quit" on a line by itself
MIIDqDCCApCgAwIBAgIQdhL5YBU9b59OQiAgMrcjVjANBgkqhkiG9w0BAQUFADAu
MRYwFAYDVQQKEw1DaXNjbyBTeXN0ZW1zMRQwEgYDVQQDEwtDQVAtUlRQLTAwMTAe
Vd54qlpc/hQDfWlbrIFkCcYhHws7vwnPsLuy1Kw2L2cP0UXxYghSsx8H4vGqdPFQ
NnYy7aKJ43SvDFt4zn37n8jrvlRuz0x3mdbcBEdHbA825Yo7a8sk12tshMJ/YdMm
vny0pmDNZXmeHjqEgVO3UFUn6GVCO+K1y1dUU1qpYJNYtqLkqj7wgccGjsHdHr3a
U+bw1uLgSGsQnxMWeMaWo8+6hMxwlANPweufgZMaywIBA6OBwzCBwDALBgNVHQ8E
c6Ea7fm53nQRlcSPmUVLjDBzKYDNbnEjizptaIC5fgB/S9S6C1q0YpTZFn5tjUjy
WXzeYSXPrcxb0UH7IQJ1ogpONAAUKLoPaZU7tVDSH3hD4+VjmLyysaLUhksGFrrN
phzZrsVVilK17qpqCPllKLGAS4fSbkruq3r/6S/SpXS6/gAoljBKixP7ZW2PxgCU
1aU9cURLPO95NDOFN3jBk3Sips7cVidcogowPQ==
quit
Certificate has the following attributes:
Fingerprint MD5: 21956CBR 4B9706DF 0F3BA6B7 7P54AZ72
Fingerprint SHA1: A9917775 F86BB37A 7H130ED2 3E528BB8 286E8C2D
% Do you accept this certificate? [yes/no]: y
Trustpoint CA certificate accepted.
% Certificate successfully imported

Router(config)# crypto pki trustpoint CiscoManufactureCA
Router(ca-trustpoint)# revocation-check none
Router(ca-trustpoint)# enrollment terminal
Router(ca-trustpoint)# exit
Router(config)# crypto pki authenticate CiscoManufactureCA

Enter the base 64 encoded CA certificate.
End with a blank line or the word "quit" on a line by itself
MIIE2TCCA8GgAwIBAgIKamlnswAAAAAAAzANBgkqhkiG9w0BAQUFADA1MRYwFAYD
D/g2qgfEMkHFp68dGf/2c5k5WnNnYhM0DR9elXBSZBcG7FNcXNtq6jUAQQIBA6OC
AecwggHjMBIGA1UdEwEB/wQIMAYBAf8CAQAwHQYDVR0OBBYEFNDFIiarT0Zg7K4F
kcfcWtGwR/dsMAsGA1UdDwQEAwIBhjAQBgkrBgEEAYI3FQEEAwIBADAZBgkrBgEE
AYI3FAIEDB4KAFMAdQBiAEMAQTAfBgNVHSMEGDAWgBQn88gVHm6aAgkWrSugiWBf
2nsvqjBDBgNVHR8EPDA6MDigNqA0hjJodHRwOi8vd3d3LmNpc2NvLmNvbS9zZWN1
cml0eS9wa2kvY3JsL2NyY2EyMDQ4LmNybDBQBggrBgEFBQcBAQREMEIwQAYIKwYB
BQUHMAKGNGh0dHA6Ly93d3cuY2lzY28uY29tL3NlY3VyaXR5L3BraS9jZXJ0cy9j
cmNhMjA0OC5jZXIwXAYDVR0gBFUwUzBRBgorBgEEAQkVAQIAMEMwQQYIKwYBBQUH
```

```
I+ii6itvaSN6go4cTAnPpE+rhC836WVg0ZrG2PML9d7QJwBcbx2RvdFOWFEdyeP3
OOfTC9Fovo4ipUsG4eakqjN9GnW6JvNwxmEApcN5JlunGdGTjaubEBEpH6GC/f08
S25l3JNFBemvM2tnIwcGhiLa69yHz1khQhrpz3B1iOAkPV19TpY4gJfVb/Cbcdi6
YBmlsGGGrd1lZva5J6LuL2GbuqEwYf2+rDUU+bgtlwavw+9tzD0865XpgdOKXrbO
+nmka9eiV2TEP0zJ2+iC7AFm1BCIolblPFft6QKoSJFjB6thJksaE5/k3Npf
quit
Certificate has the following attributes:
Fingerprint MD5: 0F3BA6B7 4B9636DF 5F54BE72 24762SBR
Fingerprint SHA1: L92BB37A S9919925 5C130ED2 3E528UP8 286E8C2D
% Do you accept this certificate? [yes/no]: y
Trustpoint CA certificate accepted.
% Certificate successfully imported

Router(config)# crypto pki trustpoint CiscoRootCA2048
Router(ca-trustpoint)# revocation-check none
Router(ca-trustpoint)# enrollment terminal
Router(ca-trustpoint)# exit
Router(config)# crypto pki authenticate CiscoRootCA2048

Enter the base 64 encoded CA certificate.
End with a blank line or the word "quit" on a line by itself
MIIDQzCCAiugAwIBAgIQX/h7KCtU3I1CoxW1aMmt/zANBgkqhkiG9w0BAQUFADA1
MRYwFAYDVQQKEw1DaXNjbyBTeXN0ZW1zMRswGQYDVQQDExJDaXNjbyBSb290IENB
IDIwNDgwHhcNMDQwNTE0MjAxNzEyWhcNMjkwNTE0MjAyNTQyWjA1MRYwFAYDVQQK
Ew1DaXNjbyBTeXN0ZW1zMRswGQYDVQQDExJDaXNjbyBSb290IENBIDIwNDgwggEg
MA0GCSqGSIb3DQEBAQUAA4IBDQAwggEIAoIBAQCwmrmrp68Kd6ficba0ZmKUeIhH
FR5umgIJFq0roIlgX9p7L6owEAYJKwYBBAGCNxUBBAMCAQAwDQYJKoZIhvcNAQEF
BQADggEBAJ2dhISjQal8dwy3U8pORFBi71R803UXHOjgxkhLtv5MOhmBVrBW7hmW
Yqpao2TB9k5UM8Z3/sUcuuVdJcr18JOagxEu5sv4dEX+5wW4q+ffy0vhN4TauYuX
cB7w4ovXsNgOnbFp1iqRe6lJT37mjpXYgyc81WhJDtSd9i7rp77rMKSsH0T8lasz
Bvt9YAretIpjsJyp8qS5UwGH0GikJ3+r/+n6yUA4iGe0OcaEb1fJU9u6ju7AQ7L4
CYNu/2bPPu8Xs1gYJQk0XuPL1hS27PKSb3TkL4Eq1ZKR4OCXPDJoBYVL0fdX4lId
kxpUnwVwwEpxYB5DC2Ae/qPOgRnhCzU=
quit
Certificate has the following attributes:
Fingerprint MD5: 2G3LZ6B7 2R1995ER 6KE4WE72 3E528BB8
Fingerprint SHA1: M9912245 5C130ED2 24762JBC 3E528VF8 956E8S5H
% Do you accept this certificate? [yes/no]: y
Trustpoint CA certificate accepted.
% Certificate successfully imported
```

Configuring Cisco Unified Communications Manager to the Secure Cisco Unified SRST Router

The following tasks are performed in Cisco Unified Communications Manager:

Adding an SRST Reference to Cisco Unified Communications Manager

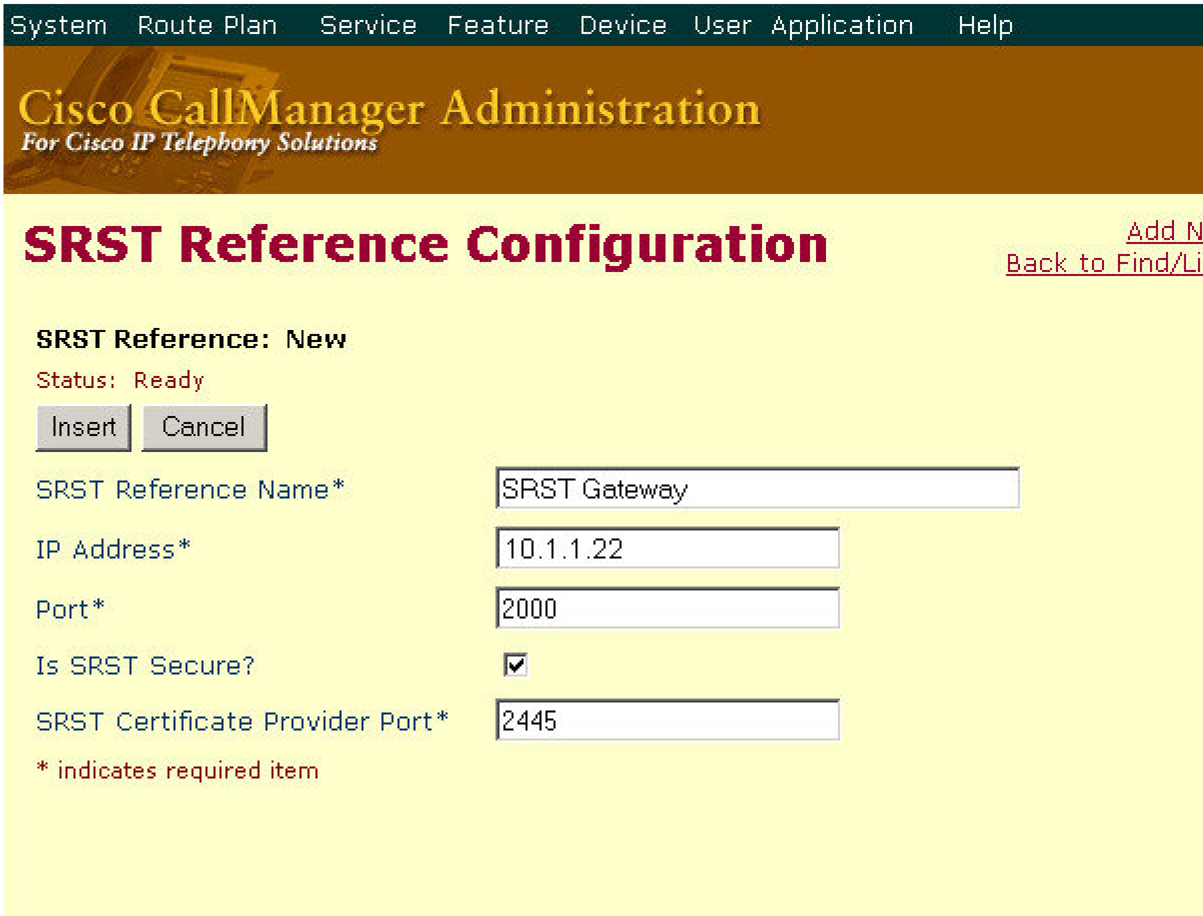
The following procedure describes how to add an SRST reference to Cisco Unified Communications Manager.

Before following this procedure, verify that credentials service is running in the Cisco Unified SRST Router. Cisco Unified Communications Manager connects to the Cisco Unified SRST Router for its device certificate. To enable credentials service, see the Enabling Credentials Service on the Secure Cisco Unified SRST Router section.

For complete information on adding Cisco Unified SRST to Cisco Unified Communications Manager, see the “Survivable Remote Site Telephony Configuration” section for the Cisco Unified Communications Manager version that you are running. All Cisco Unified CM administration guides are at the following URL: http://www.cisco.com/en/US/products/sw/voicesw/ps556/prod_maintenance_guides_list.html.

1. In the menu bar in Cisco Unified Communications Manager, choose **CCMAdmin** > **System** > **SRST** .
2. Click **Add New SRST Reference** .

3. Enter the appropriate settings. The following figure shows the available fields in the SRST Reference Configuration window.
   a. Enter the name of the SRST gateway, the IP address, and the port.
   b. Check the box asking if the SRST gateway is secure.
   c. Enter the certificate provider (credentials service) port number. Credentials service runs on default port 2445


System Route Plan Service Feature Device User Application Help
Cisco CallManager Administration
For Cisco IP Telephony Solutions
Add N
Back to Find/Li
Insert
Cancel

SRST Reference Configuration

SRST Reference: New

Status: Ready

SRST Reference Name*	SRST Gateway
IP Address*	10.1.1.22
Port*	2000
Is SRST Secure?	☑
SRST Certificate Provider Port*	2445

* indicates required item

4. To add the new SRST reference, click **Insert** . The message “Status: Insert completed” displays.
5. To add more SRST references, repeat Steps 2 to 4.

Configuring SRST Fallback on Cisco Unified Communications Manager

The following procedure describes how to configure SRST fallback on Cisco Unified Communications

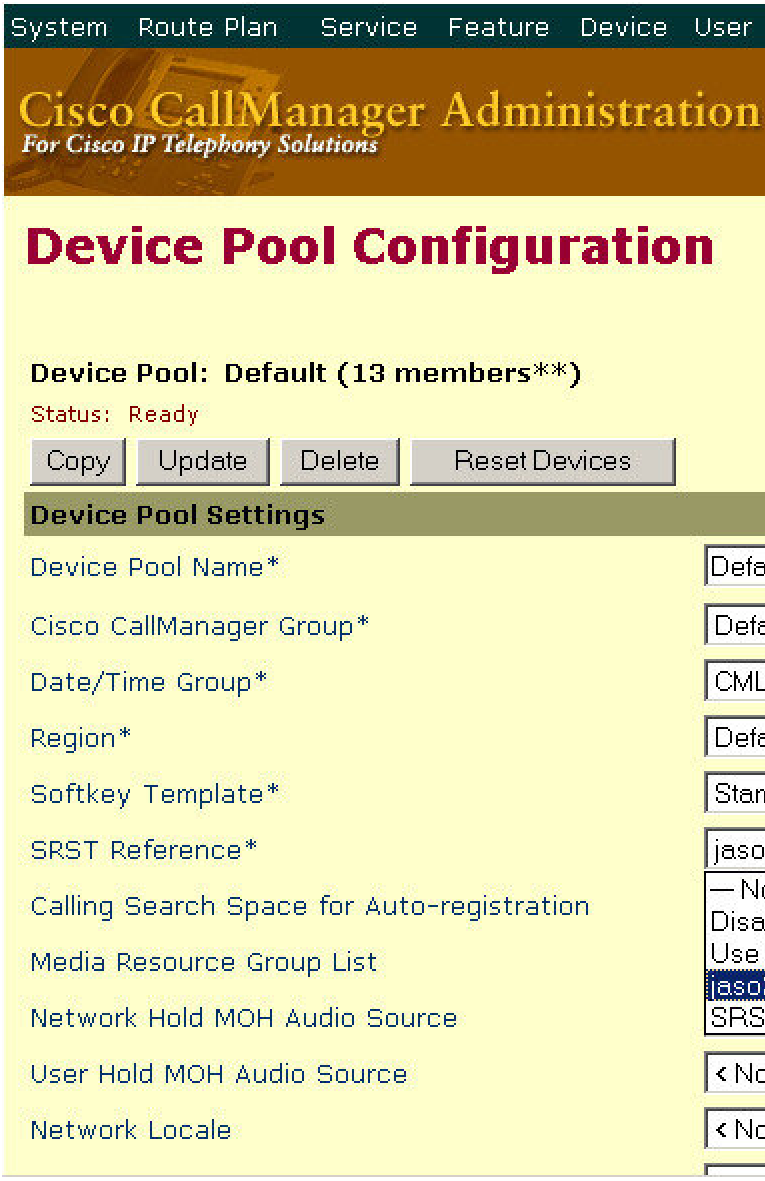
For complete information about adding a device pool to Cisco Unified Communications Manager, see the “Device Pool Configuration” section in Cisco Unified Communications Manager Administration Guide for the Cisco Unified Communications Manager version that you are running. All Cisco Unified CM administration guides are at the following URL: http://www.cisco.com/en/US/products/sw/voicesw/ps556/prod_maintenance_guides_list.html

SUMMARY STEPS

1. In the menu bar in Cisco Unified Communications Manager, choose **CCMAdmin** > **System** > **Device Pool** .
2. Use one of the following methods to add a device pool:
3. In the upper, right corner of the window, click the **Add New Device Pool** link. The Device Pool. Configuration window displays.
4. Enter the SRST reference.
5. Click **Update** to save the device pool information in the database.

DETAILED STEPS

	Command or Action	Purpose
Step 1	In the menu bar in Cisco Unified Communications Manager, choose **CCMAdmin** > **System** > **Device Pool** .	
Step 2	Use one of the following methods to add a device pool:	• If a device pool already exists with settings that are similar to the one that you want to add, choose the existing device pool to display its settings, click **Copy** , and modify the settings as needed. Continue with Step 4. • To add a device pool without copying an existing one, continue with Step 3.

	Command or Action	Purpose
Step 3	In the upper, right corner of the window, click the **Add New Device Pool** link. The Device Pool. Configuration window displays.	 System Route Plan Service Feature Device User Cisco CallManager Administration For Cisco IP Telephony Solutions Device Pool Configuration Device Pool: Default (13 members**) Status: Ready Copy Update Delete Reset Devices Device Pool Settings Device Pool Name* Defa Cisco CallManager Group* Defa Date/Time Group* CML Region* Defa Softkey Template* Stan SRST Reference* jaso — No Disa Use jaso SRS Calling Search Space for Auto-registration Media Resource Group List Network Hold MOH Audio Source User Hold MOH Audio Source < No Network Locale < No
Step 4	Enter the SRST reference.	
Step 5	Click **Update** to save the device pool information in the database.	

Configuring CAPF on Cisco Unified Communications Manager

The Certificate Authority Proxy Function (CAPF) process allows supported devices, such as Cisco Unified IP Phones to request LSC certificates from the CAPF service on Cisco Unified Communications Manager. The CAPF utility generates a key pair and certificate that are specific for CAPF, and the utility copies this certificate to all Cisco Unified Communications Manager servers in the cluster.

For complete instructions on configuring CAPF in Cisco Unified Communications Manager, see the Cisco IP Phone Authentication and Encryption for Cisco Communications Manager documentation.

Enabling SRST Mode on the Secure Cisco Unified SRST Router

To configure secure SRST on the router to support the Cisco Unified IP Phone functions, use the following commands beginning in global configuration mode.

SUMMARY STEPS

1. **call-manager-fallback**
2. **secondary-dialtone** *digit-string*
3. **transfer-system** {**blind** | **full-blind** |**full-consult** | **local-consult**}
4. **ip source-address** *ip-address* [**port***port*]
5. **max-ephones** *max-phones*
6. **max-dn** *max-directory-numbers*
7. **transfer-pattern** *transfer-pattern*
8. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**call-manager-fallback** **Example:** `Router(config)# call-manager-fallback`	Enters call-manager-fallback configuration mode.
Step 2	**secondary-dialtone** *digit-string* **Example:** `Router(config-cm-fallback)# secondary-dialtone 9`	Activates a secondary dial tone when a digit string is dialed.

Step 3	**transfer-system** {**blind** | **full-blind** |**full-consult** | **local-consult**}
	Example:
	`Router(config-cm-fallback)# transfer-system full-consult`
		Defines the call-transfer method for all lines served by the Cisco Unified SRST Router.
		• **blind** : Calls are transferred without consultation with a single phone line using the Cisco proprietary method.
		• **full-blind** : Calls are transferred without consultation using H.450.2 standard methods.
		• **full-consult** : Calls are transferred with consultation using a second phone line if available. The calls fallback to full-blind if the second line is unavailable.
		• **local-consult** : Calls are transferred with local consultation using a second phone line if available. The calls fallback to blind for nonlocal consultation or nonlocal transfer target.

Step 4	**ip source-address** *ip-address* [**port***port*] **Example:**	Enables the router to receive messages from the Cisco IP Phones through the specified IP addresses and provides for

	Command or Action	Purpose
	`Router(config-cm-fallback)# ip source-address 10.1.1.22 port 2000`	strict IP address verification. The default port number is 2000.
Step 5	**max-ephones** *max-phones* **Example:** `Router(config-cm-fallback)# max-ephones 15`	Configures the maximum number of Cisco IP phones that can be supported by the router. The maximum number is platform dependent. The default is 0. See the Supported Devices, Router Platforms and Memory Specifications, on page 55 section for further details.
Step 6	**max-dn** *max-directory-numbers* **Example:** `Router(config-cm-fallback)# max-dn 30`	Sets the maximum number of directory numbers (DNs) or virtual voice ports that can be supported by the router. • *max-directory-numbers* : Maximum number of directory numbers or virtual voice ports supported by the router. The maximum number is platform dependent. The default is 0. See the Supported Devices, Router Platforms and Memory Specifications, on page 55 section for further details.
Step 7	**transfer-pattern** *transfer-pattern* **Example:** `Router(config-cm-fallback)# transfer-pattern .....`	Allows transfer of phone calls by Cisco Unified IP Phones to specified phone number patterns. • *transfer-pattern*: String of digits for permitted call transfers. Wildcards are allowed.
Step 8	**exit** **Example:** `Router(config-cm-fallback)# exit`	Exits call-manager-fallback configuration mode.

Example

The following example enables SRST mode on your router:

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# secondary-dialtone 9
Router(config-cm-fallback)# transfer-system full-consult
Router(config-cm-fallback)# ip source-address 10.1.1.22 port 2000
Router(config-cm-fallback)# max-ephones 15
Router(config-cm-fallback)# max-dn 30
Router(config-cm-fallback)# transfer-pattern .....
Router(config-cm-fallback)# exit
```

Configuring Secure SCCP SRST

Prerequisites for Configuring Secure SCCP SRST

- Cisco Unified Communications Manager 4.1(2) or later must be installed and must support security mode (authenticate and encryption mode).
- Unified SRST 12.3 or later releases for Secure SCCP support on Cisco 4000 Series Integrated Services Routers and Cisco Analog Voice Gateways mentioned in the section Secure SCCP SRST for Analog

Voice Gateways. The configuration and behavior of Secure SCCP SRST fallback aligns with the existing support offered on Cisco Integrated Services Router Generation 2, unless specified otherwise.

Restrictions for Configuring Secure SCCP SRST

Not Supported in Secure SCCP SRST Mode (For Unified SRST 12.2 and prior releases)

- Cisco Unified Communications Manager versions before 4.1(2).
- Secure MOH; MOH stays active, but reverts to non-secure.
- Secure transcoding or conferencing.
- Secure H.323 or SIP trunks.
- SIP phones interoperability.
- Hot Standby Routing Protocol (HSRP).

Not Supported in Secure SCCP SRST Mode (For Unified SRST 12.3 and later releases)

For information on the restrictions for Secure SCCP SRST support introduced on Unified SRST 12.3, see the section SCCP SRST in Restrictions for Configuring Secure SRST.

Supported Calls in Secure SCCP SRST Mode (For Unified SRST 12.2 and prior releases)

Only voice calls are supported in secure SCCP SRST mode. Specifically, the following voice calls are supported:

- Basic call
- Call transfer (consult and blind)
- Call forward (busy, no-answer, all)
- Shared line (IP phones)
- Hold and resume

For information on the features supported on Unified SRST 12.3 and later releases, see Feature Support for Secure SRST (SCCP), Unified SRST Release 12.3.

Verifying Phone Status and Registrations

To verify or troubleshoot Cisco Unified IP Phone status and registration, complete the following steps beginning in privileged EXEC mode.

Note You can verify Phone Status and Registrations in secure SCCP SRST after you have performed the following steps:

- Enabling Credentials Service on the Secure Cisco Unified SRST Router
- Adding an SRST Reference to Cisco Unified Communications Manager
- Enabling SRST Mode on the Secure Cisco Unified SRST Router

SUMMARY STEPS

1. **show ephone**
2. **show ephone offhook**
3. **show voice call status**
4. **debug ephone register**
5. **debug ephone state**

DETAILED STEPS

<table>
<tr><th></th><th>Command or Action</th><th>Purpose</th></tr>
<tr><td><b>Step 1</b></td><td>

show ephone

Example:

```
Router# show ephone

ephone-1 Mac:1000.1111.0002 TCP socket:[5]
activeLine:0 REGISTERED in SCCP ver 5
+ Authentication + Encryption with TLS connection
mediaActive:0 offhook:0 ringing:0 reset:0
reset_sent:0 paging 0 debug:0
IP:10.1.1.40 32626 7970 keepalive 390 max_line 8
button 1: dn 14 number 2002 CM Fallback CH1 IDLE
ephone-2 Mac:1000.1111.000B TCP socket:[12]
activeLine:0 REGISTERED in SCCP ver
5 + Authentication + Encryption with TLS connection
mediaActive:0 offhook:0 ringing:0 reset:0
reset_sent:0 paging 0 debug:0
IP:10.1.1.40 32718 7970 keepalive 390 max_line 8
button 1: dn 21 number 2011 CM Fallback CH1 IDLE
ephone-3 Mac:1000.1111.000A TCP socket:[16]
activeLine:0 REGISTERED in SCCP ver
5 + Authentication + Encryption with TLS connection
mediaActive:0 offhook:0 ringing:0 reset:0
reset_sent:0 paging 0 debug:0
IP:10.1.1.40 32862 7970 keepalive 390 max_line 8
button 1: dn 2 number 2010 CM Fallback CH1 IDLE
```

</td><td>Use this command to display registered Cisco Unified IP Phones and their capabilities. The <b>show ephone</b> command also displays authentication and encryption status when used for secure SCCP SRST. In this example, authentication and encryption status is active with a TLS connection.</td></tr>
<tr><td><b>Step 2</b></td><td>

show ephone offhook

Example:

```
Router# show ephone offhook

ephone-1 Mac:1000.1111.0002 TCP socket:[5]
activeLine:1 REGISTERED in SCCP ver 5
+ Authentication + Encryption with TLS connection
mediaActive:1 offhook:1 ringing:0 reset:0
reset_sent:0 paging 0
:0
IP:10.1.1.40 32626 7970 keepalive 391 max_line 8
button 1: dn 14 number 2002 CM Fallback CH1
CONNECTED
Active Secure Call on DN 14 chan 1 :2002 10.1.1.40
29632 to 10.1.1.40 25616 via 10.1.1.40
G711Ulaw64k 160 bytes no vad
Tx Pkts 295 bytes 49468 Rx Pkts 277 bytes 46531
Lost
0
Jitter 0 Latency 0 callingDn 22 calledDn -1
ephone-2 Mac:1000.1111.000B TCP socket:[12]
```

</td><td>Use this command to display Cisco IP Phone status and quality for all phones that are off hook. In this example, authentication and encryption status is active with a TLS connection, and there is an active secure call.</td></tr>
</table>

	Command or Action	Purpose
	activeLine:1 REGISTERED in SCCP ver 5 + Authentication + Encryption with TLS connection mediaActive:1 offhook:1 ringing:0 reset:0 reset_sent:0 paging 0 debug:0 IP:10.1.1.40 32718 7970 keepalive 391 max_line 8 button 1: dn 21 number 2011 CM Fallback CH1 CONNECTED Active Secure Call on DN 21 chan 1 :2011 10.1.1.40 16382 to 10.1.1.40 16382 via 10.1.1.40 G711Ulaw64k 160 bytes no vad Tx Pkts 295 bytes 49468 Rx Pkts 277 bytes 46531 Lost 0 Jitter 0 Latency 0 callingDn -1 calledDn 11	
Step 3	**show voice call status** **Example:** CallID CID ccVdb Port DSP/Ch Called # Codec Dial-peers 0x1164 2BFE 0x8619A460 50/0/35.0 2014 g711ulaw 20035/20027 0x1165 2BFE 0x86144B78 50/0/27.0 *2014 g711ulaw 20027/20035 0x1166 2C01 0x861043D8 50/0/21.0 2012 g711ulaw 20021/20011 0x1168 2C01 0x860984C4 50/0/11.0 *2012 g711ulaw 20011/20021 0x1167 2C04 0x8610EC7C 50/0/22.0 2002 g711ulaw 20022/20014 0x1169 2C04 0x860B8894 50/0/14.0 *2002 g711ulaw 20014/20022 0x116A 2C07 0x860A374C 50/0/12.0 2010 g711ulaw 20012/20002 0x116B 2C07 0x86039700 50/0/2.0 *2010 g711ulaw 20002/20012 0x116C 2C0A 0x86119520 50/0/23.0 2034 g711ulaw 20023/20020 0x116D 2C0A 0x860F9150 50/0/20.0 *2034 g711ulaw 20020/20023 0x116E 2C0D 0x8608DC20 50/0/10.0 2022 g711ulaw 20010/20008 0x116F 2C0D 0x86078AD8 50/0/8.0 *2022 g711ulaw 20008/20010 0x1170 2C10 0x861398F0 50/0/26.0 2016 g711ulaw 20026/20028 0x1171 2C10 0x8614F41C 50/0/28.0 *2016 g711ulaw 20028/20026 0x1172 2C13 0x86159CC0 50/0/29.0 2018 g711ulaw 20029/20004 0x1173 2C13 0x8604E848 50/0/4.0 *2018 g711ulaw 20004/20029 0x1174 2C16 0x8612F04C 50/0/25.0 2026 g711ulaw 20025/20030 0x1175 2C16 0x86164F48 50/0/30.0 *2026 g711ulaw 20030/20025 0x1176 2C19 0x860D8C64 50/0/17.0 2032 g711ulaw 20017/20018 0x1177 2C19 0x860E4008 50/0/18.0 *2032 g711ulaw 20018/20017 0x1178 2C1C 0x860CE3C0 50/0/16.0 2004 g711ulaw 20016/20019	Use this command to show the call status for all voice ports on the Cisco Unified SRST router. This command is not applicable for calls between two POTS dial peers.

	Command or Action	Purpose
	0x1179 2C1C 0x860EE8AC 50/0/19.0 *2004 g711ulaw 20019/20016 0x117A 2C1F 0x86043FA4 50/0/3.0 2008 g711ulaw 20003/20024 0x117B 2C1F 0x861247A8 50/0/24.0 *2008 g711ulaw 20024/20003 0x117C 2C22 0x8608337C 50/0/9.0 2020 g711ulaw 20009/20031 0x117D 2C22 0x8616F7EC 50/0/31.0 *2020 g711ulaw 20031/20009 0x117E 2C25 0x86063990 50/0/6.0 2006 g711ulaw 20006/20001 0x117F 2C25 0x85C6BE6C 50/0/1.0 *2006 g711ulaw 20001/20006 0x1180 2C28 0x860ADFF0 50/0/13.0 2029 g711ulaw 20013/20034 0x1181 2C28 0x8618FBBC 50/0/34.0 *2029 g711ulaw 20034/20013 0x1182 2C2B 0x860C3B1C 50/0/15.0 2036 g711ulaw 20015/20005 0x1183 2C2B 0x860590EC 50/0/5.0 *2036 g711ulaw 20005/20015 0x1184 2C2E 0x8617A090 50/0/32.0 2024 g711ulaw 20032/20007 0x1185 2C2E 0x8606E234 50/0/7.0 *2024 g711ulaw 20007/20032 0x1186 2C31 0x861A56E8 50/0/36.0 2030 g711ulaw 20036/20033 0x1187 2C31 0x86185318 50/0/33.0 *2030 g711ulaw 20033/20036 18 active calls found	
Step 4	**debug ephone register** **Example:** Router# debug ephone register EPHONE registration debugging is enabled *Jun 29 09:16:02.180: New Skinny socket accepted [2] (0 active) *Jun 29 09:16:02.180: sin_family 2, sin_port 51617, in_addr 10.5.43.177 *Jun 29 09:16:02.180: skinny_socket_process: secure skinny sessions = 1 *Jun 29 09:16:02.180: add_skinny_secure_socket: pid =155, new_sock=0, ip address = 10.5.43.177 *Jun 29 09:16:02.180: skinny_secure_handshake: pid =155, sock=0, args->pid=155, ip address = 10.5.43.177 *Jun 29 09:16:02.184: Start TLS Handshake 0 10.5.43.177 51617 *Jun 29 09:16:02.184: TLS Handshake retcode OPSSLReadWouldBlockErr *Jun 29 09:16:03.188: TLS Handshake retcode OPSSLReadWouldBlockErr *Jun 29 09:16:04.188: TLS Handshake retcode OPSSLReadWouldBlockErr *Jun 29 09:16:05.188: TLS Handshake retcode OPSSLReadWouldBlockErr *Jun 29 09:16:06.188: TLS Handshake retcode OPSSLReadWouldBlockErr	Use this command to debug the process of Cisco IP phone registration.

Command or Action | **Purpose**

```
*Jun 29 09:16:07.188: TLS Handshake retcode
OPSSLReadWouldBlockErr
*Jun 29 09:16:08.188: CRYPTO_PKI_OPSSL - Verifying
 1
Certs
*Jun 29 09:16:08.212: TLS Handshake completes
```

Step 5

debug ephone state

Example:

```
Router# debug ephone state
*Jan 11 18:33:09.231:%SYS-5-CONFIG_I:Configured
from
console by console
*Jan 11 18:33:11.747:ephone-2[2]:OFFHOOK
*Jan 11
18:33:11.747:ephone-2[2]:---SkinnySyncPhoneDnOverlay
s is onhook
*Jan 11 18:33:11.747:ephone-2[2]:SIEZE on
activeLine
0 activeChan 1
*Jan 11 18:33:11.747:ephone-2[2]:SetCallState line
 1
DN 2(-1) chan 1 ref 6 TsOffHook
*Jan 11 18:33:11.747:ephone-2[2]:Check Plar Number
*Jan 11 18:33:11.751:DN 2 chan 1 Voice_Mode
*Jan 11 18:33:11.751:dn_tone_control DN=2 chan 1
tonetype=33:DtInsideDialTone onoff=1 pid=232
*Jan 11 18:33:15.031:dn_tone_control DN=2 chan 1
tonetype=0:DtSilence onoff=0 pid=232
*Jan 11 18:33:16.039:ephone-2[2]:Skinny-to-Skinny
call DN 2 chan 1 to DN 4 chan 1 instance 1
*Jan 11 18:33:16.039:ephone-2[2]:SetCallState line
 1
DN 2(-1) chan 1 ref 6 TsProceed
*Jan 11 18:33:16.039:ephone-2[2]:SetCallState line
 1
DN 2(-1) chan 1 ref 6 TsRingOut
*Jan 11 18:33:16.039:ephone-2[2]::callingNumber
6000
*Jan 11 18:33:16.039:ephone-2[2]::callingParty 6000
*Jan 11 18:33:16.039:ephone-2[2]:Call Info DN 2
line
1 ref 6 call state 1 called 6001 calling 6000
origcalled
*Jan 11 18:33:16.039:ephone-2[2]:Call Info DN 2
line
1 ref 6 called 6001 calling 6000 origcalled 6001
calltype 2
*Jan 11 18:33:16.039:ephone-2[2]:Call Info for chan
1
*Jan 11 18:33:16.039:ephone-2[2]:Original Called
Name 6001
*Jan 11 18:33:16.039:ephone-2[2]:6000 calling
*Jan 11 18:33:16.039:ephone-2[2]:6001
*Jan 11 18:33:16.047:ephone-3[3]:SetCallState line
 1
DN 4(4) chan 1 ref 7 TsRingIn
*Jan 11 18:33:16.047:ephone-3[3]::callingNumber
6000
*Jan 11 18:33:16.047:ephone-3[3]::callingParty 6000
*Jan 11 18:33:16.047:ephone-3[3]:Call Info DN 4
```

Use this command to review call setup between two secure Cisco Unified IP Phones. The debug ephone state trace shows the generation and distribution of encryption and decryption keys between the two phones.

	Command or Action	Purpose

```
line
1 ref 7 call state 7 called 6001 calling 6000
origcalled
*Jan 11 18:33:16.047:ephone-3[3]:Call Info DN 4
line
1 ref 7 called 6001 calling 6000 origcalled 6001
calltype 1
*Jan 11 18:33:16.047:ephone-3[3]:Call Info for chan
1
*Jan 11 18:33:16.047:ephone-3[3]:Original Called
Name 6001
*Jan 11 18:33:16.047:ephone-3[3]:6000 calling
*Jan 11 18:33:16.047:ephone-3[3]:6001
*Jan 11 18:33:16.047:ephone-3[3]:Ringer Inside Ring
On
*Jan 11 18:33:16.051:dn_tone_control DN=2 chan 1
tonetype=36:DtAlertingTone onoff=1 pid=232
*Jan 11 18:33:20.831:ephone-3[3]:OFFHOOK
*Jan 11
18:33:20.831:ephone-3[3]:---SkinnySyncPhoneDnOverlay
s is onhook
*Jan 11 18:33:20.831:ephone-3[3]:Ringer Off
*Jan 11 18:33:20.831:ephone-3[3]:ANSWER call
*Jan 11 18:33:20.831:ephone-3[3]:SetCallState line
 1
DN 4(-1) chan 1 ref 7 TsOffHook
*Jan 11
18:33:20.831:ephone-3[3][SEP000DEDAB3EBF]:Answer
Incoming call from ephone-(2) DN 2 chan 1
*Jan 11 18:33:20.831:ephone-3[3]:SetCallState line
 1
DN 4(-1) chan 1 ref 7 TsConnected
*Jan 11 18:33:20.831:defer_start for DN 2 chan 1
at
CONNECTED
*Jan 11 18:33:20.831:ephone-2[2]:SetCallState line
 1
DN 2(-1) chan 1 ref 6 TsConnected
*Jan 11 18:33:20.835:ephone-3[3]::callingNumber
6000
*Jan 11 18:33:20.835:ephone-3[3]::callingParty 6000
*Jan 11 18:33:20.835:ephone-3[3]:Call Info DN 4
line
1 ref 7 call state 4 called 6001 calling 6000
origcalled
*Jan 11 18:33:20.835:ephone-3[3]:Call Info DN 4
line
1 ref 7 called 6001 calling 6000 origcalled 6001
calltype 1
*Jan 11 18:33:20.835:ephone-3[3]:Call Info for chan
1
*Jan 11 18:33:20.835:ephone-3[3]:Original Called
Name 6001
*Jan 11 18:33:20.835:ephone-3[3]:6000 calling
*Jan 11 18:33:20.835:ephone-3[3]:6001
*Jan 11 18:33:20.835:ephone-2[2]:Security Key
Generation
! Ephone 2 generates a security key.
*Jan 11 18:33:20.835:ephone-2[2]:OpenReceive DN 2
chan 1 codec 4:G711Ulaw64k duration 20 ms bytes
160
*Jan 11 18:33:20.835:ephone-2[2]:Send Decryption
```

	Command or Action	Purpose
	Key ! Ephone 2 sends the decryption key. *Jan 11 18:33:20.835:ephone-3[3]:Security Key Generation !Ephone 3 generates its security key. *Jan 11 18:33:20.835:ephone-3[3]:OpenReceive DN 4 chan 1 codec 4:G711Ulaw64k duration 20 ms bytes 160 *Jan 11 18:33:20.835:ephone-3[3]:Send Decryption Key ! Ephone 3 sends its decryption key. *Jan 11 18:33:21.087:dn_tone_control DN=2 chan 1 tonetype=0:DtSilence onoff=0 pid=232 *Jan 11 18:33:21.087:DN 4 chan 1 Voice_Mode *Jan 11 18:33:21.091:DN 2 chan 1 End Voice_Mode *Jan 11 18:33:21.091:DN 2 chan 1 Voice_Mode *Jan 11 18:33:21.095:ephone-2[2]:OpenReceiveChannelAck:IP 1.1.1.8, port=25552, dn_index=2, dn=2, chan=1 *Jan 11 18:33:21.095:ephone-3[3]:StartMedia 1.1.1.8 port=25552 *Jan 11 18:33:21.095:DN 2 chan 1 codec 4:G711Ulaw64k duration 20 ms bytes 160 *Jan 11 18:33:21.095:ephone-3[3]:Send Encryption Key ! Ephone 3 sends its encryption key. *Jan 11 18:33:21.347:ephone-3[3]:OpenReceiveChannelAck:IP 1.1.1.9, port=17520, dn_index=4, dn=4, chan=1 *Jan 11 18:33:21.347:ephone-2[2]:StartMedia 1.1.1.9 port=17520 *Jan 11 18:33:21.347:DN 2 chan 1 codec 4:G711Ulaw64k duration 20 ms bytes 160 *Jan 11 18:33:21.347:ephone-2[2]:Send Encryption Key !Ephone 2 sends its encryption key.*Jan 11 18:33:21.851:ephone-2[2]::callingNumber 6000 *Jan 11 18:33:21.851:ephone-2[2]::callingParty 6000 *Jan 11 18:33:21.851:ephone-2[2]:Call Info DN 2 line 1 ref 6 call state 4 called 6001 calling 6000 origcalled *Jan 11 18:33:21.851:ephone-2[2]:Call Info DN 2 line 1 ref 6 called 6001 calling 6000 origcalled 6001 calltype 2 *Jan 11 18:33:21.851:ephone-2[2]:Call Info for chan 1 *Jan 11 18:33:21.851:ephone-2[2]:Original Called Name 6001 *Jan 11 18:33:21.851:ephone-2[2]:6000 calling *Jan 11 18:33:21.851:ephone-2[2]:6001	

Configuration Examples for Secure SCCP SRST

This section provides the following configuration examples:

Note IP addresses and hostnames in examples are fictitious.

Secure SCCP SRST: Example

This section provides a configuration example to match the identified configuration tasks in the previous sections. This example does not include using a third-party CA; it assumes the use of the Cisco IOS certificate server to generate your certificates.

```
Router# show running-config
.
.
.
! Define Unified Communications Manager.
ccm-manager fallback-mgcp
ccm-manager mgcp
ccm-manager music-on-hold
ccm-manager config server 10.1.1.13
ccm-manager config
!
! Define root CA.
crypto pki server srstcaserver
database level complete
database url nvram
issuer-name CN=srstcaserver
!
crypto pki trustpoint srstca
enrollment url http://10.1.1.22:80
revocation-check none
!
crypto pki trustpoint srstcaserver
revocation-check none
rsakeypair srstcaserver
!
! Define CTL/7970 trustpoint.
crypto pki trustpoint 7970
enrollment terminal
revocation-check none
!
crypto pki trustpoint PEM
enrollment terminal
revocation-check none
!
! Define CAPF/7960 trustpoint.
crypto pki trustpoint 7960
enrollment terminal
revocation-check none
!
! SRST router device certificate.
crypto pki certificate chain srstca
certificate 02
308201AD 30820116 A0030201 02020102 300D0609 2A864886 F70D0101 04050030
17311530 13060355 0403130C 73727374 63617365 72766572 301E170D 30343034
31323139 35323233 5A170D30 35303431 32313935 3232335A 30343132 300F0603
55040513 08443042 39453739 43301F06 092A8648 86F70D01 09021612 6A61736F
32363931 2E636973 636F2E63 6F6D305C 300D0609 2A864886 F70D0101 01050003
4B003048 024100D7 0CC354FB 5F7C1AE7 7A25C3F2 056E0485 22896D36 6CA70C19
C98F9BAE AE9D1F9B D4BB7A67 F3251174 193BB1A3 12946123 E5C1CCD7 A23E6155
FA2ED743 3FB8B902 03010001 A330302E 300B0603 551D0F04 04030205 A0301F06
03551D23 04183016 8014F829 CE97AD60 18D05467 FC293963 C2470691 F9BD300D
06092A86 4886F70D 01010405 00038181 007EB48E CAE9E1B3 D1E7A185 D7F0D565
```

```
CB84B17B 1151BD78 B3E39763 59EC650E 49371F6D 99CBD267 EB8ADF9D 9E43A5F2
FB2B18A0 34AF6564 11239473 41478AFC A86E6DA1 AC518E0B 8657CEBB ED2BDE8E
B586FE67 00C358D4 EFDD8D44 3F423141 C2D331D3 1EE43B6E 6CB29EE7 0B8C2752
C3AF4A66 BD007348 D013000A EA3C206D CF
quit
certificate ca 01
30820207 30820170 A0030201 02020101 300D0609 2A864886 F70D0101 04050030
17311530 13060355 0403130C 73727374 63617365 72766572 301E170D 30343034
31323139 34353136 5A170D30 37303431 32313934 3531365A 30173115 30130603
55040313 0C737273 74636173 65727665 7230819F 300D0609 2A864886 F70D0101
01050003 818D0030 81890281 8100C3AF EE1E4BB1 9922A8DA 2BB9DC8E 5B1BD332
1051C9FE 32A971B3 3C336635 74691954 98E765B1 059E24B6 32154E99 105CA989
9619993F CC72C525 7357EBAC E6335A32 2AAF9391 99325BFD 9B8355EB C10F8963
9D8FC222 EE8AC831 71ACD3A7 4E918A8F D5775159 76FBF499 5AD0849D CAA41417
DD866902 21E5DD03 C37D4B28 0FAB0203 010001A3 63306130 0F060355 1D130101
FF040530 030101FF 300E0603 551D0F01 01FF0404 03020186 301D0603 551D0E04
160414F8 29CE97AD 6018D054 67FC2939 63C24706 91F9BD30 1F060355 1D230418
30168014 F829CE97 AD6018D0 5467FC29 3963C247 0691F9BD 300D0609 2A864886
F70D0101 04050003 8181007A F71B25F9 73D74552 25DFD03A D8D1338F 6792C805
47A81019 795B5AAE 035400BB F859DABF 21892B5B E71A8283 08950414 8633A8B2
C98565A6 C09CA641 88661402 ACC424FD 36F23360 ABFF4C55 BB23C66A C80A3A57
5EE85FF8 C1B1A540 E818CE6D 58131726 BB060974 4E1A2F4B E6195522 122457F3
DEDBAAD7 3780136E B112A6
quit
crypto pki certificate chain srstcaserver
certificate ca 01
30820207 30820170 A0030201 02020101 300D0609 2A864886 F70D0101 04050030
17311530 13060355 0403130C 73727374 63617365 72766572 301E170D 30343034
31323139 34353136 5A170D30 37303431 32313934 3531365A 30173115 30130603
55040313 0C737273 74636173 65727665 7230819F 300D0609 2A864886 F70D0101
01050003 818D0030 81890281 8100C3AF EE1E4BB1 9922A8DA 2BB9DC8E 5B1BD332
1051C9FE 32A971B3 3C336635 74691954 98E765B1 059E24B6 32154E99 105CA989
9619993F CC72C525 7357EBAC E6335A32 2AAF9391 99325BFD 9B8355EB C10F8963
9D8FC222 EE8AC831 71ACD3A7 4E918A8F D5775159 76FBF499 5AD0849D CAA41417
DD866902 21E5DD03 C37D4B28 0FAB0203 010001A3 63306130 0F060355 1D130101
FF040530 030101FF 300E0603 551D0F01 01FF0404 03020186 301D0603 551D0E04
160414F8 29CE97AD 6018D054 67FC2939 63C24706 91F9BD30 1F060355 1D230418
30168014 F829CE97 AD6018D0 5467FC29 3963C247 0691F9BD 300D0609 2A864886
F70D0101 04050003 8181007A F71B25F9 73D74552 25DFD03A D8D1338F 6792C805
47A81019 795B5AAE 035400BB F859DABF 21892B5B E71A8283 08950414 8633A8B2
C98565A6 C09CA641 88661402 ACC424FD 36F23360 ABFF4C55 BB23C66A C80A3A57
5EE85FF8 C1B1A540 E818CE6D 58131726 BB060974 4E1A2F4B E6195522 122457F3
DEDBAAD7 3780136E B112A6
quit
crypto pki certificate chain 7970
certificate ca 353FB24BD70F14A346C1F3A9AC725675
308203A8 30820290 A0030201 02021035 3FB24BD7 0F14A346 C1F3A9AC 72567530
0D06092A 864886F7 0D010105 0500302E 31163014 06035504 0A130D43 6973636F
20537973 74656D73 31143012 06035504 03130B43 41502D52 54502D30 3032301E
170D3033 31303130 32303138 34395A17 0D323331 30313032 30323733 375A302E
31163014 06035504 0A130D43 6973636F 20537973 74656D73 31143012 06035504
03130B43 41502D52 54502D30 30323082 0120300D 06092A86 4886F70D 01010105
00038201 0D003082 01080282 010100C4 266504AD 7DC3FD8D 65556FA6 308FAE95
B570263B 575ABD96 1CC8F394 5965D9D0 D8CE02B9 F808CCD6 B7CD8C46 24801878
57DC4440 A7301DDF E40FB1EF 136212EC C4F3B50F BCAFBB4B CD2E5826 34521B65
01555FE4 D4206776 03368357 83932638 D6FC953F 3A179E44 67255A73 45C69DEE
FB4D221B 21D7A3AD 38184171 8FD8C271 42183E65 09461434 736C77CC F380EEBF
632C7B3F A5F92AA6 A8EF3490 8724A84F 4DAF7FD7 0928F585 764D3558 3C0FE9AF
1ED8763F A299A802 970004AD 1912D265 7DE335B4 BCB6F789 DC68B9FA C8FDF85E
8A28AD8F 0F4883C0 77112A47 141DBEE0 948FBE53 FE67B308 D40C8029 87BD790E
CDAB9FD7 A190C1A2 A462C5F2 4A6E0B02 0103A381 C33081C0 300B0603 551D0F04
04030201 86300F06 03551D13 0101FF04 05300301 01FF301D 0603551D 0E041604
1452922B E288EE2E 098A4E7E 702C56A5 9AB4D49B 96306F06 03551D1F 04683066
3064A062 A060862D 68747470 3A2F2F63 61702D72 74702D30 30322F43 65727445
```

```
6E726F6C 6C2F4341 502D5254 502D3030 322E6372 6C862F66 696C653A 2F2F5C5C
6361702D 7274702D 3030325C 43657274 456E726F 6C6C5C43 41502D52 54502D30
30322E63 726C3010 06092B06 01040182 37150104 03020100 300D0609 2A864886
F70D0101 05050003 82010100 56838CEF C4DA3AD1 EA8FBB15 2FFE6EE5 50A1972B
D4D7AF1F D298892C D5A2A76B C3462866 13E0E55D DC0C4B92 5AA94B6E 69277F9B
FC73C697 11266E19 451C0FAB A55E6A28 901A48C5 B9911EE6 348A8920 0AEDE1E0
B6EA781C FFD97CA4 B03C0E34 0E5B0649 8B0A34C9 B73A654E 09050C1F 4DA53E44
BF78443D B08C3A41 2EEEB873 78CB8089 34F9D16E 91512F0D 3A8674AD 0991ED1A
92841E76 36D7740E CB787F11 685B9E9D 0C67E85D AF6D05BA 3488E86D 7E2F7F65
6918DE0F BD3C7F67 D8A33F70 9C4A596E D9F62B3B 1EDEE854 D5882AD4 3D71F72B
8FAB7F3C 0B5F0759 D9828F83 954D7BB1 57A638EC 7D72BFF1 8933C16F 760BCA94
4C5B1931 67947A4F 89A1BDB5
quit
crypto pki certificate chain PEM
certificate ca 7612F960153D6F9F4E42202032B72356
308203A8 30820290 A0030201 02021076 12F96015 3D6F9F4E 42202032 B7235630
0D06092A 864886F7 0D010105 0500302E 31163014 06035504 0A130D43 6973636F
20537973 74656D73 31143012 06035504 03130B43 41502D52 54502D30 3031301E
170D3033 30323036 32333237 31335A17 0D323330 32303632 33333633 345A302E
31163014 06035504 0A130D43 6973636F 20537973 74656D73 31143012 06035504
03130B43 41502D52 54502D30 30313082 0120300D 06092A86 4886F70D 01010105
00038201 0D003082 01080282 010100AC 55BBED18 DE9B8709 FFBC8F2D 509AB83A
21C1967F DEA7F4B0 969694B7 80CC196A 463DA516 54A28F47 5D903B5F 104A3D54
A981389B 2FC7AC49 956262B8 1C143038 5345BB2E 273FA7A6 46860573 CE5C998D
55DE78AA 5A5CFE14 037D695B AC816409 C6211F0B 3BBF09CF B0BBB2D4 AC362F67
0FD145F1 620852B3 1F07E2F1 AA74F150 367632ED A289E374 AF0C5B78 CE7DFB9F
C8EBBE54 6ECF4C77 99D6DC04 47476C0F 36E58A3B 6BCB24D7 6B6C84C2 7F61D326
BE7CB4A6 60CD6579 9E1E3A84 8153B750 5527E865 423BE2B5 CB575453 5AA96093
58B6A2E4 AA3EF081 C7068EC1 DD1EBDDA 53E6F0D6 E2E0486B 109F1316 78C696A3
CFBA84CC 7094034F C1EB9F81 931ACB02 0103A381 C33081C0 300B0603 551D0F04
04030201 86300F06 03551D13 0101FF04 05300301 01FF301D 0603551D 0E041604
14E917B1 82C71FCF ACA91B6E F4A9269C 70AE05A0 9A306F06 03551D1F 04683066
3064A062 A060862D 68747470 3A2F2F63 61702D72 74702D30 30312F43 65727445
6E726F6C 6C2F4341 502D5254 502D3030 312E6372 6C862F66 696C653A 2F2F5C5C
6361702D 7274702D 3030315C 43657274 456E726F 6C6C5C43 41502D52 54502D30
30312E63 726C3010 06092B06 01040182 37150104 03020100 300D0609 2A864886
F70D0101 05050003 82010100 AB64FDEB F60C32DC 360F0E10 5FE175FA 0D574AB5
02ACDCA3 C7BBED15 A4431F20 7E9286F0 770929A2 17E4CDF4 F2629244 2F3575AF
E90C468C AE67BA08 AAA71C12 BA0C0E79 E6780A5C F814466C 326A4B56 73938380
73A11AED F9B9DE74 1195C48F 99454B8C 30732980 CD6E7123 8B3A6D68 80B97E00
7F4BD4BA 0B5AB462 94D9167E 6D8D48F2 597CDE61 25CFADCC 5BD141FB 210275A2
0A4E3400 1428BA0F 69953BB5 50D21F78 43E3E563 98BCB2B1 A2D4864B 0616BACD
A61CD9AE C5558A52 B5EEAA6A 08F96528 B1804B87 D26E4AEE AB7AFFE9 2FD2A574
BAFE0028 96304A8B 13FB656D 8FC60094 D5A53D71 444B3CEF 79343385 3778C193
74A2A6CE DC56275C A20A303D
quit
crypto pki certificate chain 7960
certificate ca F301
308201F7 30820160 A0030201 020202F3 01300D06 092A8648 86F70D01 01050500
3041310B 30090603 55040613 02555331 1A301806 0355040A 13114369 73636F20
53797374 656D7320 496E6331 16301406 03550403 130D4341 50462D33 35453038
33333230 1E170D30 34303430 39323035 3530325A 170D3139 30343036 32303535
30315A30 41310B30 09060355 04061302 5553311A 30180603 55040A13 11436973
636F2053 79737465 6D732049 6E633116 30140603 55040313 0D434150 462D3335
45303833 33323081 9F300D06 092A8648 86F70D01 01010500 03818D00 30818902
818100C8 BD9B6035 366B44E8 0F693A47 250FF865 D76C35F7 89B1C4FD 1D122CE0
F5E5CDFF A4A87EFF 41AD936F E5C93163 3E55D11A AF82A5F6 D563E21C EB89EBFA
F5271423 C3E875DC E0E07967 6E1AAB4F D3823E12 53547480 23BA1A09 295179B6
85A0E83A 77DD0633 B9710A88 0890CD4D DB55ADD0 964369BA 489043BB B667E60F
93954B02 03010001 300D0609 2A864886 F70D0101 05050003 81810056 60FD3AB3
6F98D2AD 40C309E2 C05B841C 5189271F 01D864E8 98BCE665 2AFBCC8C 54007A84
8F772C67 E3047A6C C62F6508 B36A6174 B68C1D78 C2228FEA A89ECEFB CC8BA9FC
0F30E151 431670F9 918514D9 868D1235 18137F1E 50DFD32E 1DC29CB7 95EF4096
421AF22F 5C1D5804 B83F8E8E 95B04F45 86563BFE DF976C5B FB490A
```

```
quit
!
!
no crypto isakmp enable
!
! Enable IPSec.
crypto isakmp policy 1
authentication pre-share
lifetime 28800
crypto isakmp key cisco123 address 10.1.1.13
! The crypto key should match the key configured on Cisco Unified Communications Manager.
!
! The crypto IPSec configuration should match your Cisco Unified Communications Manager
configuration.
crypto ipsec transform-set rtpset esp-des esp-md5-hmac
!
!
crypto map rtp 1 ipsec-isakmp
set peer 10.1.1.13
set transform-set rtpset
match address 116
!
!
interface FastEthernet0/0
ip address 10.1.1.22 255.255.255.0
duplex auto
speed auto
crypto map rtp
!
interface FastEthernet0/1
no ip address
shutdown
duplex auto
speed auto
!
ip classless
!
ip http server
no ip http secure-server
!
!
! Define traffic to be encrypted by IPSec.
access-list 116 permit ip host 10.1.1.22 host 10.1.1.13
!
!
control-plane
!
!
call application alternate DEFAULT
!
!
voice-port 1/0/0
!
voice-port 1/0/1
!
voice-port 1/0/2
!
voice-port 1/0/3
!
voice-port 1/1/0
timing hookflash-out 50
!
voice-port 1/1/1
!
```

```
voice-port 1/1/2
!
voice-port 1/1/3
!
! Enable MGCP voice protocol.
mgcp
mgcp call-agent 10.1.1.13 2427 service-type mgcp version 0.1
mgcp dtmf-relay voip codec all mode out-of-band
mgcp rtp unreachable timeout 1000 action notify
mgcp package-capability rtp-package
mgcp package-capability sst-package
no mgcp package-capability fxr-package
no mgcp timer receive-rtcp
mgcp sdp simple
mgcp fax t38 inhibit
mgcp rtp payload-type g726r16 static
!
mgcp profile default
!
!
dial-peer voice 81235 pots
application mgcpapp
destination-pattern 81235
port 1/1/0
forward-digits all
!
dial-peer voice 81234 pots
application mgcpapp
destination-pattern 81234
port 1/0/0
!
dial-peer voice 999100 pots
application mgcpapp
port 1/0/0
!
dial-peer voice 999110 pots
application mgcpapp
port 1/1/0
!
!
! Enable credentials service on the gateway.
credentials
ip source-address 10.1.1.22 port 2445
trustpoint srstca
!
!
! Enable SRST mode.
call-manager-fallback
transport-tcp-tls
secondary-dialtone 9
transfer-system full-consult
ip source-address 10.1.1.22 port 2000
max-ephones 15
max-dn 30
transfer-pattern .....
.
.
.
```

Control Plane Policing: Example

This section provides a configuration example for the security best practice of protecting the credentials service port using control plane policing. Control plane policing protects the gateway and maintains packet forwarding and protocol states despite a heavy traffic load. For more information on control planes, see the Control Plane Policing documentation.

```
Router# show running-config
.
.
.
! Allow trusted host traffic.
access-list 140 deny tcp host 10.1.1.11 any eq 2445
! Rate-limit all other traffic.
access-list 140 permit tcp any any eq 2445
access-list 140 deny ip any any
! Define class-map "sccp-class."
class-map match-all sccp-class
match access-group 140
policy-map control-plane-policy
class sccp-class
police 8000 1500 1500 conform-action drop exceed-action drop
! Define aggregate control plane service for the active Route Processor.
control-plane
service-policy input control-plane-policy
```

Configuring Secure SIP Call Signaling and SRTP Media with Cisco SRST

Cisco Unified Survivable Remote Site Telephony (Cisco SRST) provides secure call signaling and Secure Real-time Transport Protocol (SRTP) for media encryption to establish a secure, encrypted connection between Cisco Unified IP Phones and gateway devices.

Prerequisites for Configuring Secure SIP Call Signaling and SRTP Media with Cisco SRST

- Cisco IOS Release 15.0(1)XA and later releases.
- Cisco Unified IP Phone firmware release 8.5(3) or later.
- Complete the prerequisites and necessary tasks found in Prerequisites for Configuring SIP SRST Features Using Back-to-Back User Agent Mode.
- Prepare the Cisco Unified SIP SRST device to use certificates as documented in in Preparing the Cisco Unified SRST Router for Secure Communication.

Restrictions for Configuring Secure SIP Call Signaling and SRTP Media with Cisco SRST

SIP phones may be configured on the Cisco Unified CM with an authenticated device security mode. The Cisco Unified CM ensures integrity and authentication for the phone using a TLS connection with NULL-SHA cipher for signaling. If an authenticated SIP phone fails over to the Cisco Unified SRST device, it will register using TCP instead of TLS/TCP, thus disabling the authenticated mode until the phone fails back to the Cisco Unified CM.

- By default, non-secure TCP SIP phones are permitted to register to the SRST device on failover from the primary call control. Support for TCP SIP phones requires the secure SRST configuration described in this section even if no encrypted phones are deployed. Without the secure SIP SRST configuration, TCP phones will register to the SRST device using UDP for signaling transport.

Information About Cisco Unified SIP SRST Support of Secure SIP Signaling and SRTP Media

Beginning with Cisco IP Phone firmware 8.5(3) and Cisco IOS Release 15.0(1)XA, Cisco SRST supports SIP signaling over UDP, TCP, and TLS connections, providing both RTP and SRTP media connections based on the security settings of the IP phone.

Cisco SRST SIP-to-SIP and SIP-to-PSTN support includes the following features:

- Basic calling
- Hold/resume
- Conference
- Transfer
- Blind transfer
- Call forward

Cisco SRST SIP-to-other (including SIP-to-SCCP) support includes basic calling, although other features may work.

Configuring Cisco Unified Communications Manager

Like SCCP-controlled devices, SIP-controlled devices will use the SRST Reference profile that is listed in their assigned Device Pool. The SRST Reference profile must have the "Is SRST Secure" check box selected if SIP/TLS communication is desired in the event of a WAN failure.

Note All Cisco Unified IP Phones must have their firmware updated to version 8.5(3) or later. Devices with firmware earlier than 8.5(3) will need to have a separate Device Pool and SRST Reference profile created without the "Is SRST Secure" option selected; SIP-controlled devices in this Device Pool will use SIP over UDP to attempt to register to the SRST router.

In Cisco Unified CM Administration, under **System** > **SRST**:

- For the secure SRST profile, Is SRST Secure? must be checked. The SIP port must be 5061.
- For the non-secure SRST profile, the Is SRST Secure? checkbox should NOT be checked and the SIP port should be 5060.

Under **Device** > **Phone**:

- Secure phones must belong to the pool that uses the secure SRST profile.
- Non-secure phones must belong to the pool that uses the non-secure SRST profile.

Note SIP phones will use the transport method assigned to them by their Phone Security Profile.

Configuring Phones

This section specifies that SRTP should be used to enable secure calls and allows non-secure calls to "fallback" to using RTP media.

SUMMARY STEPS

1. **enable**
2. **configure terminal**

3. **voice service voip**
4. **srtp**
5. **allow-connections sip to h323**
6. **allow-connections sip to sip**
7. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice service voip** **Example:** `Router(config)# voice service voip`	Enters voice service configuration mode.
Step 4	**srtp** **Example:** `Router(config-voi-serv)# srtp`	Specifies that SRTP be used to enable secure calls.
Step 5	**allow-connections sip to h323** **Example:** `Router(config-voi-serv)# allow-connections sip to h323`	(Optional) Allows connections from SIP endpoints to H.323 endpoints.
Step 6	**allow-connections sip to sip** **Example:** `Router(config-voi-serv)# allow-connections sip to sip`	Allows connections from SIP endpoints to SIP endpoints.

Step 7	**end**
	Example:
	`Router(conf-voi-serv)# end`
		Ends the current configuration session and returns to privileged EXEC mode.

Configuring SIP options for Secure SIP SRST

This section explains how to configure secure SIP SRTP.

SUMMARY STEPS

1. **enable**
2. **configure terminal**

3. **voice service voip**
4. **sip**
5. **url sip | sips**
6. **srtp negotiate cisco**
7. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice service voip** **Example:** `Router(config)# voice service voip`	Enters voice service configuration mode.
Step 4	**sip** **Example:** `Router(config-voi-serv)# sip`	Enters SIP configuration mode.
Step 5	**url sip | sips** **Example:** `Router(conf-serv-sip)# url sips`	To configure secure mode, use the **sips** keyword to generate URLs in SIP secure (SIPS) format for VoIP calls. To configure device-default mode, use the **sip** keyword to generate URLs in SIP format for VoIP calls.
Step 6	**srtp negotiate cisco** **Example:** `Router(conf-serv-sip)# srtp negotiate cisco`	Enables a Cisco IOS SIP gateway to negotiate the sending and accepting of RTP profiles in response to SRTP offers.

Step 7	**end**
	Example:
	`Router(conf-serv-sip)# end`
		Ends the current configuration session and returns to privileged EXEC mode.

Configuring SIP SRST Security Policy

This section explains how to secure mode to block registration of non-secure phones to the SRST router.

SUMMARY STEPS

1. **voice register global**
2. **security-policy secure**
3. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**voice register global** **Example:** `Router(config)# voice register global`	Enters voice register global configuration mode.
Step 2	**security-policy secure** **Example:** `Router(config-register-global)# security-policy secure`	Configures SIP registration security policy so that only SIP/TLS/TCP connections are allowed. For device-default mode, use the **no security-policy** command. Device-default mode allows non-secure devices to register without using TLS. **Note** We recommend that **security-policy secure** is configured for the Secure SRST feature, so that non-secure phones do not fall back on Secure SRST.

Step 3	**end**
	Example:
	`Router(config-register-global)# end`
		Ends the current configuration session and returns to privileged EXEC mode.

Configuring SIP User Agent for Secure SIP SRST

This section explains how the strict-cipher limits the allowed encryption algorithms.

Multiple Trustpoints

Use the default trustpoint configuration under **sip-ua** config mode for phones registering to Unified SRST in secure mode. For example, **srstca** is the default trustpoint for Secure SRST. This default signaling trustpoint is used for all SIP TLS interactions from SIP phones to Unified Secure SRST router.

In a deployment scenario with multiple trustpoints, communication with a service provider over a secure trunk with certificate issued by CA is achieved using the CLI command *8.41.20.20 255.255.0.0***trustpoint** *srst-trunk1* under **sip-ua** config mode.

SUMMARY STEPS

1. **sip-ua**
2. **registrar ipv4**: *destination-address* **expires** *seconds*
3. **xfer target dial-peer**
4. **crypto signaling default trustpoint** *string***[strict-cipher]**
5. **crypto signaling remote-addr**{ *ip address* |*subnet mask* }**trustpoint** *trustpoint-name*
6. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**sip-ua** **Example:**	Enters SIP user-agent configuration mode.

	Command or Action	Purpose
	`Router(config)# sip-ua`	
Step 2	**registrar ipv4**: *destination-address* **expires** *seconds* **Example:** `Router(config-sip-ua)# registrar ipv4:192.168.2.10 expires 3600`	Enables the gateway to register E.164 telephone numbers with primary and secondary external SIP registrars. *destination-address* is the IP address of the primary SIP registrar server.
Step 3	**xfer target dial-peer** **Example:** `Router(config-sip-ua)# xfer target dial-peer`	Specifies that SRST should use the dial-peer as a transfer target instead of what is in the message body.
Step 4	**crypto signaling default trustpoint** *string***[strict-cipher]** **Example:** `Router(config-sip-ua)# crypto signaling default trustpoint 3745-SRST strict-cipher`	identifies the **trustpoint** *string* keyword and argument used during the TLS handshake. The **trustpoint***string* keyword and argument refer to the gateway's certificate generated as part of the enrollment process, using Cisco IOS public-key infrastructure (PKI) commands. The **strict-cipher** keyword restricts support to TLS RSA encryption with the Advanced Encryption Standard-128 (AES-128) cipher-block-chaining (CBC) Secure Hash Algorithm (SHA) (TLS_RSA_WITH_AES_128_CBC_SHA) cipher suite. To configure device-default mode, omit the **strict-cipher** keyword.
Step 5	**crypto signaling remote-addr**{ *ip address* |*subnet mask* }**trustpoint** *trustpoint-name* **Example:** `Router(config-sip-ua)# crypto signaling remote-addr 8.41.20.20 255.255.0.0 trustpoint srst-trunk1`	The trustpoint label refers to the CUBE's certificate that is generated with the Cisco IOS PKI commands as part of the enrollment process. Keywords and arguments are as follows: • **remote-addr** *ip address*—Associates an IP address to a trustpoint. • **trustpoint** *trustpoint-name*—Refers to the SIP gateways certificate generated as part of the enrollment process using Cisco IOS PKI commands

Step 6	**end**
	Example:
	`Router(config-sip-ua)# end`
		Ends the current configuration session and returns to privileged EXEC mode.

Example

The following example shows a sample configuration of multiple trustpoints for a Unified SRST deployment. In this example, the *srst-trunk1* trustpoint points to the network with IP address *8.39.0.0*, and *srst-trunk2* trustpoint points to the network with IP address *8.41.20.20*.

```
sip-ua
crypto signaling remote-addr 8.39.0.0 255.255.0.0 trustpoint srst-trunk1
crypto signaling remote-addr 8.41.20.20 255.255.0.0 trustpoint srst-trunk2
crypto signaling default trustpoint secsrst
```

Verifying the Configuration

The following examples show a sample configuration displayed by the **show sip-ua status registrar** command and the **show voice register global** command.

The **show sip-ua status registrar** command in privileged EXEC mode displays all SIP endpoints that are currently registered with the contact address.

```
Router# show sip-ua status registrar
Line         destination     expires(sec) contact
transport    call-id
             peer
============ =============== ============ ===============
3029991      192.168.2.108   388          192.168.2.108
TLS          00120014-4ae40064-f1a3e9fe-8d301072@192.168.2.1
             40004
3029993      192.168.2.103    382          192.168.2.103
TCP          001bd433-1c840052-655cd596-4e992eed@192.168.2.1
             40011
3029982      192.168.2.106    406          192.168.2.106
UDP          001d452c-dbba0056-0481d321-1f3f848d@192.168.2.1
             40001
3029983      192.168.2.106    406          192.168.2.106
UDP          001d452c-dbba0057-1c69b699-d8dc6625@192.168.2.1
             40003
3029992      192.168.2.107    414           192.168.2.107
TLS          001e7a25-50c9002c-48ef7663-50c71794@192.168.2.1
             40005
```

The **show voice register global** command in privileged EXEC mode displays all global configuration parameters associated with SIP phones.

```
Router# show voice register global
    CONFIG [Version=8.0]
    ========================
    Version 8.0
    Mode is srst
    Max-pool is 50
    Max-dn is 100
    Outbound-proxy is enabled and will use global configured value
    Security Policy: DEVICE-DEFAULT
    timeout interdigit 10
    network-locale[0] US (This is the default network locale for this box)
    network-locale[1] US
    network-locale[2] US
    network-locale[3] US
    network-locale[4] US
    user-locale[0] US (This is the default user locale for this box)
    user-locale[1] US
    user-locale[2] US
    user-locale[3] US
    user-locale[4] US
    Router#
```

Configuration Example for Cisco Unified SIP SRST

```
Current configuration : 15343 bytes
!
! Last configuration change at 05:34:06 UTC Tue Jun 13 2017
! NVRAM config last updated at 11:57:03 UTC Thu Jun 8 2017
!
version 16.7
service timestamps debug datetime msec
service timestamps log datetime msec
platform qfp utilization monitor load 80
no platform punt-keepalive disable-kernel-core
!
hostname router
!
boot-start-marker
boot-end-marker
!
vrf definition Mgmt-intf
!
address-family ipv4
exit-address-family
!
address-family ipv6
exit-address-family
!
! card type command needed for slot/bay 0/3
no logging queue-limit
logging buffered 20000000
no logging rate-limit
no logging console
enable password xxxx
!
no aaa new-model
!
subscriber templating
!
multilink bundle-name authenticated
!
crypto pki server SRST-CA-2
database level complete
no database archive
grant auto
!
crypto pki trustpoint TRUSTPT-SRST-CA-2
enrollment url http://10.0.0.1:80
serial-number
revocation-check none
rsakeypair srstcakey 2048
rsakeypair SRST-CA-2
!
crypto pki trustpoint SRST-CA-2
revocation-check crl
rsakeypair SRST-CA-2
!
crypto pki trustpoint Cisco_Manufacturing_CA
enrollment terminal
revocation-check none
!
crypto pki trustpoint CAPF-3a66269a
enrollment terminal
revocation-check none
!
crypto pki trustpoint Cisco_Root_CA_2048
enrollment terminal
revocation-check none
!
!
crypto pki certificate chain TRUSTPT-SRST-CA-2
```

```
certificate 02
3082020B 30820174 A0030201 02020102 300D0609 2A864886 F70D0101 05050030
14311230 10060355 04031309 53525354 2D43412D 32301E17 0D313730 36303831
31333131 325A170D 31383036 30383131 33313132 5A303231 30301206 03550405
130B4647 4C313735 31313150 42301A06 092A8648 86F70D01 0902160D 416E7473
41726D79 2D343430 3030819F 300D0609 2A864886 F70D0101 01050003 818D0030
81890281 81009E24 6259A98D A61C1973 45A95DA8 DE83ECAD C2B1B448 741F7E64
3D753BF1 19BD54FB 9A4D4A8E 7A2BA416 B93C40B3 A63A7C4D 7303498F 098EF07F
96F26F5F 49AD4E39 EC113DF4 696CB887 607D545A 52A11469 958F4C04 05868DF9
317456F6 3D23837C D46331FA 69FB29E8 3211E01C A7AB19A3 94DAC09F 97601196
A08D7073 76210203 010001A3 4F304D30 0B060355 1D0F0404 030205A0 301F0603
551D2304 18301680 142110B8 F25BD9BD E1D401EC 9D11DC0E AE52CDB8 2F301D06
03551D0E 04160414 2110B8F2 5BD9BDE1 D401EC9D 11DC0EAE 52CDB82F 300D0609
2A864886 F70D0101 05050003 8181003A DC409694 26D08A31 7B4F495F 002D4E57
B28669A9 10E93C68 A9556659 97D326EC A5508201 C1A86659 B1CDC910 73097FCA
F6174794 1057DDDE DBA666D6 0BAFC503 96A10BE5 5FCA3B93 5D377ABE BC9B2774
3732DF01 CE3BF12B 1899AA69 F7EC8726 A1964C5A D6A99A0E E27EE2A0 15A7D364
793C6C8D 961C77E4 397F9CB4 C6A271
quit
certificate ca 01
30820201 3082016A A0030201 02020101 300D0609 2A864886 F70D0101 04050030
14311230 10060355 04031309 53525354 2D43412D 32301E17 0D313730 36303831
31323135 305A170D 32303036 30373131 32313530 5A301431 12301006 03550403
13095352 53542D43 412D3230 819F300D 06092A86 4886F70D 01010105 0003818D
00308189 02818100 9E246259 A98DA61C 197345A9 5DA8DE83 ECADC2B1 B448741F
7E643D75 3BF119BD 54FB9A4D 4A8E7A2B A416B93C 40B3A63A 7C4D7303 498F098E
F07F96F2 6F5F49AD 4E39EC11 3DF4696C B887607D 545A52A1 1469958F 4C040586
8DF93174 56F63D23 837CD463 31FA69FB 29E83211 E01CA7AB 19A394DA C09F9760
1196A08D 70737621 02030100 01A36330 61300F06 03551D13 0101FF04 05300301
01FF300E 0603551D 0F0101FF 04040302 0186301F 0603551D 23041830 16801421
10B8F25B D9BDE1D4 01EC9D11 DC0EAE52 CDB82F30 1D060355 1D0E0416 04142110
B8F25BD9 BDE1D401 EC9D11DC 0EAE52CD B82F300D 06092A86 4886F70D 01010405
00038181 0018859E D39C6A05 63509442 8746D970 BB716DE2 E82BA822 58AA55AD
AC37260F 36BFDFE6 F2D0E489 A8D23690 791AD903 F19AC857 5002E621 A5927ACC
DCB759C0 B126ACAB C53BF054 1F62D895 A895C50A E3AE83E3 EC68F346 50B88D39
BB053EE9 5D466AE4 C6B4593D 7EFA7A78 213C0766 7307A051 78FED92E 5A34AAB6
98D2A59C 31
quit
crypto pki certificate chain SRST-CA-2
certificate ca 01
30820201 3082016A A0030201 02020101 300D0609 2A864886 F70D0101 04050030
14311230 10060355 04031309 53525354 2D43412D 32301E17 0D313730 36303831
31323135 305A170D 32303036 30373131 32313530 5A301431 12301006 03550403
13095352 53542D43 412D3230 819F300D 06092A86 4886F70D 01010105 0003818D
00308189 02818100 9E246259 A98DA61C 197345A9 5DA8DE83 ECADC2B1 B448741F
7E643D75 3BF119BD 54FB9A4D 4A8E7A2B A416B93C 40B3A63A 7C4D7303 498F098E
F07F96F2 6F5F49AD 4E39EC11 3DF4696C B887607D 545A52A1 1469958F 4C040586
8DF93174 56F63D23 837CD463 31FA69FB 29E83211 E01CA7AB 19A394DA C09F9760
1196A08D 70737621 02030100 01A36330 61300F06 03551D13 0101FF04 05300301
01FF300E 0603551D 0F0101FF 04040302 0186301F 0603551D 23041830 16801421
10B8F25B D9BDE1D4 01EC9D11 DC0EAE52 CDB82F30 1D060355 1D0E0416 04142110
B8F25BD9 BDE1D401 EC9D11DC 0EAE52CD B82F300D 06092A86 4886F70D 01010405
00038181 0018859E D39C6A05 63509442 8746D970 BB716DE2 E82BA822 58AA55AD
AC37260F 36BFDFE6 F2D0E489 A8D23690 791AD903 F19AC857 5002E621 A5927ACC
DCB759C0 B126ACAB C53BF054 1F62D895 A895C50A E3AE83E3 EC68F346 50B88D39
BB053EE9 5D466AE4 C6B4593D 7EFA7A78 213C0766 7307A051 78FED92E 5A34AAB6
98D2A59C 31
quit
crypto pki certificate chain Cisco_Manufacturing_CA
certificate ca 6A6967B3000000000003
308204D9 308203C1 A0030201 02020A6A 6967B300 00000000 03300D06 092A8648
86F70D01 01050500 30353116 30140603 55040A13 0D436973 636F2053 79737465
6D73311B 30190603 55040313 12436973 636F2052 6F6F7420 43412032 30343830
1E170D30 35303631 30323231 3630315A 170D3239 30353134 32303235 34325A30
```

```
39311630 14060355 040A130D 43697363 6F205379 7374656D 73311F30 1D060355
04031316 43697363 6F204D61 6E756661 63747572 696E6720 43413082 0120300D
06092A86 4886F70D 01010105 00038201 0D003082 01080282 010100A0 C5F7DC96
943515F1 F4994EBB 9B41E17D DB791691 BBF354F2 414A9432 6262C923 F79AE7BB
9B79E807 294E30F5 AE1BC521 5646B0F8 F4E68E81 B816CCA8 9B85D242 81DB7CCB
94A91161 121C5CEA 33201C9A 16A77DDB 99066AE2 36AFECF8 0AFF9867 07F430EE
A5F8881A AAE8C73C 1CCEEE48 FDCD5C37 F186939E 3D71757D 34EE4B14 A9C0297B
0510EF87 9E693130 F548363F D8ABCE15 E2E8589F 3E627104 8726A415 620125AA
D5DFC9C9 5BB8C9A1 077BBE68 92939320 A86CBD15 75D3445D 454BECA8 DA60C7D8
C8D5C8ED 41E1F55F 578E5332 9349D5D9 0FF836AA 07C43241 C5A7AF1D 19FFF673
99395A73 67621334 0D1F5E95 70526417 06EC535C 5CDB6AEA 35004102 0103A382
01E73082 01E33012 0603551D 130101FF 04083006 0101FF02 0100301D 0603551D
0E041604 14D0C522 26AB4F46 60ECAE05 91C7DC5A D1B047F7 6C300B06 03551D0F
04040302 01863010 06092B06 01040182 37150104 03020100 30190609 2B060104
01823714 02040C1E 0A005300 75006200 43004130 1F060355 1D230418 30168014
27F3C815 1E6E9A02 0916AD2B A089605F DA7B2FAA 30430603 551D1F04 3C303A30
38A036A0 34863268 7474703A 2F2F7777 772E6369 73636F2E 636F6D2F 73656375
72697479 2F706B69 2F63726C 2F637263 61323034 382E6372 6C305006 082B0601
05050701 01044430 42304006 082B0601 05050730 02863468 7474703A 2F2F7777
772E6369 73636F2E 636F6D2F 73656375 72697479 2F706B69 2F636572 74732F63
72636132 3034382E 63657230 5C060355 1D200455 30533051 060A2B06 01040109
15010200 30433041 06082B06 01050507 02011635 68747470 3A2F2F77 77772E63
6973636F 2E636F6D 2F736563 75726974 792F706B 692F706F 6C696369 65732F69
6E646578 2E68746D 6C305E06 03551D25 04573055 06082B06 01050507 03010608
2B060105 05070302 06082B06 01050507 03050608 2B060105 05070306 06082B06
01050507 0307060A 2B060104 0182370A 0301060A 2B060104 01823714 02010609
2B060104 01823715 06300D06 092A8648 86F70D01 01050500 03820101 0030F330
2D8CF2CA 374A6499 24290AF2 86AA42D5 23E8A2EA 2B6F6923 7A828E1C 4C09CFA4
4FAB842F 37E96560 D19AC6D8 F30BF5DE D027005C 6F1D91BD D14E5851 1DC9E3F7
38E7D30B D168BE8E 22A54B06 E1E6A4AA 337D1A75 BA26F370 C66100A5 C379265B
A719D193 8DAB9B10 11291FA1 82FDFD3C 4B6E65DC 934505E9 AF336B67 23070686
22DAEBDC 87CF5921 421AE9CF 707588E0 243D5D7D 4E963880 97D56FF0 9B71D8BA
6019A5B0 6186ADDD 6566F6B9 27A2EE2F 619BBAA1 3061FDBE AC3514F9 B82D9706
AFC3EF6D CC3D3CEB 95E981D3 8A5EB6CE FA79A46B D7A25764 C43F4CC9 DBE882EC
0166D410 88A256E5 3C57EDE9 02A84891 6307AB61 264B1A13 9FE4DCDA 5F
quit
crypto pki certificate chain CAPF-3a66269a
certificate ca 583BD5B4844C8BC172B8C4979092A067
308203C3 308202AB A0030201 02021058 3BD5B484 4C8BC172 B8C49790 92A06730
0D06092A 864886F7 0D01010B 05003071 310B3009 06035504 06130249 4E310E30
0C060355 040A0C05 63697363 6F311230 10060355 040B0C09 75637467 2D656467
65311630 14060355 04030C0D 43415046 2D336136 36323639 61311230 10060355
04080C09 6B61726E 6174616B 61311230 10060355 04070C09 62616E67 616C6F72
65301E17 0D313730 35323931 30333631 335A170D 32323035 32383130 33363132
5A307131 0B300906 03550406 1302494E 310E300C 06035504 0A0C0563 6973636F
31123010 06035504 0B0C0975 6374672D 65646765 31163014 06035504 030C0D43
4150462D 33613636 32363961 31123010 06035504 080C096B 61726E61 74616B61
31123010 06035504 070C0962 616E6761 6C6F7265 30820122 300D0609 2A864886
F70D0101 01050003 82010F00 3082010A 02820101 00BC774F BAED3986 05BDFFBC
4EABBFA7 1F73D150 2989EFF2 902502F6 248DA7AB 261E474C 08A4BB6F 35B10449
0A6A3D94 E2C6EB98 57BECE0C 34F30517 CA6CC9B2 710B511B 8826E0AB 733FF26F
F7ADC4B9 76118300 6156072C 43F78E5E 3AD7C92B 54CB5BDB 00B53FC8 875100C4
056BC4A7 0F96CE69 E58B1C22 194CCEC6 968ECF9B 08B7B7B2 0FF0800E 43764BB1
E6ED36C0 A738F762 81A88F6D E464E2A5 FD74207F 1EC7ACAC 2F63B04D E0E9DA4C
901A1710 E3D1C069 82EFF77E 0597254D 149C1263 EC67DAE9 305FD8BF C7410B17
8C6DE9FF 28A37514 86AF828C BC698DD5 F18A3B66 9D8D895A 5562E08D 383F790A
A5C7F6F6 915CB558 042E5B99 71F7169D B3AFA699 2B020301 0001A357 3055300B
0603551D 0F040403 0202A430 13060355 1D25040C 300A0608 2B060105 05070301
301D0603 551D0E04 16041475 71EC5D35 1A431511 7E8C8462 6E65E570 7C551930
12060355 1D130101 FF040830 060101FF 02010030 0D06092A 864886F7 0D01010B
05000382 0101008F 0D3E9F3E 3574100D 97AD876D B4015C21 300A1BD0 59D5C9BF
41A8448D 597CD278 718A6431 BA94C042 7EC64BA0 71F04501 C33C1664 16484373
F3C226A7 256363A9 8BE97291 6B25B8B4 E3DB84C3 3DDB63E7 A9D8D577 6B8F37B3
7CFCE019 D6F09573 946191F7 C4028465 B072DF74 9D6DED45 CA9E6A3B 1401D1A3
```

```
5449EDCE 9FA593E3 2FD71031 C7C7EB9C 045DAAFE C67603BF DAB40EE0 352C009F
EAAA6816 A11F6D8B 7C406211 1045A0C6 488B34E1 AF968FAF 3705A364 1EE21A1D
B7080EDC 40D4AA15 E110C5F1 D8A57561 DB2B09F1 0779B855 3998CE22 C471B5CB
09605E24 99855176 2D1CA40E BEBC2F23 7434CA2B 8D1C5EFB 822147CC 81F98825
47A1A14F DC5480
quit
crypto pki certificate chain Cisco_Root_CA_2048
certificate ca 5FF87B282B54DC8D42A315B568C9ADFF
30820343 3082022B A0030201 0202105F F87B282B 54DC8D42 A315B568 C9ADFF30
0D06092A 864886F7 0D010105 05003035 31163014 06035504 0A130D43 6973636F
20537973 74656D73 311B3019 06035504 03131243 6973636F 20526F6F 74204341
20323034 38301E17 0D303430 35313432 30313731 325A170D 32393035 31343230
32353432 5A303531 16301406 0355040A 130D4369 73636F20 53797374 656D7331
1B301906 03550403 13124369 73636F20 526F6F74 20434120 32303438 30820120
300D0609 2A864886 F70D0101 01050003 82010D00 30820108 02820101 00B09AB9
ABA7AF0A 77A7E271 B6B46662 94788847 C6625584 4032BFC0 AB2EA51C 71D6BC6E
7BA8AABA 6ED21588 48459DA2 FC83D0CC B98CE026 68704A78 DF21179E F46105C9
15C8CF16 DA356189 9443A884 A8319878 9BB94E6F 2C53126C CD1DAD2B 24BB31C4
2BFF8344 6FB63D24 7709EABF 2AA81F6A 56F6200F 11549781 75A725CE 596A8265
EFB7EAE7 E28D758B 6EF2DD4F A65E629C CF100A64 D04E6DCE 2BCC5BF5 60A52747
8D69F47F CE1B70DE 701B20D6 6ECDA601 A83C12D2 A93FA06B 5EBB8E20 8B7A91E3
B568EEA0 E7C40174 A8530B2B 4A9A0F65 120E824D 8E63FDEF EB9B1ADB 53A61360
AFC27DD7 C76C1725 D473FB47 64508180 944CE1BF AE4B1CDF 92ED2E05 DF020103
A351304F 300B0603 551D0F04 04030201 86300F06 03551D13 0101FF04 05300301
01FF301D 0603551D 0E041604 1427F3C8 151E6E9A 020916AD 2BA08960 5FDA7B2F
AA301006 092B0601 04018237 15010403 02010030 0D06092A 864886F7 0D010105
05000382 0101009D 9D8484A3 41A97C77 0CB753CA 4E445062 EF547CD3 75171CE8
E0C6484B B6FE4C3A 198156B0 56EE1996 62AA5AA3 64C1F64E 5433C677 FEC51CBA
E55D25CA F5F0939A 83112EE6 CBF87445 FEE705B8 ABE7DFCB 4BE13784 DAB98B97
701EF0E2 8BD7B0D8 0E9DB169 D62A917B A9494F7E E68E95D8 83273CD5 68490ED4
9DF62EEB A7BEEB30 A4AC1F44 FC95AB33 06FB7D60 0ADEB48A 63B09CA9 F2A4B953
0187D068 A4277FAB FFE9FAC9 40388867 B439C684 6F57C953 DBBA8EEE C043B2F8
09836EFF 66CF3EEF 17B35818 2509345E E3CBD614 B6ECF292 6F74E42F 812AD592
91E0E097 3C326805 854BD1F7 57E2521D 931A549F 0570C04A 71601E43 0B601EFE
A3CE8119 E10B35
quit
!
voice service voip
no ip address trusted authenticate
media bulk-stats
media disable-detailed-stats
allow-connections sip to sip
srtp
no supplementary-service sip refer
supplementary-service media-renegotiate
no supplementary-service sip handle-replaces
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
sip
registrar server expires max 120 min 60
!
voice register global
default mode
no allow-hash-in-dn
security-policy secure
max-dn 50
max-pool 40
!
voice register pool 1
id network 10.0.0.1 mask 255.255.0.0
dtmf-relay rtp-nte
codec g711ulaw
!
voice hunt-group 1 sequential
final 89898
```

```
list 1008,2005
timeout 5
pilot 1111
!
voice-card 0/1
no watchdog
!
voice-card 0/2
no watchdog
!
voice-card 0/3
no watchdog
!
voice-card 1/0
no watchdog
!
license udi pid ISR4451-X/K9 sn FOC1743565L
license accept end user agreement
license boot level uck9
license boot level securityk9
no license smart enable
diagnostic bootup level minimal
!
spanning-tree extend system-id
!
redundancy
mode none
!
interface GigabitEthernet0/0/0
ip address 10.0.0.1 255.255.0.0
negotiation auto
!
interface GigabitEthernet0/0/1
no ip address
negotiation auto
!
interface GigabitEthernet0/0/2
ip address 10.0.0.1 255.0.0.0
negotiation auto
!
interface GigabitEthernet0/0/3
no ip address
negotiation auto
!
interface Service-Engine0/1/0
shutdown
!
interface Service-Engine0/2/0
shutdown
!
interface Service-Engine0/3/0
!
interface Service-Engine1/0/0
!
interface GigabitEthernet0
vrf forwarding Mgmt-intf
no ip address
negotiation auto
!
ip forward-protocol nd
ip http server
no ip http secure-server
ip route 0.0.0.0 0.0.0.0 10.0.0.1
!
```

```
ip ssh server algorithm encryption aes128-ctr aes192-ctr aes256-ctr
ip ssh client algorithm encryption aes128-ctr aes192-ctr aes256-ctr
!
control-plane
!
!
voice-port 0/1/0
!
voice-port 0/1/1
!
voice-port 0/2/0
!
voice-port 0/2/1
!
voice-port 0/2/2
!
voice-port 0/2/3
!
mgcp behavior rsip-range tgcp-only
mgcp behavior comedia-role none
mgcp behavior comedia-check-media-src disable
mgcp behavior comedia-sdp-force disable
!
mgcp profile default
!
sip-ua
crypto signaling default trustpoint TRUSTPT-SRST-CA-2
!
!
credentials
ip source-address 10.0.0.1 port 2445
trustpoint TRUSTPT-SRST-CA-2
!
!
call-manager-fallback
max-conferences 8 gain -6
transfer-system full-consult
max-ephones 50
max-dn 50
call-park system application
fac standard
!
!
line con 0
exec-timeout 0 0
length 0
transport input none
stopbits 1
line aux 0
stopbits 1
line vty 0 4
exec-timeout 0 0
password xxxx
no login
length 0
transport preferred none
transport input telnet ssh
!
end
```

Configuration Example for SIP OAuth

The following is a configuration example to enable SIP OAuth in Secure SRST.

```
Router(config)#voice register pool 1
Router(config-register-pool)#?
voice register pool configuration commands:
  after-hour                After-hours call blocking
  alias                     Associate alias pattern
  application               Define application
  ata-ivr-pwd               Define ATA IVR Password using 0-9 digit
  busy-trigger-per-button   Define the number of calls that triggers call
                            forward busy per line on the sip phone
  call-forward              Define E.164 telephone number for call forward
  codec                     select the preferred codec to be used for SIP phone
  conference                Adhoc hardware conference configuration
  conference-pattern        Customized conference-pattern configuration
  cor                       Class of Restriction on dial-peer for this dn
  default                   Set a command to its defaults
  dialplan-pattern          Define E.164 telephone number prefix
  digit                     Enable digit collect command
  dtmf-relay                Transport DTMF digits across IP link
  emergency                 Emergency Assistance
  exclude                   Exclude Local Services
  exit                      Exit from voice register pool configuration mode
  feature-button            Define programmable line key
  id                        define phone or device id
  incoming                  Incoming called number
  logout-profile            enable extension mobility
  lpcor                     Voice registry pool lpcor setup
  max                       voice register pool max commands
  media                     Media mode setting for SIP extension
  night-service             Define night-service bell
  no                        Negate a command or set its defaults
  number                    Define E.164 telephone number
  overlap-signal            Configure Overlap Signaling support
  paging-dn                 set audio paging dn group for phone (use ephone
                            paging-dn number)
  phone-mode                Phone mode configuration in voice register pool
  pin                       Define 4-8 digit personal identification number
  preference                Configure the preference for the voip dial-peers to
                            be created
  presence                  enable call list feature
  provision-tag             define phone provision_tag
  proxy                     Define SIP proxy for this pool
  registration-timer        keepalive registration expires timer
  session-server            define controlling session-server
  sip-oauth                 Enable sip-oauth on Pool
  tone                      Generate tones
  transfer                  Transfer related configuration
  transfer-pattern          Customized transfer-pattern configuration
  translate-outgoing        Translation rule
  translation-profile       Translation profile
  vad                       Enable vad on dial-peer and phone
  voice-class               Set voice class parameters

Router(config-register-pool)#sip_oauth ?
  <cr>  <cr>
Router(config-register-pool)#sip_oauth
Router(config-register-pool)#end
```

The following is a configuration example to disable SIP OAuth in Secure SRST.

```
Router(config)#voice register pool 1
Router(config-register-pool)#no sip-oauth
Router(config-register-pool)#end
```

Configuration Examples for SHA2 Cipher Suites

The following is a configuration example to enable the SHA2 cipher suite for TLS v1.2 or v1.3 in SRST:

```
Device(config)#call-manager-fallback
Device(config-cm-fallback)#transport-tcp-tls ?
  v1.0  Enable TLS Version 1.0
  v1.1  Enable TLS Version 1.1
  v1.2  Enable TLS Version 1.2
  v1.3  Enable TLS Version 1.3

Device(config-cm-fallback)#transport-tcp-tls v1.2 ?
 sha2 Allow SHA2 ciphers only
  <cr>  <cr>
Device(config-cm-fallback)#transport-tcp-tls v1.2 sha2

(OR)

Device(config-cm-fallback)#transport-tcp-tls v1.3 ?
 sha2 Allow SHA2 ciphers only
  <cr>  <cr>
Device(config-cm-fallback)#transport-tcp-tls v1.3 sha2
```

Syslog Messages

For Unified SRST 14.4 and later releases, when a handshake between CUCM and SIP SRST (CAPF) fails due to TLS version mismatch, then it generates a syslog message in the following format:

```
Dec 13 06:15:54.587: %EPHONE_CRED-2-CTL_TLS_HANDSHAKE_FAILED: Reason: unsupported
 protocol,
Socket ID: 1, local address : 10.1.20.11:2445 and  remote address :
10.5.10.50:56372
```

When a handshake between Voice Gateway and SCCP SRST fails due to TLS version mismatch, then it generates a syslog message in the following format:

```
Dec 13 15:31:52.949: %SKINNYSECURESERVICE-3-TLS_HANDSHAKE_FAILED:
Reason: unsupported protocol, SocketID: 0, local address : 10.1.20.11:2443 and
remote address : 10.1.20.179:12239
```

When a handshake between CUCM and SIP SRST fails due to cipher mismatch while fetching the sip-oauth keys, then it generates a syslog message in the following format:

```
Dec 13 06:42:59.612: %HTTPC-6-TLS_HANDSHAKE_FAILED:  Reason: sslv3 alert handshake
 failure, Connection ID: 0, remote address : [2001:10:5:10::50]:8443
```

Additional References

The following sections provide references related to this feature.

Related Documents

Related Topic	Document Title
Cisco IOS voice configuration	• Cisco IOS Voice Configuration Library • Cisco IOS Voice Command Reference
Cisco Unified Communications Manager Documentation Guide for Release 8.0(2)	• Cisco Unified Communications Manager Documentation Guide for Release 8.0(2)
Cisco Unified SRST configuration	• Cisco Unified SRST and SIP SRST Command Reference
Cisco Unified SRST	• Cisco Unified SRST 8.0 Supported Firmware, Platforms, Memory, and Voice Products
Cisco Unified Communications Operating System Administration Guide, Release 6.1(1)	• Security
Configuring a Secure Survivable Remote Site Telephony (SRST) Reference	• Configuring a Secure Survivable Remote Site Telephony (SRST) Reference

Standards

Standard	Title
No new or modified standards are supported by this feature, and support for existing standards has not been modified by this feature.	—

MIBs

MIB	MIBs Link
No new or modified MIBs are supported by this feature, and support for existing MIBs has not been modified by this feature.	To locate and download MIBs for selected platforms, Cisco IOS releases, and feature sets, use Cisco MIB Locator found at the following URL: http://www.cisco.com/go/mibs

RFCs

RFC	Title
No new or modified RFCs are supported by this feature, and support for existing RFCs has not been modified by this feature.	—

Technical Assistance

Description	Link
The Cisco Support website provides extensive online resources, including documentation and tools for troubleshooting and resolving technical issues with Cisco products and technologies. To receive security and technical information about your products, you can subscribe to various services, such as the Product Alert Tool (accessed from Field Notices), the Cisco Technical Services Newsletter, and Really Simple Syndication (RSS) Feeds. Access to most tools on the Cisco Support website requires a Cisco.com user ID and password.	http://www.cisco.com/techsupport

Command Reference

The following commands are introduced or modified in the feature or features documented in this section. For information about these commands, see the *Cisco IOS Voice Command Reference* at http://www.cisco.com/en/US/docs/ios/voice/command/reference/vr_book.html. For information about all Cisco IOS commands, use the Command Lookup Tool at http://tools.cisco.com/Support/CLILookup or *Cisco IOS Command List, All Releases* at http://www.cisco.com/en/US/docs/ios/mcl/allreleasemcl/all_book.html.

- **security-policy**
- **show voice register global**
- **show voice register all**
- **transport-tcp-tls**

Feature Information for Secure SCCP and SIP SRST

The Feature Information for Secure SIP Call Signaling and SRTP Media with Cisco SRST table lists the release history for this feature.

Not all commands may be available in your Cisco IOS software release. For release information about a specific command, see the command reference documentation.

Use a Cisco Feature Navigator to find information about platform support and software image support. Cisco Feature Navigator enables you to determine which Cisco IOS and Catalyst OS software images support a specific software release, feature set, or platform. To access Cisco Feature Navigator, go to http://www.cisco.com/go/cfn. An account on Cisco.com isn’t required.

Note The Feature Information for Secure SIP Call Signaling and SRTP Media with Cisco SRST table lists only the Cisco IOS software release that introduced support for a given feature in a given Cisco IOS software release train. Unless noted otherwise, subsequent releases of that Cisco IOS software release train also support that feature.

Feature Name	Releases	Feature Information
Secure SIP Call Signaling and SRTP Media with Cisco SRST	15.0(1)XA	Adds Session Initiation Protocol/Transport Layer Security/Transmission Control Protocol (SIP/TLS/TCP) support for secure call signaling and Secure Real-time Transport Protocol (SRTP) for media encryption to establish a secure, encrypted connection between Cisco Unified IP Phones and a failover device using Cisco Unified Survivable Remote Site Telephony (Cisco SRST). The following commands were introduced or modified: **security-policy, show voice register global, show voice register all.**
SHA2-Cipher-Only Mode for Unified Secure SRST	Cisco IOS XE Cupertino 17.8.1a	Restricts Secure SIP SRST and Secure SCCP SRST to only using TLS 1.2 Cipher Suites.
SIP OAuth Client Registration for Unified Secure SRST	Cisco IOS XE Cupertino 17.8.1a	Introduced support for SIP OAuth authentication for Secure SRST.
Secure SIP with TLS version 1.3 Support	Cisco IOS XE 17.14.1a	Introduces the following support for secure SRST: • Secure SIP SRST and Secure SCCP SRST now support TLS version 1.3 and associated cipher suites. • Secure SIP SRST supports **minimum** configuration with only TLS version 1.2. • Secure SCCP SRST is enhanced to support SHA2 ciphers with TLS version 1.3.

CHAPTER 11

Configuring SIP Trunking on Unified SRST

This chapter describes how to configure SIP trunking on Cisco Unified Survivable Remote Site Telephony (Unified SRST).

This chapter describes the configuration recommendations and details on the various line side and SIP trunking features on Unified SRST. Also, details are provided on the co-location of Unified Border Element and Unified SRST.

Unified SRST and Unified Border Element Co-location

For Unified SRST Release 12.1 and later releases, you can deploy product instances of Cisco Unified Border Element and Unified SRST (only for SIP) on the same Cisco 4000 Series Integrated Services Router. Co-location of Unified SRST and Unified Border element is supported from the release Cisco IOS XE Fuji 16.7.1. All the Cisco SIP IP Phones are supported for this deployment. The phone support includes, but is not limited to:

- Cisco IP Phone 7800 Series
- Cisco IP Phone 8800 Series
- Cisco Unified IP Phone 9900 Series

When the Wide Area Network (WAN) is available, the router acts as a pure Cisco Unified Border Element, and not as a Unified SRST.

During a WAN outage, the phones registered to the Unified Communications Manager fall back on the Unified SRST. However, phones registered to Unified SRST can place or receive PSTN calls through SIP trunk.

The Unified SRST and the Unified Border Element feature set is limited to the features mentioned. The following features are supported on the phone when registered to Unified SRST:

- Incoming or Outgoing Basic Call
- Hold/Resume
- Call Forward
- Call Transfer

- Conference (Built-in Bridge)
- Hunt Groups
- MOH (for SIP lines in SRST mode)

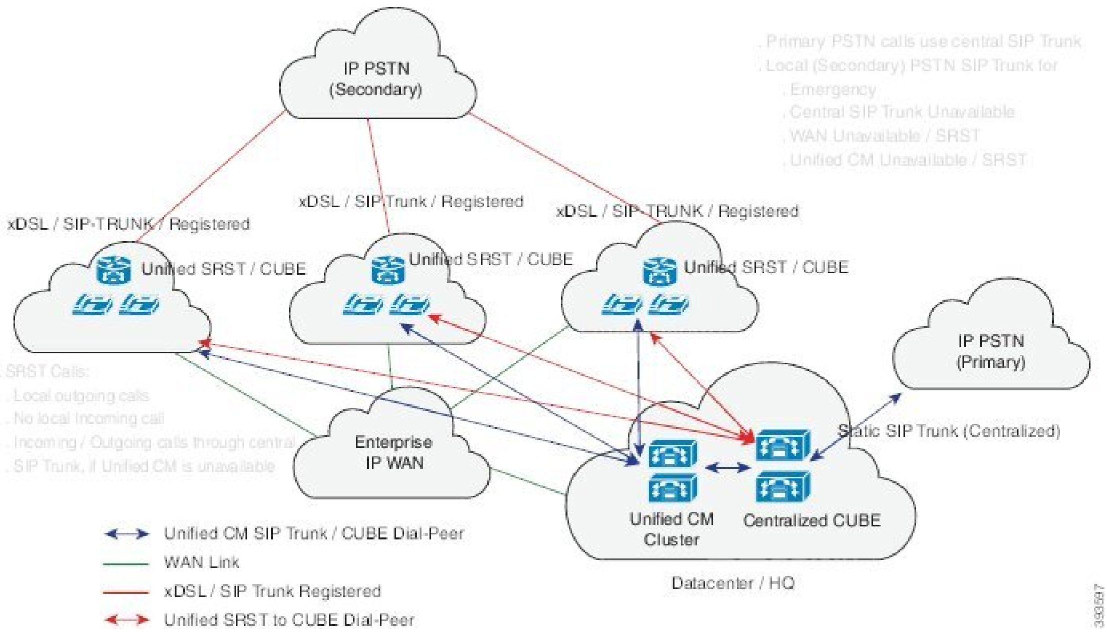
The list of SIP trunk features supported for Unified SRST and Unified Border Element co-location are:

- SIP-UA Registration/Authentication, Registrar, Register/Register Refresh
- SIP-Server, Outbound Proxy
- DNS Service Record
- Bind Global / Dial-peer
- SRTP / TLS, SRTP – RTP Interworking
- Connection Reuse
- IP Trust List
- Voice class tenant
- RTP-NTE DTMF
- P-Called-Party ID, Privacy Header (PAI)
- SIP Normalization

For more information on configuring tenants on SIP trunks, see Cisco Unified Border Element Configuration Guide. For more information on the recommended configurations for the Unified Border Element co-location, see Configuration Recommendations for Unified SRST and Unified Border Element Co-location, on page 347.

The Figure shows a co-located deployment of Unified SRST with Cisco Unified Border Element.

Figure 4: Co-located Deployment of Unifed SRST and Cisco Unified Border Elelement


IP PSTN
(Secondary)
. Primary PSTN calls use central SIP Trunk
. Local (Secondary) PSTN SIP Trunk for
. Emergency
. Central SIP Trunk Unavailable
. WAN Unavailable / SRST
. Unified CM Unavailable / SRST
xDSL / SIP-TRUNK / Registered
xDSL / SIP Trunk / Registered
xDSL / SIP-TRUNK / Registered
Unified SRST / CUBE
Unified SRST / CUBE
Unified SRST / CUBE
IP PSTN
(Primary)
Static SIP Trunk (Centralized)
. SRST Calls:
. Local outgoing calls
. No local Incoming call
. Incoming / Outgoing calls through central
. SIP Trunk, if Unified CM is unavailable
Enterprise
IP WAN
Unified CM
Cluster
Centralized CUBE
Datacenter / HQ
Unified CM SIP Trunk / CUBE Dial-Peer
WAN Link
xDSL / SIP Trunk Registered
Unified SRST to CUBE Dial-Peer
393597

Configuration Recommendations for Unified SRST and Unified Border Element Co-location

Note The recommended configurations have considered single SIP trunk dial-peer, acting as both inbound and outbound dial-peer to handle calls to and from the Service Provider. Similarly, a single dial-peer, acting as both inbound and outbound dial-peer to handle calls to and from the Communication Manager.

The dial-peers created after the phones (registered to Unified Communications Manager) fall back on Unified SRST are dynamic dial-peers. Hence, the configurations under **voice service voip** and **sip-ua** are inherited by these dynamic dial-peers. Move **voice service voip** and **sip-ua** configurations under **voice class tenant** configuration mode to avoid configuration conflict. The **voice class tenant** is included in the SIP trunk dial-peer configuration.

Similarly, the relevant global configurations are grouped under a **voice class tenant** and can be applied on the dial-peer toward Unified Communications Manager as well. These configurations grouped under the **voice class tenant** are used whenever the Unified Communications Manager is available (WAN is available). For sample configurations of the co-located deployment of Unified SRST and Unified Border Element, see Examples, on page 350.

The following are the configuration recommendations for the Unified SRST and Unified Border Element co-location:

- Move SIP trunk specific **voice service voip** and **sip-ua** configurations under **voice class tenant**. This is to avoid configuration conflict between SIP trunk and line side dial-peer configurations. When tenant is configured under dial-peer, the configurations are applied in the following order of preference:

1. Dial-peer configuration

2. Tenant configuration
3. Global configuration

Note Certain CLI commands which need to be moved under **tenant**, are moved under **dial-peer** configuration mode. This is because these CLIs are not available under **voice class tenant**. For example, the CLI command **srtp fallback** needs to be configured under **dial-peer**, not **voice class tenant** configuration mode.

- Use dial-peer groups feature to group multiple outbound dial-peers into a dial-peer group and configure this dial-peer group as the destination of an inbound dial-peer (Unified CM trunk). For more information on dial-peer groups, see Dial Peer Configuration Guide.
- Configure SIP Options Request Keepalives to monitor reachability towards Unified Communications Manager. For example:

```
voice class sip-options-keepalive 101
 up-interval 30
 retry 3 transport tcp

Options keepalive under dialpeer

dial-peer voice 101 voip
 description **CUCM/PBX**
 voice-class sip options-keepalive profile 101
```

- The relevant CLI commands for configuring dial-peer groups are:

  voice class dpg*dial-peer-group-id* (Creates a dial-peer group).

  destination dpg*dial-peer-group-id*(Specifies the dial-peer group from which the outbound dial-peer(s) is chosen).

- Avoid configuring dial-peer groups on the Service Provider SIP trunk dial-peer.
- Configure the destination pattern (.T) on the Unified Communications Manager dial-peer.
- It is mandatory to configure**voice class tenant** on the Service Provider SIP trunk dial-peer router. A configuration with voice class tenant on the Unified Communications Manager dial-peer is also validated, though it is not mandatory.
- Configure the CLI command **destination dpg***dial-peer-group-id* (destination dpg 101) on the Unified Communications Manager dial-peer. This dpg configuration has Service Provider SIP trunk dial-peer information. You can configure preferences for the dial-peers within the dial-peer group:

```
voice class dpg 1
dial-peer 2900 preference 2
dial-peer 3900 preference 1
```

- Do not configure **incoming called-number** (.T), on the Service Provider SIP trunk dial-peer. Match the incoming call from SIP trunk using the address information From URI.

```
voice class uri 201 sip
host dns:sip-trunk.sample

Under dial-peer:
incoming uri from 201
```

- Configure the CLI command **transport tcp tls v1.2** under**sip-ua** configuration mode, not **voice class tenant**.
- Avoid modification of contact header in a Secure SIP to SIP (and vice versa) call flow, as it leads to call establishment issues. If sip-profiles are used to modify header information from sips: to sip: in SIP REQUESTS and RESPONSES, there must be rules to include 'transport=tls' in the contact header.
- If dial-peers are using **voice class codec** , configure the same **voice class codec** under **voice register pool** too.
- Ensure that an srtp voice-class is created using the **voice class srtp-crypto***crypto-tag* command. A sample configuration is as follows:

  ```
  voice class srtp-crypto 1
  crypto 1 AES_CM_128_HMAC_SHA1_32
  crypto 2 AES_CM_128_HMAC_SHA1_80
  ```

- Configure the SIP Registrar under **voice service voip sip** configuration mode with maximum and minimum expiry time for an incoming registration using the CLI command **registrar server** [**expires** [**max** *sec*] [**min***sec*]].

  registrar server expires max*120***min***60*

- Move all the CLI commands related to SIP Bind feature under **voice class tenant** configuration mode. For example, it is recommended to have the CLI commands **voice-class sip bind control**, and **voice-class sip bind media**, under **voice class tenant** configuration mode.
- Exclude SIP ports from NAT services, if NAT is configured on the router. The recommended CLIs for excluding SIP ports from NAT services are:

  no ip nat service sip udp port 5060

  no ip nat service sip tcp port 5060

- Configure the CLI commands **no supplementary-service sip refer** , **no supplementary-service sip moved-temporarily**, **supplementary-service media-renegotiate** under **voice service voip** configuration mode.
- For the co-located deployment of Unified SRST and Unified Border Element, do not configure the CLI command **no transport udp** under **sip-ua** configuration mode. This is because, phones register to the Unified SRST device using UDP for signaling transport with the non-secure SIP SRST configuration.
- Playback of MOH from the flash memory of the router is supported for SIP lines in SRST mode in a co-located deployment of Unified SRST and Cisco Unified Border Element. Cisco IOS XE Fuji 16.7.1 and later releases support this feature.
- Redundancy is not supported for the co-located deployment of Unified SRST and Unified Border Element.
- Virtual interfaces are not supported for the co-located deployment of Unified SRST and Unified Border Element.
- Configure Media Inactivity Timer to enable router to monitor and disconnect calls if no Real-Time Protocol (RTP) packets are received within a configurable time period. A sample configuration is as follows:

  ```
  ip rtcp report interval 9000
  gateway
  media-inactivity-criteria all
  ```

```
timer receive-rtp 1200
timer receive-rtcp 5
```

Restrictions

The following restrictions are observed for a co-located deployment of Unified SRST and Unified Border Element:

- You need to disable the NAT firewall support for SIP trunk side, using the CLI commands **no ip nat service sip udp port 5060** and **no ip nat service sip tcp port 5060**.
- All the SIP trunk features are not supported in a Unified SRST and Unified Border Element co-location deployment. For the list of supported features, see Unified SRST and Unified Border Element Co-location, on page 345.

Examples

The following is a sample configuration for a voice class tenant:

```
voice class tenant 1
registrar ipv4:10.64.86.64:5061:5061 scheme sips expires 240 tcp tls auth-realm
sip-trunk.sample
credentials number +492281844672 username xxxx password xxxx realm sip-trunk.sample
authentication username xxxx password xxxx realm sip-trunk.sample
no remote-party-id
timers expires 900000
timers register 100
sip-server dns:sip-trunk.sample:5061
connection-reuse
asserted-id pai
bind control source-interface GigabitEthernet0/0/1
bind media source-interface GigabitEthernet0/0/1
conn-reuse
sip-profiles 3000
outbound-proxy dns:reg.sip-trunk.sample
privacy-policy passthru
call-route p-called-party-id
midcall-signaling preserve-codec
```

In the following configuration, the voice class tenant configured in the previous example is part of the dial-peer on the SIP trunk.

```
dial-peer voice 201 voip
description **SIP-TRUNK.SAMPLE**
session protocol sipv2
session target sip-server
session transport tcp tls
destination e164-pattern-map 201
incoming uri from 201
voice-class codec 1
voice-class sip url sips
voice-class sip asserted-id pai
voice-class sip outbound-proxy dns:reg.sip-trunk.sample
voice-class sip tenant 1
voice-class sip srtp-crypto 1
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
srtp
fax-relay ecm disable
fax rate 14400
```

```
ip qos dscp cs6 signaling
clid strip name
no vad
```

The following example provides the show running-config command output for the co-located deployment of Unified SRST and Unified Border Element:

```
Building configuration...
Current configuration : 15564 bytes
!
! Last configuration change at 17:52:50 IST Tue Jul 4 2017
! NVRAM config last updated at 17:52:54 IST Tue Jul 4 2017
!
version 16.7
service timestamps debug datetime msec
service timestamps log datetime msec
service sequence-numbers
platform qfp utilization monitor load 80
no platform punt-keepalive disable-kernel-core
platform shell
platform trace runtime slot F0 bay 0 process forwarding-manager module aom level debug
platform trace runtime slot F0 bay 0 process forwarding-manager module dsp level verbose
platform trace runtime slot F0 bay 0 process forwarding-manager module sbc level debug
platform trace runtime slot R0 bay 0 process forwarding-manager module dsp level verbose
platform trace runtime slot R0 bay 0 process forwarding-manager module om level debug
platform trace runtime slot R0 bay 0 process forwarding-manager module sbc level debug
!
hostname be4k-technium
!
boot-start-marker
boot-end-marker
!
!
vrf definition Mgmt-intf
!
address-family ipv4
exit-address-family
!
address-family ipv6
exit-address-family
!
! card type command needed for slot/bay 0/1
no logging queue-limit
logging buffered 100000000
no logging rate-limit
no logging console
!
no aaa new-model
process cpu statistics limit entry-percentage 10 size 7200
clock timezone IST 5 30
!
!
!
ip host gauss-lnx.cisco.com 10.64.86.64
ip name-server 8.41.20.1
ip dhcp excluded-address 8.39.23.13 8.39.23.50
!
ip dhcp pool phones
network 8.39.0.0 255.255.0.0
default-router 8.39.23.13
domain-name cisco.com
dns-server 8.39.23.13
!
!
!
```

```
!
!
!
!
!
!
!
subscriber templating
!
!
!
!
!
!
!
multilink bundle-name authenticated
!
!
!
!
!
!
trunk group 1
xsvc
!
!
crypto pki trustpoint sipgw1
enrollment url http://8.41.20.1:80
serial-number
ip-address 8.39.23.13
subject-name CN=sipgw1
revocation-check crl
rsakeypair cisco123
!
!
crypto pki certificate chain sipgw1
certificate 02
30820234 3082019D A0030201 02020102 300D0609 2A864886 F70D0101 05050030
13311130 0F060355 04031308 63617365 72766572 301E170D 31373036 32383134
32393330 5A170D31 38303632 38313432 3933305A 305C310F 300D0603 55040313
06736970 67773131 49301206 03550405 130B4644 4F323031 31413132 33301706
092A8648 86F70D01 0908130A 382E3339 2E32332E 3133301A 06092A86 4886F70D
01090216 0D626534 6B2D7465 63686E69 756D3081 9F300D06 092A8648 86F70D01
01010500 03818D00 30818902 818100B5 3CE45902 52517DBE E735F0B5 9D6A412F
FBF398A8 F306F28F A4C79A41 198A19D7 06025696 F5EC6237 EFCB1BBD C7430263
1D0D3C7E AF06B4B2 0D30547C F049A3CD CC4FCFA1 335DA8C5 602A2D18 F91ECC32
E0A7E279 60945941 DF5B53F9 102B9067 8782C1E0 874D6CBC DB0CDA82 C64B7423
E56C5C33 2E13C729 9AB7FEEA 068E7102 03010001 A34F304D 300B0603 551D0F04
04030205 A0301F06 03551D23 04183016 8014265B 6595680C E517CC42 F54AE9EC
1F328FBE BF33301D 0603551D 0E041604 14BA096E DE4E2289 12E8F4D8 95E06E4A
F93876E7 96300D06 092A8648 86F70D01 01050500 03818100 9B172FF6 291C193A
E505ABE9 45AC3202 621BBE2B 6BA45F19 AE0DA7A0 EF5FBC19 5197094E 7A50BCF3
CC49656E A0D991AC FED14749 EAB50892 0239E39C 345ED555 7CD74760 66B0DF49
7E26B654 B8F9E1B1 72FD4039 8A13C9AC EBE75F21 B457D8E3 24BA70E3 F1B3A0C9
5C3153FA B3C744B7 D81F706F B836617F 9E95AD51 813F20AD
quit
certificate ca 01
308201FF 30820168 A0030201 02020101 300D0609 2A864886 F70D0101 04050030
13311130 0F060355 04031308 63617365 72766572 301E170D 31373036 32383134
32383131 5A170D32 30303632 37313432 3831315A 30133111 300F0603 55040313
08636173 65727665 7230819F 300D0609 2A864886 F70D0101 01050003 818D0030
81890281 8100A3AC A4003239 62667AB4 6E8ACE2B 90672DD8 1E2A2952 AFC8A1F6
D56173C9 269F9176 747E93D1 6F699B6F 0C2E600D 8C864F27 4379ED8A E88187F7
17A77C63 B87B7EF6 1556D949 43C743F6 01D9941D 946FCEC8 880B342C 97CC9CEA
```

```
9F015EAC A667F30B 505281AA 29EB10A3 F1C75A99 2A224653 F3B985DD F17BC8DD
40C8C609 62C90203 010001A3 63306130 0F060355 1D130101 FF040530 030101FF
300E0603 551D0F01 01FF0404 03020186 301F0603 551D2304 18301680 14265B65
95680CE5 17CC42F5 4AE9EC1F 328FBEBF 33301D06 03551D0E 04160414 265B6595
680CE517 CC42F54A E9EC1F32 8FBEBF33 300D0609 2A864886 F70D0101 04050003
81810077 C36A6C9A B7C18856 EBDA4504 C38565F0 CF6385EE 29AFC38B 8B90C741
B20C8C36 E979FD72 7B849B34 0BBE3EFA 191E1776 C28FDCF8 5D5F7CFF 170CF615
B4105ABD CD6E0318 4B576FFD 44D115FF 2817E279 78B2794E 577F694F DD129820
B500BB08 E57BFAA9 87835645 4EA53352 B80B51AD 2CC0633A AB9974EB E523A944 0EC230
quit
!
!
!
!
voice service voip
ip address trusted list
ipv4 8.55.0.0 255.255.0.0
ipv4 10.64.0.0 255.255.0.0
address-hiding
mode border-element license capacity 50
media statistics
media bulk-stats
media disable-detailed-stats
allow-connections sip to sip
no supplementary-service sip moved-temporarily
no supplementary-service sip refer
supplementary-service media-renegotiate
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
sip
registrar server expires max 240 min 60
!
!
voice class uri 101 sip
host ipv4:10.64.86.136
!
voice class uri 201 sip
host dns:sip-trunk.sample
!
voice class uri 301 sip
host ipv4:10.64.86.138
voice class codec 1
codec preference 1 g711alaw
codec preference 2 g722-64
codec preference 3 g711ulaw
!
!
voice class sip-profiles 3000
rule 1 request REGISTER sip-header SIP-Req-URI modify "sips:(.*)" "sip:\1"
rule 2 request REGISTER sip-header To modify "<sips:(.*)" "<sip:\1"
rule 3 request REGISTER sip-header From modify "<sips:(.*)" "<sip:\1"
rule 4 request REGISTER sip-header Contact modify "<.*:.*@(.*)>"
"<sip:\1;transport=tls;bnc>"
rule 6 request REGISTER sip-header Proxy-Require add "Proxy-Require: gin"
rule 7 request REGISTER sip-header Require add "Require: gin"
!
voice class sip-profiles 201
rule 1 request ANY sip-header P-Asserted-Identity modify "<sips:(.*)>"
"<sip:+4922842293220@sip-trunk.sample>"
rule 2 request ANY sip-header SIP-Req-URI modify "sips:(.*)" "sip:\1"
rule 3 request ANY sip-header To modify "<sips:(.*)" "<sip:\1"
rule 4 request ANY sip-header From modify "<sips:(.*)" "<sip:\1"
rule 5 request ANY sip-header Contact modify "<sips:(.*)>" "<sip:\1;transport=tls>"
rule 6 response ANY sip-header To modify "<sips:(.*)" "<sip:\1"
rule 7 response ANY sip-header From modify "<sips:(.*)" "<sip:\1"
rule 8 response ANY sip-header Contact modify "<sips:(.*)>" "<sip:\1;transport=tls>"
rule 9 request ANY sip-header Min-SE remove
rule 10 request ANY sip-header Diversion remove
rule 11 request ANY sdp-header Connection-Info remove
rule 12 response ANY sdp-header Connection-Info remove
rule 13 request INVITE sip-header Allow-Header modify "INFO," ""
!
voice class sip-profiles 101
rule 1 request INVITE sip-header Supported modify "100rel," ""
!
voice class sip-profiles 102
rule 1 request INVITE sip-header Privacy add "Privacy:id"
rule 2 request INVITE sip-header P-Called-Party-ID add "P-Called-Party-ID:
sip:2001@10.64.86.64"
!
!
voice class sip-copylist 201
sip-header FROM
!
voice class e164-pattern-map 101
e164 +492284229322T
!
!
voice class e164-pattern-map 201
e164 11[02]
e164 11[68]T
e164 11[025]
e164 +T
e164 0T
e164 2...
!
!
voice class e164-pattern-map 301
e164 3...
!
!
voice class dpg 201
!
voice class dpg 101
dial-peer 201
!
voice class dpg 301
dial-peer 301
!
voice class server-group 1
ipv4 10.64.86.136
description **CUCM Server Group**
!
voice class sip-options-keepalive 101
up-interval 30
retry 3
transport tcp
sip-profiles 3000
!
voice class tenant 1
registrar dns:sip-trunk.sample:5061 scheme sips expires 240 tcp tls auth-realm
sip-trunk.sample
credentials number +492281844672 username xxxx password 7 060506324F41 realm
sip-trunk.sample
authentication username xxxx password 7 121A0C041104 realm sip-trunk.sample
no remote-party-id
timers expires 60000
timers register 100
timers buffer-invite 1000
timers dns registrar-cache ttl
sip-server dns:sip-trunk.sample:5061
connection-reuse
asserted-id pai
bind control source-interface GigabitEthernet0/0/1
bind media source-interface GigabitEthernet0/0/1
no pass-thru content custom-sdp
conn-reuse
sip-profiles 3000
outbound-proxy dns:reg.sip-trunk.sample
privacy-policy passthru
call-route p-called-party-id
midcall-signaling preserve-codec
!
voice class tenant 2
registrar dns:sip-trunk.sample:5060 expires 240 tcp auth-realm sip-trunk.sample
credentials number +492281844673 username xxxx password 7 030752180500 realm
sip-trunk.sample
authentication username xxxx password 7 121A0C041104 realm sip-trunk.sample
no remote-party-id
timers expires 900000
timers register 100
timers buffer-invite 10000
timers dns registrar-cache ttl
sip-server dns:sip-trunk.sample:5060
connection-reuse
asserted-id pai
bind control source-interface GigabitEthernet0/0/1
bind media source-interface GigabitEthernet0/0/1
no pass-thru content custom-sdp
conn-reuse
sip-profiles 3000
outbound-proxy dns:reg.sip-trunk.sample
privacy-policy passthru
call-route p-called-party-id
midcall-signaling preserve-codec
!
voice class tenant 3
registrar dns:sipp.sample:6600 expires 240 auth-realm sip-trunk.sample
credentials number +492281844672 username xxxx password 7 121A0C041104 realm
sip-trunk.sample
authentication username xxxx password 7 05080F1C2243 realm sip-trunk.sample
no remote-party-id
timers expires 900000
timers register 500
timers buffer-invite 1000
timers dns registrar-cache ttl
sip-server dns:sipp.sample
connection-reuse
asserted-id pai
bind control source-interface GigabitEthernet0/0/1
bind media source-interface GigabitEthernet0/0/1
no pass-thru content custom-sdp
conn-reuse
sip-profiles 3000
outbound-proxy dns:sipp.sample:6600
privacy-policy passthru
call-route p-called-party-id
midcall-signaling preserve-codec
!
voice class tenant 4
timers expires 60000
timers buffer-invite 10000
connection-reuse
```

```
asserted-id pai
bind control source-interface GigabitEthernet0/0/0
bind media source-interface GigabitEthernet0/0/0
no pass-thru content custom-sdp
privacy-policy passthru
call-route p-called-party-id
midcall-signaling preserve-codec
!
voice class srtp-crypto 1
crypto 1 AES_CM_128_HMAC_SHA1_32
crypto 2 AES_CM_128_HMAC_SHA1_80
!
!
!
voice register global
default mode
no allow-hash-in-dn
max-dn 40
max-pool 40
!
voice register pool 1
id network 8.55.0.0 mask 255.255.0.0
dtmf-relay rtp-nte
voice-class codec 1
!
voice hunt-group 1 parallel
list 1001,1002,1003
timeout 15
statistics collect
pilot 1234
!
!
voice hunt-group 2 sequential
list 1002,1003,1004
timeout 5
statistics collect
pilot 2345
!
!
!
!
!
!
voice-card 0/1
dsp services dspfarm
no watchdog
!
license udi pid ISR4321/K9 sn FDO201115PV
license boot level uck9
license boot level securityk9
no license smart enable
diagnostic bootup level minimal
!
spanning-tree extend system-id
!
!
!
username xxxx privilege 15 password 0 cisco
username xxxx password 0 cisco
!
redundancy
mode none
!
!
```

```
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
template 1
!
!
!
!
!
interface GigabitEthernet0/0/0
ip address 8.39.23.13 255.255.0.0
ip nat inside
media-type rj45
negotiation auto
!
interface GigabitEthernet0/0/1
ip address 10.64.86.64 255.255.0.0
ip nat outside
negotiation auto
!
interface Service-Engine0/1/0
!
interface GigabitEthernet0
vrf forwarding Mgmt-intf
no ip address
negotiation auto
!
no ip nat service sip tcp port 5060
no ip nat service sip udp port 5060
ip nat pool pool1 8.39.0.0 8.39.255.255 netmask 255.255.0.0
ip nat inside source list 100 interface GigabitEthernet0/0/1 overload
ip forward-protocol nd
ip http server
no ip http secure-server
ip tftp source-interface GigabitEthernet0/0/0
ip tftp blocksize 1520
ip rtcp report interval 9000
ip route 0.0.0.0 0.0.0.0 8.39.0.1
ip route 10.0.0.0 255.0.0.0 10.64.86.1
!
ip ssh server algorithm encryption aes128-ctr aes192-ctr aes256-ctr
ip ssh client algorithm encryption aes128-ctr aes192-ctr aes256-ctr
!
!
ip access-list extended nat-list
access-list 100 permit ip 8.39.23.0 0.0.0.255 any
!
!
tftp-server flash:fbi88xx.BE-01-010.sbn
```

```
tftp-server flash:kern88xx.12-0-1MN-113.sbn
tftp-server flash:rootfs88xx.12-0-1MN-113.sbn
tftp-server flash:sb288xx.BE-01-020.sbn
tftp-server flash:sip88xx.12-0-1MN-113.loads
tftp-server flash:vc488xx.12-0-1MN-113.sbn
!
!
ipv6 access-list preauth_v6
permit udp any any eq domain
permit tcp any any eq domain
permit icmp any any nd-ns
permit icmp any any nd-na
permit icmp any any router-solicitation
permit icmp any any router-advertisement
permit icmp any any redirect
permit udp any eq 547 any eq 546
permit udp any eq 546 any eq 547
deny ipv6 any any
!
control-plane
!
!
voip trunk group 1
xsvc
!
uc wsapi
message-exchange max-failures 99
response-timeout 2
source-address 8.39.23.13
probing interval keepalive 60
probing max-failures 2
provider xcc
remote-url http://8.39.23.13:8090/xcc
!
!
provider xsvc
!
!
!
mgcp behavior rsip-range tgcp-only
mgcp behavior comedia-role none
mgcp behavior comedia-check-media-src disable
mgcp behavior comedia-sdp-force disable
!
mgcp profile default
!
!
!
!
dial-peer voice 201 voip
description **SIP-TRUNK.SAMPLE**
session protocol sipv2
session target sip-server
session transport tcp tls
destination e164-pattern-map 201
incoming uri from 201
voice-class codec 1
voice-class sip url sips
voice-class sip profiles 201
voice-class sip tenant 1
voice-class sip srtp-crypto 1
dtmf-relay rtp-nte
srtp
fax-relay ecm disable
```

```
fax rate 14400
clid strip name
no vad
!
dial-peer voice 301 voip
description **SIP-TRUNK.SAMPLE**
session protocol sipv2
session target sip-server
session transport tcp
destination e164-pattern-map 301
incoming uri from 201
voice-class codec 1
voice-class sip url sip
voice-class sip profiles 201
voice-class sip tenant 2
dtmf-relay rtp-nte
srtp fallback
fax-relay ecm disable
fax rate 14400
clid strip name
no vad
!
dial-peer voice 401 voip
description **SIP-TRUNK.SAMPLE**
destination-pattern 4...
session protocol sipv2
session target sip-server
session transport udp
incoming uri from 301
voice-class codec 1
voice-class sip url sip
voice-class sip profiles 201
voice-class sip tenant 3
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate 14400
clid strip name
no vad
!
dial-peer voice 101 voip
description **CUCM/PBX**
destination-pattern .T
session protocol sipv2
session transport tcp
session server-group 1
destination dpg 101
incoming uri via 101
voice-class codec 1
no voice-class sip outbound-proxy
voice-class sip srtp negotiate cisco
voice-class sip profiles 102 inbound
voice-class sip tenant 4
voice-class sip srtp-crypto 1
voice-class sip options-keepalive profile 101
dtmf-relay rtp-nte
srtp fallback
fax-relay ecm disable
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback pass-through g711alaw
no vad
!
!
presence
!
```

```
gateway
media-inactivity-criteria all
timer receive-rtcp 5
timer receive-rtp 180
!
sip-ua
transport tcp tls v1.2
crypto signaling default trustpoint sipgw1
!
alias exec cl clear logg
alias exec rtp show voip rtp connections
alias exec pool show voice register pool all brief
!
line con 0
exec-timeout 0 0
password cisco
width 0
transport input none
stopbits 1
line aux 0
stopbits 1
line vty 0 4
exec-timeout 0 0
password cisco
login local
length 0
transport input all
!
!
!
!
!
!
end
```

Feature Information for Configuring SIP Trunking on Cisco Unified SRST

Not all commands may be available in your Cisco IOS Software release. For release information about specific command, see the command reference documentation.

Use Cisco Feature Navigator to find information about the platform support and software image support. Cisco Feature Navigator enables you to determine which Cisco IOS and Cisco Catalyst operating system software images support a specific software release, feature set, or platform. To access Cisco Feature Navigator, go to http://www.cisco.com/go/cfn . You do not need an account on Cisco.com.

Note The table lists only the Cisco IOS Software release that introduced support for a given feature in a given Cisco IOS Software release train. Unless noted otherwise, subsequent releases of that Cisco IOS Software release train also support that feature.

The following table lists the release history for this feature.

Feature Name	Releases	Feature Information
Cisco Unified SRST and Cisco Unified Border Element Co-location	Cisco IOS XE Fuji 16.7.1	Added Support for co-location of Cisco Unified SRST and Cisco Unified Border Element on Cisco 4000 Series Integrated Services Router.

CHAPTER 12

Integrating Voice Mail with Cisco Unified SRST

This chapter describes how to make your existing voicemail system run on phones connected to a Cisco Unified SRST router during Cisco Unified Communications Manager fallback.

Cisco Unified SRST also supports incoming and outgoing Session Initiation Protocol (SIP) calls to and from Cisco Unified IP phones and router voice gateway voice ports. SIP may be used in situations where the Cisco Unified SRST Router is separate from the PSTN gateway and the SRST and PSTN gateways are linked together using SIP (instead of H.323).

For more information about SIP, see Cisco IOS SIP Configuration Guide.

Information About Integrating Voicemail with Cisco Unified SRST

Cisco Unified SRST can send and receive voicemail messages from Cisco Unity and other voicemail systems during Cisco Unified CM fallback. When the WAN is down, a voicemail system with BRI or PRI access to the Cisco Unified SRST system uses ISDN signaling (see figure 5 - Cisco Unified Communications Manager Fallback with BRI or PRI). Systems with Foreign Exchange Office (FXO) or Foreign Exchange Station (FXS) access connect to a PSTN and use in-band dual tone multifrequency (DTMF) signaling (see figure 6 - Cisco Unified Communications Manager Fallback with PSTN).

From Unified SRST Release 12.0 onwards, Unified SRST supports voicemail on IPv6 protocols for SIP IP phones.

Figure 5: Cisco Unified Communications Manager Fallback with BRI or PRI

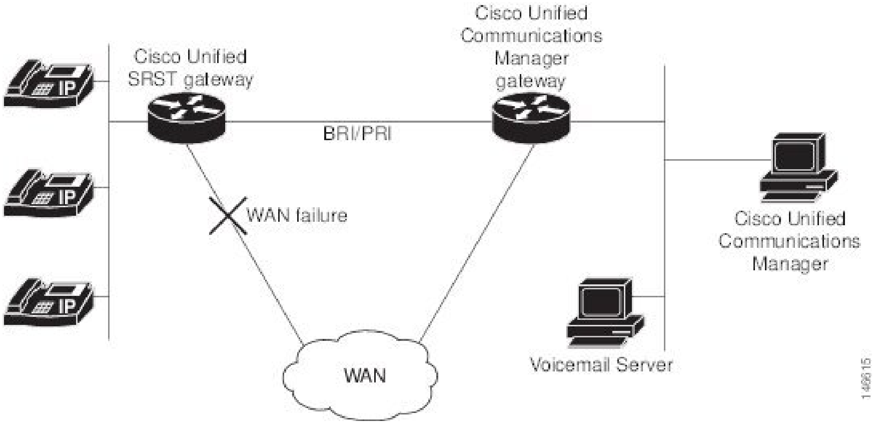
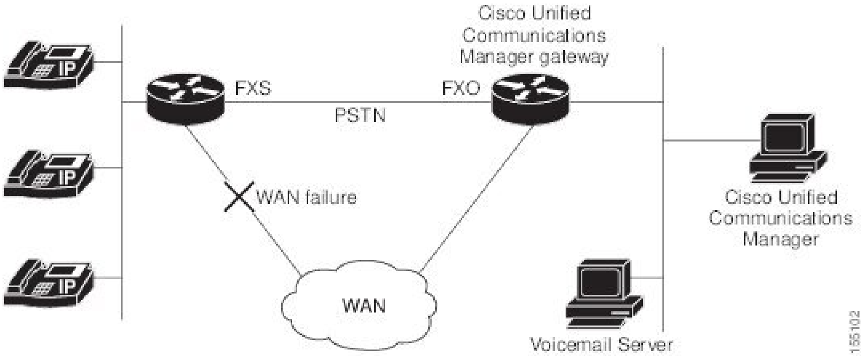

Cisco Unified SRST gateway
Cisco Unified Communications Manager gateway
BRI/PRI
WAN failure
Cisco Unified Communications Manager
Voicemail Server
WAN
146615

Figure 6: Cisco Unified Communications Manager Fallback with PSTN


Cisco Unified Communications Manager gateway
FXS
FXO
PSTN
WAN failure
Cisco Unified Communications Manager
Voicemail Server
WAN
155102

Both configurations allow phone message buttons to remain active and calls to busy or unanswered numbers to be forwarded to the dialed numbers' mailboxes.

Calls that reach a busy signal, calls that are unanswered, and calls made by pressing the message button are forwarded to the voicemail system. To make this happen, you must configure access from the dial peers to the voicemail system and establish routing to the voicemail system for busy and unanswered calls and for message buttons.

If the voicemail system is accessed over FXO or FXS, you must configure instructions (DTMF patterns) for the voicemail system so that it can access the correct voicemail system mailbox. If your voicemail system is accessed over BRI or PRI, no instructions are necessary because the voicemail system can log in to the calling phone's mailbox directly.

How to Integrate Voicemail with Cisco Unified SCCP and SIP SRST

This section contains the following tasks:

Note Support for SIP SRST is added from IOS release 15.1(4)M3 and 15.2(1)T2.

Configuring Direct Access to Voicemail

You can configure direct access to voicemail system using BRI/PRI or FXO/FXS. To access voicemail messages with BRI/PRI or FXO/FXS access, you must have POTS dial peers configured with a destination pattern that matches the voicemail system's number. Also, you must associate the dial peer with the port to which the voicemail system is accessed.

Both sets of configurations are done in dial-peer configuration mode. The summary and detailed steps below include only the basic commands necessary to perform this task. You may require additional commands for your particular dial-peer configuration.

Table 3: Valid Entries for the String Argument in the destination-pattern command

Entry	Description
Digits 0 to 9	—
Letters A through D	—
Asterisk (*) and pound sign (#)	These appear on standard touch-tone dial pads.
Comma (,)	Inserts a pause between digits.
Period (.)	Indicates that the preceding digit occurred zero or more times; similar to the wildcard usage.
Percent sign (%)	Indicates that the preceding digit occurred zero or more times; similar to the wildcard usage.
Plus sign (+)	Indicates that the preceding digit occurred one or more times. **Note** The plus sign used as part of a digit string is different from the plus sign that can be used in front of a digit string to indicate that the string is an E.164 standard number.
Circumflex (^)	Indicates a match to the beginning of the string. Parentheses (( )), which indicate a pattern and are the same as the regular expression rule.
Dollar sign ($)	Matches the null string at the end of the input string.
Backslash symbol (\)	Is followed by a single character and matches that character. Can be used with a single character with no other significance (matching that character).
Question mark (?)	Indicates that the preceding digit occurred zero or one time.
Brackets ([ ])	Indicates a range. A range is a sequence of characters enclosed in the brackets; only numeric characters from 0 to 9 are allowed in the range.

SUMMARY STEPS

1. **dial-peer voice** *tag* {**pots** |**voatm** | **vofr** | **voip**}
2. **destination-pattern** [+]*string*[**T**]
3. **port**{*slot-number/subunit-number/port* |*slot/port*:*ds0-group-no*}
4. **forward-digits** {*num-digit* |**all** | **extra**}
5. Do the following to configure a video codec:
   - **video codec** *codec*
6. **exit**

DETAILED STEPS

	Command or Action	Purpose

Step 1	**dial-peer voice** *tag* {**pots** |**voatm** | **vofr** | **voip**}
	Example:
	`Router(config)# dial-peer voice 1002 pots`
		(FXO or FXS and BRI or PRI) Defines a particular dial peer, specifies the method of voice encapsulation, and enters dial-peer configuration mode. The **dial-peer** command provides different syntax for individual routers. This example is syntax for Cisco 3600 series routers.
		• **tag**: Digits that define a particular dial peer. Range is from 1 to 2147483647.
		• **pots**: Indicates that this is a POTS dial peer that uses VoIP encapsulation on the IP backbone.
		• **voatm**: Specifies that this is a VoATM dial peer that uses real-time AAL5 voice encapsulation on the ATM backbone network.
		• **vofr**: Specifies that this is a VoFR dial peer that uses FRF.11 encapsulation on the Frame Relay backbone network.
		• **voip**: Indicates that this is a VoIP dial peer that uses voice encapsulation on the POTS network.

Step 2	**destination-pattern** [+]*string*[**T**] **Example:** `Router(config-dial-peer)# destination-pattern 1100T`	(FXO or FXS and BRI or PRI) Specifies either the prefix or the full E.164 telephone number (depending on your dial plan) to be used for a dial peer. • **+**: (Optional) Character that indicates an E.164 standard number. • *string*: See Table *Valid Entries for the String Argument in the destination-pattern command*. • **T**: (Optional) Control character that indicates that the destination-pattern value is a variable-length dial string.
Step 3	**port**{*slot-number/subunit-number/port* |*slot/port*:*ds0-group-no*} **Example:**	(FXO or FXS and BRI or PRI) Associates a dial peer with a specific voice port on Cisco routers.

	Command or Action	Purpose
	`Router(config-dial-peer)# port 1/1/1`	• **slot-number**: Number of the slot in the router in which the voice interface card (VIC) is installed. Valid entries are from 0 to 3, depending on the slot in which it is installed. • **subunit-number**: Subunit on the VIC in which the voice port is located. Valid entries are 0 or 1. • **port**: Voice port number. Valid entries are 0 and 1. • **ds0-group-no**: Specifies the DS0 group number. Each defined DS0 group number is represented on a separate voice port. This allows you to define individual DS0s on the digital T1/E1 card.

Step 4	**forward-digits** {*num-digit* |**all** | **extra**}
	Example:
	`Router(config-dial-peer)# forward-digits all`
		(Optional for FXO or FXS) Specifies which digits to forward for voice calls.
		• **num-digit**: The number of digits to be forwarded. If the number of digits is greater than the length of a destination phone number, the length of the destination number is used. Range is 0 to 32. Setting the value to 0 is equivalent to entering the**no forward-digits** command.
		• **all**: Forwards all digits. If **all** is entered, the full length of the destination pattern is used.
		• **extra**: If the length of the dialed digit string is greater than the length of the dial-peer destination pattern, the extra right-justified digits are forwarded. However, if the dial-peer destination pattern is variable length and ends with the character "T" (for example: T, 123T, 123...T), extra digits are not forwarded.

Step 5	Do the following to configure a video codec: • **video codec** *codec* **Example:** For Video Codec `Device(config-dial-peer)# video codec h261`	Configures a video codec at the dial peer level.
Step 6	**exit** **Example:** `Router(config-dial-peer)# exit`	(FXO or FXS and BRI or PRI) Exits dial-peer configuration mode.

Examples

The following FXO and FXS example sets up a POTS dial peer named 1102, matches dial-peer 1102 to voicemail extension 1101, and assigns dial-peer 1102 to voice-port 1/1/1 where the voicemail system is connected. Other dial peers are configured for direct access to voicemail.

```
voice-port 1/1/1
timing digit 250
timing inter-digit 250
dial-peer voice 1102 pots
destination-pattern 1101
port 1/1/1
forward-digits all
dial-peer voice 1103 pots
destination-pattern 1101
port 1/1/1
forward-digits all
dial-peer voice 1104 pots
destination-pattern 1101
port 1/1/1
forward-digits all
```

The following example sets up a POTS dial peer named 1102 to go directly to 1101 through port 2/0:23:

```
controller T1 2/0
framing esf
clock source line primary
linecode b8zs
cablelength short 133
pri-group timeslots 21-24
interface Serial2/0:23
no ip address
no logging event link-status
isdn switch-type primary-net5
isdn incoming-voice voice
isdn T309-enable
no cdp enable
voice-port 2/0:23
dial-peer voice 1102 pots
destination-pattern 1101T
port 2/0:23
```

Configuring Message Buttons

To activate the message buttons on Cisco Unified IP phones connected to the Cisco Unified SCCP and SIP SRST router during Cisco Unified Communications Manager fallback, you must program a speed-dial number to the voicemail system. The speed-dial number is dialed when message buttons on phones connected to the Cisco Unified SCCP and SIP SRST router are pressed during Cisco Unified CM fallback. In addition, call forwarding must be configured so that calls to busy and unanswered numbers are sent to the voicemail number.

This configuration is required for FXO or FXS and BRI or PRI.

SUMMARY STEPS

1. **call-manager-fallback**
2. **voicemail** *phone-number*

3. **call-forward busy** *directory-number*
4. **call-forward noan** *directory-number* **timeout** *seconds*
5. **exit**
6. **voice register pool** *tag*
7. **call-forward b2bua busy** *directory-number*
8. **call-forward b2bua noan** *directory-number***timeout** *seconds*
9. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**call-manager-fallback** **Example:** `Router(config)# call-manager-fallback`	Enters call-manager-fallback configuration mode.
Step 2	**voicemail** *phone-number* **Example:** `Router(config-cm-fallback)# voicemail 5550100`	Configures the telephone number that is dialed when the message button on a Cisco Unified SCCP IP Phone is pressed. *phone-number* : Phone number configured as a speed-dial number for retrieving messages.
Step 3	**call-forward busy** *directory-number* **Example:** `Router(config-cm-fallback)# call-forward busy 2000`	Configures call forwarding to another number when the Cisco SCCP IP phone is busy. *directory-number* : Selected directory number representing a fully qualified E.164 number. This number can contain "." wildcard characters that correspond to the right-justified digits in the directory number extension.
Step 4	**call-forward noan** *directory-number* **timeout** *seconds* **Example:** `Router(config-cm-fallback)# call-forward noan 2000 timeout 10`	Configures call forwarding to another number when no answer is received from the Cisco SCCP IP phone. *directory-number* : Selected directory number representing a fully qualified E.164 number. This number can contain "." wildcard characters that correspond to the right-justified digits in the directory number extension. **timeout** *seconds* : Sets the waiting time, in seconds, before the call is forwarded to another phone. The seconds range is from 3 to 60000.
Step 5	**exit** **Example:** `Router(config-cm-fallback)# exit`	Exits call-manager-fallback configuration mode.
Step 6	**voice register pool** *tag* **Example:** `Router(config)# voice register pool 1`	Enters voice register pool configuration mode.

	Command or Action	Purpose
Step 7	**call-forward b2bua busy** *directory-number* **Example:** `Router(config-register-pool)# call-forward b2bua busy 2000`	Configures call forwarding to another number when the Cisco SIP IP phone is busy. *directory-number* : Selected directory number representing a fully qualified E.164 number. This number can contain "." wildcard characters that correspond to the right-justified digits in the directory number extension.
Step 8	**call-forward b2bua noan** *directory-number***timeout** *seconds* **Example:** `Router(config-register-pool)# call-forward noan 2000 timeout 10`	Configures call forwarding to another number when no answer is received from the Cisco SIP IP phone. *directory-number* : Selected directory number representing a fully qualified E.164 number. This number can contain "." wildcard characters that correspond to the right-justified digits in the directory number extension. **timeout** *seconds* : Sets the waiting time, in seconds, before the call is forwarded to another phone. The seconds range is from 3 to 60000.
Step 9	**exit** **Example:** `Router(config-register-pool)# exit`	Exits voice register pool configuration mode.

Examples

The following example specifies 1101 as the speed-dial number that is issued when message buttons are pressed on Cisco Unified IP Phones connected to the Cisco Unified SRST router. All busy and unanswered calls are configured to be forwarded to the voicemail number (1101).

```
call-manager-fallback
voicemail 1101
call-forward busy 1101
call-forward noan 1101 timeout 3
voice register pool 1
call-forward b2bua busy 1101
call-forward b2bua noan 1101 timeout 3
```

Redirecting to Cisco Unified Communications Manager Gateway

Before you begin

Note The following task is required for voicemail systems with BRI or PRI access.

In addition to supporting message buttons for retrieving personal messages, Cisco Unified SRST allows the automatic forwarding of calls to busy and unanswered numbers to voicemail systems. Voicemail systems with BRI or PRI access can log in to the calling phone's mailbox directly. For this to happen, some Cisco Unified CM configuration is recommended. If your voicemail system supports Redirected Dialed Number identification Service (RDNIS), RDNIS must be included in the outgoing SETUP message to Cisco Unified

CM to declare the last redirected number and the originally dialed number to and from configured devices and applications.

SUMMARY STEPS

1. From any page in Cisco Unified CM, click **Device** and **Gateway**
2. From the Find and List Gateways page, click **Find**.
3. From the Find and List Gateways page, choose a device name.
4. From the Gateway Configuration page, check **Redirecting Number IE Delivery - Outgoing**.

DETAILED STEPS

	Command or Action	Purpose
Step 1	From any page in Cisco Unified CM, click **Device** and **Gateway**	
Step 2	From the Find and List Gateways page, click **Find**.	
Step 3	From the Find and List Gateways page, choose a device name.	
Step 4	From the Gateway Configuration page, check **Redirecting Number IE Delivery - Outgoing**.	

Configuring Call Forwarding to Voicemail

Note The following task is required for voicemail systems with FXO or FXS access.

In addition to supporting message buttons for retrieving personal messages, Cisco Unified SRST allows the automatic forwarding of calls to busy or unanswered numbers to voicemail systems. The forwarded calls can be routed to almost any location in the voicemail system. Typically, calls are forwarded to a location in the called number’s mailbox where the caller can leave messages.

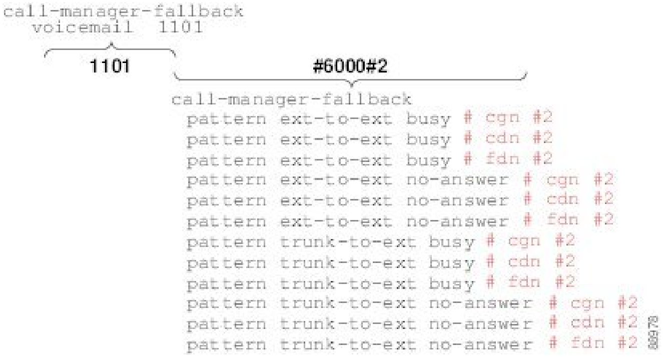
Call Routing Instructions Using DTMF Digit Patterns

Cisco Unified SRST Cisco Unified SRST call-routing instructions are required so that forwarded calls can be sent to the correct voicemail boxes. These instructions consist of DTMF digits configured in patterns that match the dial sequences required by the voicemail system to get to a particular voicemail location. For example, a voicemail system may be designed so that callers must do the following to leave a message:

1. Dial the central voicemail number (1101) and press #.
2. Dial an extension number (6000) and press #.
3. Dial 2 to select the menu option for leaving messages in the extension number’s mailbox.

For Cisco Unified SRST to forward a call to a busy or unanswered number to extension 6000’s mailbox, it must be programmed to issue a sequence of 1101#6000#2. As shown in the below figure, this is accomplished through the **voicemail** and **pattern** commands.

Figure 7: How Voicemail Dial Sequence 1101#6000#2 Is Configured in Cisco Unified SRST



```
call-manager-fallback
  voicemail  1101
```

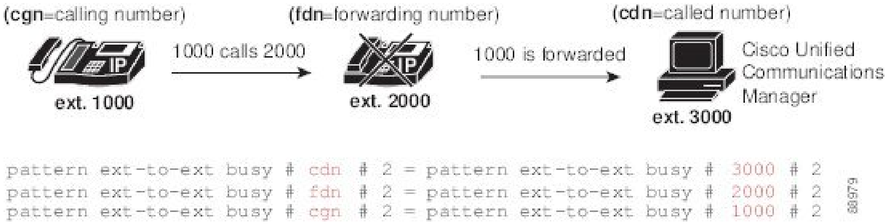
```
call-manager-fallback
 pattern ext-to-ext busy # cgn #2
 pattern ext-to-ext busy # cdn #2
 pattern ext-to-ext busy # fdn #2
 pattern ext-to-ext no-answer # cgn #2
 pattern ext-to-ext no-answer # cdn #2
 pattern ext-to-ext no-answer # fdn #2
 pattern trunk-to-ext busy # cgn #2
 pattern trunk-to-ext busy # cdn #2
 pattern trunk-to-ext busy # fdn #2
 pattern trunk-to-ext no-answer # cgn #2
 pattern trunk-to-ext no-answer # cdn #2
 pattern trunk-to-ext no-answer # fdn #2
```

The # cgn #2, # cdn #2, and # fdn #2 portions of the **pattern** commands shown in are DTMF digit patterns. These patterns are composed of tags and tokens. Tags are sets of characters representing DTMF tones. Tokens consist of three command keywords (**cgn**, **cdn**, and **fdn**) that declare the state of an incoming call transferred to voicemail.

A tag can be up to three character from the DTMF tone set (A to D, 0 to 9, # and *). Voicemail systems can use limited sets of DTMF tones. For example, Cisco Unity uses all DTMF tones but A to D. Tones can be defined in multiple ways. For example, when the star (*) is placed in front of a token by itself, it can mean “dial the following token number,” or, if it is at the end of a token, it can mark the end of a token number. If the asterisk is between other tag characters, it can mean dial *. The use of tags depends on how DTMF tones are defined by your voicemail system.

Tokens tell Cisco Unified SRST what telephone number in the call forwarding chain to use in the pattern. As shown in the following figure, there are three types of tokens that correspond to three possible call states during voicemail forwarding.

Figure 8: How Numbers Are Extracted from Tokens



```
pattern ext-to-ext busy # cdn # 2 = pattern ext-to-ext busy # 3000 # 2
pattern ext-to-ext busy # fdn # 2 = pattern ext-to-ext busy # 2000 # 2
pattern ext-to-ext busy # cgn # 2 = pattern ext-to-ext busy # 1000 # 2
```

Sets of tags and tokens or patterns activate a voicemail system when one of the following occurs:

- A user presses the message button on a phone (**pattern direct** command).
- An internal extension attempts to connect to a busy extension and the call is forwarded to voicemail (**pattern ext-to-ext busy** command).
- An internal extension fails to connect to an extension and the call is forwarded to voicemail (**pattern ext-to-ext no-answer** command).
- An external trunk call reaches a busy extension and the call is forwarded to voicemail (**pattern trunk-to-ext busy** command).
- An external trunk call reaches an unanswered extension and the call is forwarded to voicemail (**pattern trunk-to-ext no-answer** command).

Prerequisites

- FXO hairpin-forwarded calls to voicemail systems must have disconnect supervision from the central office. For further information, see the FXO Answer and Disconnect Supervision document.
- To configure patterns that your voicemail system will interpret correctly, you must know how the system routes voicemail calls and interprets DTMF tones (see the Call Routing Instructions Using DTMF Digit Patterns section).

  You can find information about how Cisco Unity handles voicemail calls in the How to Transfer a Caller Directly into a Cisco Unity Mailbox document. Additional call-handling information can be found in the "Subscriber and Operator Orientation" chapters of any Cisco Unity system administration guide.

  For other voicemail systems, see the analog voicemail integration configuration guide or information about the system's call handling.

Configuring Call Forwarding to Voicemail

SUMMARY STEPS

1. **vm-integration**
2. **pattern direct** *tag1* {**CGN** |**CDN** | **FDN**} [*tag2* {**CGN** |**CDN** | **FDN**}] [*tag3* {**CGN** |**CDN** | **FDN**}] [*last-tag*]
3. **pattern ext-to-ext busy** *tag1* {**CGN** | **CDN** | **FDN**} [*tag2* {**CGN** | **CDN** | **FDN**}] [*tag3* {**CGN** | **CDN** | **FDN**}] [*last-tag*]
4. **pattern ext-to-ext no-answer***tag1* {**CGN** | **CDN** | **FDN**} [*tag2* {**CGN** | **CDN** | **FDN**}] [*tag3* {**CGN** | **CDN** | **FDN**}] [*last-tag*]
5. **pattern trunk-to-ext busy***tag1*{**CGN** | **CDN** | **FDN**} [*tag2* {**CGN** | **CDN** | **FDN**}] [*tag3* {**CGN** | **CDN** | **FDN**}] [*last-tag*]
6. **pattern trunk-to-ext no-answer***tag1* {**CGN** | **CDN** | **FDN**} [*tag2* {**CGN** | **CDN** | **FDN**}] [*tag3* {**CGN** | **CDN** | **FDN**}] [*last-tag*]

DETAILED STEPS

	Command or Action	Purpose
Step 1	**vm-integration** **Example:** `Router(config)# vm-integration`	Enters voicemail integration mode and enables voicemail integration with DTMF and analog voicemail systems.
Step 2	**pattern direct** *tag1* {**CGN** |**CDN**	**FDN**} [*tag2* {**CGN** |**CDN**	**FDN**}] [*tag3* {**CGN** |**CDN**	**FDN**}] [*last-tag*] **Example:** `Router(config-vm-int)# pattern direct 2 CGN *`	Configures the DTMF digit pattern forwarding necessary to activate the voicemail system when the user presses the messages button on the phone. • **tag1**: Alphanumeric string fewer than four DTMF digits in length. The alphanumeric string consists of a combination of four letters (A, B, C, and D), two symbols (* and #), and ten digits (0 to 9). The tag numbers match the numbers defined in the voicemail system's integration file, immediately preceding either the number of the calling party, the number of the called party, or a forwarding number.

	Command or Action	Purpose
		• **tag2**and **tag3**: (Optional) See **tag1**. • **last-tag:** See **tag1**. This tag indicates the end of the pattern. • **CGN**: Calling number (CGN) information is sent to the voicemail system. • **CDN**: Called number (CDN) information is sent to the voicemail system. • **FDN**: Forwarding number (FDN) information is sent to the voicemail system.
Step 3	**pattern ext-to-ext busy** *tag1* {**CGN** | **CDN** | **FDN**} [*tag2* {**CGN** | **CDN** | **FDN**}] [*tag3* {**CGN** | **CDN** | **FDN**}] [*last-tag*] **Example:** `Router(config-vm-int)# pattern ext-to-ext busy 7 FDN * CGN *`	Configures the DTMF digit pattern forwarding necessary to activate the voicemail system once an internal extension attempts to connect to a busy extension and the call is forwarded to voicemail.
Step 4	**pattern ext-to-ext no-answer***tag1* {**CGN** | **CDN** | **FDN**} [*tag2* {**CGN** | **CDN** | **FDN**}] [*tag3* {**CGN** | **CDN** | **FDN**}] [*last-tag*] **Example:** `Router(config-vm-int)# pattern ext-to-ext no-answer 5 FDN * CGN *`	Configures the DTMF digit pattern forwarding necessary to activate the voicemail system once an internal extension fails to connect to an extension and the call is forwarded to voicemail. For argument and keyword information, see Step 1.
Step 5	**pattern trunk-to-ext busy***tag1*{**CGN** | **CDN** | **FDN**} [*tag2* {**CGN** | **CDN** | **FDN**}] [*tag3* {**CGN** | **CDN** | **FDN**}] [*last-tag*] **Example:** `Router(config-vm-int)# pattern trunk-to-ext busy 6 FDN * CGN *`	Configures the DTMF digit pattern forwarding necessary to activate the voicemail system once an external trunk call reaches a busy extension and the call is forwarded to voicemail.
Step 6	**pattern trunk-to-ext no-answer***tag1* {**CGN** | **CDN** | **FDN**} [*tag2* {**CGN** | **CDN** | **FDN**}] [*tag3* {**CGN** | **CDN** | **FDN**}] [*last-tag*] **Example:** `Router(config-vm-int)# pattern trunk-to-ext no-answer 4 FDN * CGN *`	Configures the DTMF digit pattern forwarding necessary to activate the voicemail system when an external trunk call reaches an unanswered extension and the call is forwarded to voicemail.

Examples

For the following configuration, if the voicemail number is 1101, and 3001 is a phone with a message button, 1101*3001 would be dialed automatically when the 3001 message button is pressed. Under these circumstances, 3001 is considered to be a calling number or inbound call number.

```
vm-integration
pattern direct * CGN
```

For the following configuration, if 3001 calls 3006 and 3006 does not answer, the Unified SRST router will forward 3001 to the voicemail system (1101) and send to the voicemail system the DTMF pattern # 3006 #2. This pattern is intended to select voicemail box number 3006 (3006's voice mailbox). For this pattern to be sent, 3001 must be a forwarding number.

```
vm-integration
pattern ext-to-ext no-answer # FDN #2
```

For the following configuration, if 3006 is busy and 3001 calls 3006, the Unified SRST router will forward 3001 to the voicemail system (1101) and send to the voicemail system the DTMF pattern # 3006 #2. This pattern is intended to select voice mailbox number 3006 (3006's voice mailbox). For this pattern to be sent, 3001 must be a forwarding number.

```
vm-integration
pattern ext-to-ext busy # FDN #2
```

Configuring Message Waiting Indication (Cisco Unified SRST Routers)

The Message Waiting Indication (MWI) relay mechanism is initiated after someone leaves a voicemail message on the remote voicemail message system. MWI relay is required when one Cisco Unity Voicemail system is shared by multiple Cisco Unified SRST routers. Unified SRST routers use the SIP Subscribe and Notify methods for MWI. See Configuring Cisco IOS SIP Configuration Guide for more information on SIP MWI and the Subscribe and Notify methods. The Unified SRST router that is the SIP MWI relay server acts as the SIP notifier. The other remote routers act as the SIP subscribers.

Restriction

MWI is not supported during a fallback to Unified SRST. The MWI (the phone LED indication) will not correctly reflect when new messages arrive or when all messages have been listened to. We recommend resynchronizing MWIs after the WAN link is available, and connection with Unified Communications Manager is reestablished. The MWI behavior is consistent across voicemail support for IPv4 as well as IPv6 on Unified SRST.

SUMMARY STEPS

1. **call-manager-fallback**
2. **configure terminal**
3. **mwi relay**
4. **mwi reg-e164**
5. **exit**
6. **sip-ua**
7. **mwi-server** {**ipv4:***destination-address* | **dns**:*host-name*}**[expires** *seconds***[port** *port*]**[transport**] {**tcp** | **udp**}] [unsolicited]]
8. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**call-manager-fallback** **Example:** `Router(config)# call-manager-fallback`	Enters call-manager-fallback configuration mode.

	Command or Action	Purpose
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**mwi relay** **Example:** `Router(config-cm-fallback)# mwi relay`	Enables the Unified SRST router to relay MWI information to remote Cisco IP phones.
Step 4	**mwi reg-e164** **Example:** `Router(config-cm-fallback)# mwi reg-e164`	Registers E.164 numbers rather than extension numbers with a SIP proxy or registrar.
Step 5	**exit** **Example:** `Router(config-cm-fallback)# exit`	Exits call-manager-fallback configuration mode.
Step 6	**sip-ua** **Example:** `Router(config)# sip-ua`	Enters SIP user-agent configuration mode.
Step 7	**mwi-server** {**ipv4:***destination-address* | **dns**:*host-name*}**[expires** *seconds***[port** *port*]**[transport**] {**tcp** | **udp**}] [unsolicited]] **Example:** `Router(config-sip-ua)# mwi-server ipv4:10.0.2.254`	Configures voicemail server settings on a voice gateway or user agent. The IP address and port for the SIP-based MWI server should be in the same LAN as the voicemail server. The MWI server is a Cisco Unified SRST router. Keywords and arguments are as follows: • **ipv4:***destination-address*: IP address of the voicemail server. • **dns:***host-name*: The argument should contain the complete hostname to be associated with the target address; for example, dns:test.cisco.com. • **expires** *seconds*: Subscription expiration time, in seconds. Range is from 1 to 999999. Default is 3600. • **port** *port*: Port number on the voicemail server. Default is 5060. • **transport**: Transport protocol to the voicemail server. Valid values are tcp and udp. Default is UDP. • **unsolicited**: Requires the voicemail server to send a SIP notification message to the voice gateway or UA if the mailbox status changes. Removes the requirement that the voice gateway subscribe for MWI service.

	Command or Action	Purpose
Step 8	**exit** **Example:** `Router(config-sip-ua)# exit`	Exits SIP user-agent configuration mode.

Configuring Message Waiting Indication (SIP Phones in SRST Mode)

On SIP phones operating in the SIP SRST mode, you can use the **mwi unsolicited** command to configure a message-waiting notification when a message is sent by the Cisco Unity Express (CUE). The SIP phone then displays the notification when indicated by the voice messaging system. To configure message-waiting notification, perform the following steps.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **sip-ua**
4. **mwi-server** {**ipv4:***destination-address* | **dns**:*host*-**name**}[**unsolicited**]
5. **exit**
6. **voice register global**
7. **mwi unsolicited**
8. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**sip-ua** **Example:** `Router(config)# sip-ua`	Enters Session Initiation Protocol (SIP) user agent (ua) configuration mode for configuring the user agent.
Step 4	**mwi-server** {**ipv4:***destination-address* | **dns**:*host*-**name**}[**unsolicited**] **Example:** For g711alaw Codec `Router(config-sip-ua)# mwi-server ipv4:10.0.2.254 unsolicited` Or	Configures voicemail server settings on a voice gateway or user agent. Keywords and arguments are as follows: • **ipv4:***destination-address*: IP address of the voicemail server. • **dns:***host-name*: The argument should contain the complete hostname to be associated with the target address; for example, dns:test.cisco.com.

	Command or Action	Purpose
	`Router(config-sip-ua)# mwi-server dns:server.yourcompany.com unsolicited`	• **unsolicited**: Requires the voicemail server to send a SIP notification message to the voice gateway or UA if the mailbox status changes. Removes the requirement that the voice gateway subscribe for MWI service.
Step 5	**exit** **Example:** `Router(config-sip-ua)# exit`	Exits SIP user-agent configuration mode.
Step 6	**voice register global** **Example:** `Router(config)# voice register global`	Enters voice register global configuration mode to set parameters for all supported SIP phones in SIP SRST mode.
Step 7	**mwi unsolicited** **Example:** `Router(config-register-global)# mwi unsolicited`	Enables all SIP phones to receive MWI notification.

Step 8	**end**
	Example:
	`Router(config-register-global)# end`
		Exits to privileged EXEC mode.

Configuration Examples for Unified SRST

This section provides the following configuration examples:

Configuring Local Voicemail System (FXO and FXS): Example

The “Dial-Peer Configuration for Integration of Voicemail with Cisco Unified SRST” section of the example below shows a legacy dial-peer configuration for a local voicemail system. The “Cisco Unified SRST Voicemail Integration Pattern Configuration” section must be compatible with your voicemail system configuration.

```
! Dial-Peer Configuration for Integration of voicemail with Cisco Unified SRST
!
dial-peer voice 101 pots
destination-pattern 14011
port 3/0/0
!
dial-peer voice 102 pots
preference 1
destination-pattern 14011
port 3/0/1
!
dial-peer voice 103 pots
preference 2
destination-pattern 14011
port 3/1/0
!
dial-peer voice 104 pots
```

```
destination-pattern 14011
port 3/1/1
!
! Cisco Unified SRST configuration
!
call-manager-fallback
max-ephones 24
max-dn 144
ip source-address 1.4.214.104 port 2000
voicemail 14011
call-forward busy 14011
call-forward noan 14011 timeout 3
! Cisco Unified SRST voicemail Integration Pattern Configuration
!
vm-integration
pattern direct 2 CGN *
pattern ext-to-ext no-answer 5 FDN * CGN *
pattern ext-to-ext busy 7 FDN * CGN *
pattern trunk-to-ext no-answer 4 FDN * CGN *
pattern trunk-to-ext busy 6 FDN * CGN *
```

Configuring Central Location Voicemail System (FXO and FXS): Example

The "Dial-Peer Configuration for Integration of voicemail with Cisco Unified SRST in Central Location" section of the example shows a legacy dial-peer configuration for a central voicemail system. The "Cisco Unified SRST Voicemail Integration Pattern Configuration" section must be compatible with your voicemail system configuration.

Note Message waiting indicator (MWI) integration is not supported for PSTN access to voicemail systems at central locations.

```
! Dial-Peer Configuration for Integration of voicemail with Cisco Unified SRST in Central
! Location
!
dial-peer voice 101 pots
destination-pattern 14011
port 3/0/0
!
! Cisco Unified SRST configuration
!
call-manager-fallback
max-ephones 24
max-dn 144
ip source-address 1.4.214.104 port 2000
voicemail 14011
call-forward busy 14011
call-forward noan 14011 timeout 3
!
! Cisco Unified SRST Voicemail Integration Pattern Configuration
!
vm-integration
pattern direct 2 CGN *
pattern ext-to-ext no-answer 5 FDN * CGN *
pattern ext-to-ext busy 7 FDN * CGN *
pattern trunk-to-ext no-answer 4 FDN * CGN *
pattern trunk-to-ext busy 6 FDN * CGN *
```

Configuring Voicemail Access over FXO and FXS: Example

The following example shows how to configure the Cisco Unified SRST router to forward unanswered calls to voicemail. In this example, the voicemail number is 1101, the voicemail system is connected to FXS voice port 1/1/1, and the voice mailbox numbers are 3001, 3002, and 3006.

```
voice-port 1/1/1
timing digit 250
timing inter-digit 250
dial-peer voice 1102 pots
destination-pattern 1101T
port 1/1/1
call-manager-fallback
timeouts interdigit 5
ip source-address 1.6.0.199 port 2000
max-ephones 24
max-dn 24
transfer-pattern 3...
voicemail 1101
call-forward busy 1101
call-forward noan 1101 timeout 3
moh minuet.au
vm-integration
pattern direct * CGN
pattern ext-to-ext no-answer # FDN #2
pattern ext-to-ext busy # FDN #2
pattern trunk-to-ext no-answer # FDN #2
pattern trunk-to-ext busy # FDN #2
```

Configuring Voicemail Access over BRI and PRI: Example

The following example shows how to configure the Cisco Unified SRST router to forward unanswered calls to voicemail. In this example, the voicemail number is 1101, the voicemail system is connected to a BRI or PRI voice port, and the voice mailbox numbers are 3001, 3002, and 3006.

```
controller T1 2/0
framing esf
clock source line primary
linecode b8zs
cablelength short 133
pri-group timeslots 21-24
interface Serial2/0:23
no ip address
no logging event link-status
isdn switch-type primary-net5
isdn incoming-voice voice
isdn T309-enable
no cdp enable
voice-port 2/0:23
dial-peer voice 1102 pots
destination-pattern 1101T
direct-inward-dial
port 2/0:23
call-manager-fallback
timeouts interdigit 5
ip source-address 1.6.0.199 port 2000
max-ephones 24
max-dn 24
transfer-pattern 3...
voicemail 1101
call-forward busy 1101
```

```
call-forward noan 1101 timeout 3
moh minuet.au
```

Message Waiting Indication for SIP SRST: Example

The following is an example of a NOTIFY message received at SRST indicating that there is a voicemail for extension 32002:

```
Received:
NOTIFY sip:32002@10.4.49.65:5060;transport=udp SIP/2.0
Via: SIP/2.0/UDP 10.4.49.66:5060;branch=z9hG4bK.D6.7wAl9CN6khf305D1MQ~~194
Max-Forwards: 70
To: <sip:32002@10.4.49.65:5060>
From: <sip:32002@10.4.49.66:5060>;tag=dsd3d29b2f
Call-ID: f0e7ae97-1227@sip:32002@10.4.49.66:5060
CSeq: 1 NOTIFY
Content-Length: 112
Contact: <sip:32002@10.4.49.66:5060>
Content-Type: application/simple-message-summary
Event: message-summary
Messages-Waiting: yes
Message-Account: sip:32002@10.4.49.66
Voice-Message: 1/0 (1/0)
Fax-Message: 0/0 (0/0)
```

How to Configure DTMF Relay for SIP Applications and Voicemail

For SIP SRST forwarding call to voicemail configuration, see the Configuring Call Handling section.

Note Voicemail number associate with SIP phone message button in SRST is configured by Cisco Unified Communications Manager (CUCM), and not configurable by SIP SRST. The administrator needs to know the voicemail number set by CUCM to configure proper dial peer to voicemail system in SIP SRST.

DTMF relay for SIP applications can be used in two voicemail situations:

DTMF Relay Using SIP RFC 2833

Cisco Unified Skinny Client Control Protocol (SCCP) Phones, such as those used with Cisco Unified SRST systems, provide only out-of-band DTMF digit indications. To enable SCCP phones to send digit information to remote SIP-based IVR and voicemail applications, Cisco Unified SRST 3.2 and later versions provide conversion from the out-of-band SCCP digit indication to the SIP standard for DTMF relay, which is RFC 2833. You select this method in the SIP VoIP dial peer using the dtmf-relay rtp-nte command.

The SIP DTMF relay method is needed in the following situations:

- When SIP is used to connect a Cisco Unified SRST system to a remote SIP-based IVR or voicemail application, such as Cisco Unity.
- When SIP is used to connect a Cisco Unified SRST system to a remote SIP-PSTN voice gateway that goes through the PSTN to a voicemail or IVR application.

Note The need to use out-of-band DTMF relay conversion is limited to SCCP phones. SIP phones natively support in-band DTMF relay as specified in RFC 2833.

To enable SIP DTMF relay using RFC 2833, the commands in this section must be used on both originating and terminating gateways.

SUMMARY STEPS

1. **dial-peer voice***tag***voip**
2. **dtmf-relay rtp-nte**
3. **dtmf-relay rtp-nte**
4. **exit**
5. **sip-ua**
6. **notify telephone-event max-duration***time*
7. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**dial-peer voice***tag***voip** **Example:** `Router(config)# dial-peer voice 2 voip`	Enters dial-peer configuration mode.
Step 2	**dtmf-relay rtp-nte** **Example:** `Router(config-dial-peer)# dtmf-relay rtp-nte`	Enters global configuration mode.
Step 3	**dtmf-relay rtp-nte** **Example:** `Router(config)# voice register global`	Forwards DTMF tones by using Real-Time Transport Protocol (RTP) with the Named Telephone Event (NTE) payload type.
Step 4	**exit** **Example:** `Router(config-dial-peer)# exit`	Exits dial-peer configuration mode.
Step 5	**sip-ua** **Example:** `Router(config)# sip-ua`	Enables SIP user-agent configuration mode.
Step 6	**notify telephone-event max-duration***time* **Example:** `Router(config-sip-ua)# notify telephone-event max-duration 2000`	Configures the maximum time interval allowed between two consecutive NOTIFY messages for a single DTMF event. • **max-duration***time* : Time interval between consecutive NOTIFY messages for a single DTMF

	Command or Action	Purpose
		event, in milliseconds. Range is from 500 to 3000. Default is 2000.
Step 7	**exit** **Example:** `Router(config-sip-ua)# exit`	Exits SIP user-agent configuration mode.

Troubleshooting Tips

The dial-peer section of the **show running-config** command output displays DTMF relay status when it is configured, as shown in this excerpt:

```
dial-peer voice 123 voip
destination-pattern [12]...
monitor probe icmp-ping
session protocol sipv2
session target ipv4:10.8.17.42
dtmf-relay rtp-nte
```

DTMF Relay Using SIP Notify (Nonstandard)

To use voicemail on a SIP network that connects to a Cisco Unity Express system, use a nonstandard SIP Notify format. To configure the Notify format, use the **sip-notify** keyword with the **dtmf-relay** command. Using the **sip-notify** keyword may be required for backward compatibility with Cisco Unified SRST Versions 3.0 and 3.1.

SUMMARY STEPS

1. **dial-peer voice***tag***voip**
2. **dtmf-relay sip-notify**
3. **exit**
4. **sip-ua**
5. **notify telephone-event max-duration***time*
6. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**dial-peer voice***tag***voip** **Example:** `Router(config)# dial-peer voice 2 voip`	Enters dial-peer configuration mode.
Step 2	**dtmf-relay sip-notify** **Example:** `Router(config-dial-peer)# dtmf-relay sip-notify`	Forwards DTMF tones using SIP NOTIFY messages.

	Command or Action	Purpose
Step 3	**exit** **Example:** `Router(config-dial-peer)# exit`	Exits dial-peer configuration mode.
Step 4	**sip-ua** **Example:** `Router(config)# sip-ua`	Enables SIP user-agent configuration mode.
Step 5	**notify telephone-event max-duration***time* **Example:** `Router(config-sip-ua)# notify telephone-event max-duration 2000`	Configures the maximum time interval allowed between two consecutive NOTIFY messages for a single DTMF event. • **max-duration***time* : Time interval between consecutive NOTIFY messages for a single DTMF event, in milliseconds. Range is from 500 to 3000. Default is 2000.
Step 6	**exit** **Example:** `Router(config-sip-ua)# exit`	Exits SIP user-agent configuration mode.

Troubleshooting Tips

The **show sip-ua status** command output displays the time interval between consecutive NOTIFY messages for a telephone event. In the following example, the time interval is 2000 ms:

```
Router# show sip-ua status
SIP User Agent Status
SIP User Agent for UDP :ENABLED
SIP User Agent for TCP :ENABLED
SIP User Agent bind status(signaling):DISABLED
SIP User Agent bind status(media):DISABLED
SIP early-media for 180 responses with SDP:ENABLED
SIP max-forwards :6
SIP DNS SRV version:2 (rfc 2782)
NAT Settings for the SIP-UA
Role in SDP:NONE
Check media source packets:DISABLED
Maximum duration for a telephone-event in NOTIFYs:2000 ms
SIP support for ISDN SUSPEND/RESUME:ENABLED
Redirection (3xx) message handling:ENABLED
SDP application configuration:
Version line (v=) required
Owner line (o=) required
Timespec line (t=) required
Media supported:audio image
Network types supported:IN
Address types supported:IP4
Transport types supported:RTP/AVP udptl
```

CHAPTER 13

Setting Video Parameters

This chapter describes how to set video parameters for a Cisco Unified Survivable Remote Site Telephony (SRST) Router.

Prerequisites for Setting Video Parameters

- Ensure that you are using Cisco Unified SRST 4.0 or a later version.
- Ensure that you are using Cisco Unified Communications Manager 4.0 or a later version.
- Ensure that the Cisco IP phones are registered with the Cisco Unified SRST router. Use the **show ephone registered** command to verify ephone registration.
- Ensure that the connection between the Cisco Unified Video Advantage application and the Cisco Unified IP phone is up.

  From a PC with Cisco Unified Video Advantage 1.02 or a later version installed, ensure that the line between the Cisco Unified Video Advantage and the Cisco Unified IP phone is green. For more information, see Cisco Unified Video Advantage End-User Guides.
- Ensure that you install the correct video firmware on the Cisco Unified IP phone. Use the **show ephone phone-load** command to view current ephone firmware. The following lists the minimum firmware version for video-enabled Cisco Unified IP phones:

  Cisco Unified IP Phone 7940G version 6.0(4)

  Cisco Unified IP Phone 7960G version 6.0(4)

  Cisco Unified IP Phone 7970G version 6.0(2)
- Perform basic Cisco Unified SRST configuration. For more information, see Cisco Unified SRST V4.0: Setting Up the Network.
- Perform basic ephone configuration. For more information, see Cisco Unified SRST V4.0: Setting Up Cisco Unified IP Phones.

Restrictions for Setting Video Parameters

- This feature supports only the following video codecs:

  H.261

  H.263

  H.264 (for CUVA from SRST 7.1)

- This feature supports only the following video formats:

  Common Intermediate Format (CIF): Resolution 352x288

  One-Quarter Common Intermediate Format (QCIF): Resolution 176x144

  Sub QIF (SQCIF): Resolution 128x96

  4CIF: Resolution 704x576

  16CIF: Resolution 1408x1152

- The call start fast feature does not support an H.323 video connection. You must configure call start slow for H.323 video.
- Video capabilities are configured per ephone, not per line.
- All call feature controls (for example, mute and hold) apply to both audio and video calls, if applicable.
- This feature does not support the following:

  Dynamic addition of video capability: The video capability must be present before the call setup starts to allow the video connection.

  T-120 data connection between two SCCP endpoints

  Video security

  Far-end camera control (FECC) for SCCP endpoints

  Video codec renegotiation: The negotiated video codec must match or the call falls back to audio-only. The negotiated codec for the existing call can be used for an incoming call. Video codec transcoding

- When a video-capable endpoint connects to an audio-only endpoint, the call falls back to audio-only. During audio-only calls, video messages are skipped.

Information About Setting Video Parameters

This feature allows you to set video parameters for the Cisco Unified SRST to maintain close feature parity with Cisco Unified Communications Manager. When the Cisco Unified SRST is enabled, Cisco Unified IP phones do not have to be reconfigured for video capabilities because all ephones retain the same configuration used with Cisco Unified Communications Manager. However, you must enter call-manager-fallback configuration mode to set video parameters for Cisco Unified SRST. The feature set for video is the same as the Cisco Unified SRST audio calls.

To set video parameters, refer the following concepts:

Matching Endpoint Capabilities

Cisco Unified SRST stores Endpoint capabilities during the phone registration. These capabilities are used to match with other endpoints during the call setup. Endpoints can update at any time; however, the router recognizes endpoint capability changes only during the call setup. If you add a video feature to a phone, the information about it is updated in the router's internal data structure. However, the information does not take effect until the next call. If a video feature is revoked, the router continues to view the video capability until the call stops. However, there is no video stream that is exchanged between the two endpoints.

Note The endpoint capability match is executed every time when an incoming call is set up or an existing call is resumed.

Retrieving Video Codec Information

Voice gateways use dial-peer configurations to retrieve codec information for audio codecs. Video codec selection is done by the endpoints and is not controlled by the H.323 service-provider interface (SPI) through dial-peer or other configuration. The video-codec information is retrieved from the SCCP endpoint using a capabilities request during the call setup.

Call Fallback to and Audio-Only Endpoint

When a video-capable endpoint connects to an audio-only endpoint, the call falls back to an audio-only connection. Also, for certain features such as conferencing, where video support is not available, the call falls back to audio-only.

Cisco Unified SRST routers use a call-type flag to indicate whether the call is video-capable or audio-only. The call-type flag is set to video when the video capability is matched or set to audio-only when connecting to an audio-only TDM or an audio-only SIP endpoint.

Note During an audio-only connection, all video-related media messages are skipped.

Call Setup for Video Endpoints

The process for handling SCCP video endpoints is the same as that for handling SCCP audio endpoints. The video call must be part of the audio call. If the audio call setup fails, the video call fails.

During call setup for video, media setup handling determines if a video-media path is required or not. If so, the corresponding video-media-path setup actions are taken.

- For an SCCP endpoint, video-media-path setup includes sending messages to the endpoints to open a multimedia path and start the multimedia transmission.
- For an H.323 endpoint, video-media-path setup includes an Exchange between the endpoints to open a logical channel for the video stream.

A call-type flag is set during the call setup on the basis of the endpoint and capability match. After call setup, the call -type flag is used to determine whether an extra video-media path is required. Call signaling is managed by the Cisco Unified Communications Manager Express router, and the media stream is directly connected between the two video-enabled SCCP endpoints on the same router. Video-related commands and flow-control messages are forwarded to the other endpoint. Routers do not interpret these messages.

Call Setup Between Two Local SCCP Endpoints

For interoperation between two local SCCP endpoints (that exist on the same router), video call setup uses all existing audio-call-setup handling, except during the media setup. During the media setup, a message is sent to establish the video-media path. If the endpoint responds, the video-media path is established and invokes a start-multimedia-transmission function.

Call Setup Between SCCP and H.323 Endpoints

Call setup between SCCP and H.323 endpoints is the same as it is between SCCP endpoints except that, if video capability is selected, the event is posted to the H.323 call leg to send out a video open logical channel (OLC) and the gateway generates an OLC for the video channel. Because the router needs to both stop and originate the media stream, video must be enabled on the router before call setup begins.

Call Setup Between Two SCCP Endpoints Across an H.323 Network

If the call setup between SCCP endpoints occurs across an H.323 network, the setup is a combination of the processes listed in the previous two sections. The router controls the video media setup between the two endpoints, and the event is posted to the H.323 call leg so that the gateway can generate an OLC.

Flow of the RTP Video Stream

For video streams between two local SCCP endpoints, the Real-Time Transport Protocol (RTP) stream is in flow-around mode. For video streams between SCCP and H.323 endpoints or two SCCP endpoints on different Cisco Unified Communications Manager Express routers, the RTP stream is in flow-through mode.

- Media flow-around mode enables RTP packets to stream directly between the endpoints of a VoIP call without the involvement of the gateway. By default, the gateway receives the incoming media, stops the call, and then reoriginates it on the outbound call leg. In flow-around mode, only signaling data is passed to the gateway, improving scalability and performance.
- Media flow-through mode involves the same video-media path as for an audio call. Media packets flow through the gateway, thus hiding the networks from each other.

To display information about RTP named-event packets, such as caller-ID number, IP address, and port for both the local and remote endpoints, use the **show voip rtp connection** command as shown in the following sample output:

```
Router# show voip rtp connections
VoIP RTP active connections :
No. CallId dstCallId LocalRTP RmtRTP LocalIP RemoteIP
1 102 103 18714 18158 10.1.1.1 192.168.1.1
2 105 104 17252 19088 10.1.1.1 192.168.1.1
Found 2 active RTP connections
============================
```

How to Set Video Parameters for Cisco Unified SRST

When you enable the Cisco Unified SRST, do not reconfigure the Cisco Unified IP phones for video capabilities. All ephones retain the same configuration used with Cisco Unified Communications Manager. However, you can set video parameters for Cisco Unified SRST.

The following are the task for setting Video parameters for Cisco Unified SRST:

Configuring Slow Connect Procedures

Video streams require slow-connect procedures for Cisco Unified SRST. H.323 endpoints require a slow connect because the endpoint-capability match occurs after the connect message.

Note For more information about slow-connect procedures, see Configuring Quality of Service for Voice.

Use the following procedure to configure slow-connect procedures.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **h323**
5. **call start slow**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice service voip** **Example:** `Router(config)# voice service voip`	Enters voice-service configuration mode.
Step 4	**h323** **Example:** `Router(config-voi-serv)# h323`	Enters H.323 voice-service configuration mode.
Step 5	**call start slow** **Example:** `Router(config-serv-h323)# call start slow`	Forces an H.323 gateway to use slow-connect procedures for all VoIP calls.

Verifying Cisco Unified SRST

Use the following procedure to verify that the Cisco Unified SRST feature is enabled and to verify Cisco Unified IP phone configuration settings.

SUMMARY STEPS

1. **enable**
2. **show running config**
3. **show call-manager-fallback all**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**show running config** **Example:** `Router# show running config`	Displays the entire contents of the running configuration file.
Step 3	**show call-manager-fallback all** **Example:** `Router# show call-manager-fallback all`	Displays the detailed configuration of all Cisco Unified IP phones, directory numbers, voice ports, and dial peers in your network while in fallback mode. **Note** Use the Settings display on the Cisco Unified IP phones in your network to verify that the default router IP address on the phones matches the IP address of the Cisco Unified SRST router.

Example

The following example shows output from the show call-manager-fallback all command:

```
Router# show call-manager-fallback all
CONFIG (Version=3.3)
=====================
Version 3.3
For on-line documentation please see:
www.cisco.com/univercd/cc/td/doc/product/access/ip_ph/ip_ks/index.htm
ip source-address 10.1.1.1 port 2000
max-video-bit-rate 384(kbps)
max-ephones 52
max-dn 110
max-conferences 16 gain -6
dspfarm units 0
dspfarm transcode sessions 0
huntstop
dialplan-pattern 1 4084442... extension-length 4
voicemail 6001
moh music-on-hold.au
time-format 24
```

```
date-format dd-mm-yy
timezone 0 Greenwich Standard Time
call-forward busy 6001
call-forward noan 6001 timeout 8
call-forward pattern .T
transfer-pattern .T
keepalive 45
timeout interdigit 10
timeout busy 10
timeout ringing 180
caller-id name-only: enable
Limit number of DNs per phone:
7910: 34
7935: 34
7936: 34
7940: 34
7960: 34
7970: 34
Log (table parameters):
max-size: 150
retain-timer: 15
transfer-system full-consult
local directory service: enabled.
ephone-dn 1
number 1001
name 1001
description 1001
label 1001
preference 0 secondary 9
huntstop
call-forward busy 6001
call-forward noan 6001 timeout 8
call-waiting beep
ephone-dn 2
number 1002
name 1002
description 1002
preference 0 secondary 9
huntstop
call-forward busy 6001
call-forward noan 6001 timeout 8
call-waiting beep
ephone-dn 3
preference 0 secondary 9
huntstop
call-waiting beep
ephone-dn 4
preference 0 secondary 9
huntstop
call-waiting beep
ephone-dn 5
preference 0 secondary 9
huntstop
call-waiting beep
ephone-dn 6
preference 0 secondary 9
huntstop
call-waiting beep
ephone-dn 7
preference 0 secondary 9
huntstop
call-waiting beep
ephone-dn 8
preference 0 secondary 9
```

```
huntstop
call-waiting beep
ephone-dn 9
preference 0 secondary 9
huntstop
call-waiting beep
ephone-dn 10
preference 0 secondary 9
huntstop
call-waiting beep
ephone-dn 11
preference 0 secondary 9
huntstop
call-waiting beep
ephone-dn 12
preference 0 secondary 9
huntstop
call-waiting beep
ephone-dn 13
preference 0 secondary 9
huntstop
call-waiting beep
ephone-dn 14
preference 0 secondary 9
huntstop
call-waiting beep
ephone-dn 15
preference 0 secondary 9
huntstop
call-waiting beep
ephone-dn 16
preference 0 secondary 9
huntstop
call-waiting beep
ephone-dn 17
preference 0 secondary 9
huntstop
call-waiting beep
ephone-dn 18
preference 0 secondary 9
huntstop
call-waiting beep
ephone-dn 19
preference 0 secondary 9
huntstop
call-waiting beep
ephone-dn 20
preference 0 secondary 9
huntstop
call-waiting beep
Number of Configured ephones 0 (Registered 2)
voice-port 50/0/1
station-id number 1001
station-id name 1001
timeout ringing 8
!
voice-port 50/0/2
station-id number 1002
station-id name 1002
timeout ringing 8
!
voice-port 50/0/3
!
voice-port 50/0/4
```

```
!
voice-port 50/0/5
!
voice-port 50/0/6
!
voice-port 50/0/7
!
voice-port 50/0/8
!
voice-port 50/0/9
!
voice-port 50/0/10
!
voice-port 50/0/11
!
voice-port 50/0/12
!
voice-port 50/0/13
!
voice-port 50/0/14
!
voice-port 50/0/15
!
voice-port 50/0/16
!
voice-port 50/0/17
!
voice-port 50/0/18
!
voice-port 50/0/19
!
voice-port 50/0/20
!
dial-peer voice 20055 pots
destination-pattern 1001
huntstop
call-forward busy 6001
call-forward noan 6001
progress_ind setup enable 3
port 50/0/1
dial-peer voice 20056 pots
destination-pattern 1002
huntstop
call-forward busy 6001
call-forward noan 6001
progress_ind setup enable 3
port 50/0/2
dial-peer voice 20057 pots
huntstop
progress_ind setup enable 3
port 50/0/3
dial-peer voice 20058 pots
huntstop
progress_ind setup enable 3
port 50/0/4
dial-peer voice 20059 pots
huntstop
progress_ind setup enable 3
port 50/0/5
dial-peer voice 20060 pots
huntstop
progress_ind setup enable 3
port 50/0/6
dial-peer voice 20061 pots
```

```
huntstop
progress_ind setup enable 3
port 50/0/7
dial-peer voice 20062 pots
huntstop
progress_ind setup enable 3
port 50/0/8
dial-peer voice 20063 pots
huntstop
progress_ind setup enable 3
port 50/0/9
dial-peer voice 20064 pots
huntstop
progress_ind setup enable 3
port 50/0/10
dial-peer voice 20065 pots
huntstop
progress_ind setup enable 3
port 50/0/11
dial-peer voice 20066 pots
huntstop
progress_ind setup enable 3
port 50/0/12
dial-peer voice 20067 pots
huntstop
progress_ind setup enable 3
port 50/0/13
dial-peer voice 20068 pots
huntstop
progress_ind setup enable 3
port 50/0/14
dial-peer voice 20069 pots
huntstop
progress_ind setup enable 3
port 50/0/15
dial-peer voice 20070 pots
huntstop
progress_ind setup enable 3
port 50/0/16
dial-peer voice 20071 pots
huntstop
progress_ind setup enable 3
port 50/0/17
dial-peer voice 20072 pots
huntstop
progress_ind setup enable 3
port 50/0/18
dial-peer voice 20073 pots
huntstop
progress_ind setup enable 3
port 50/0/19
dial-peer voice 20074 pots
huntstop
progress_ind setup enable 3
port 50/0/20
tftp-server system:/its/SEPDEFAULT.cnf
tftp-server system:/its/SEPDEFAULT.cnf alias SEPDefault.cnf
tftp-server system:/its/XMLDefault.cnf.xml alias XMLDefault.cnf.xml
tftp-server system:/its/ATADefault.cnf.xml
tftp-server system:/its/united_states/7960-tones.xml alias United_States/7960-tones.xml
tftp-server system:/its/united_states/7960-font.xml alias
English_United_States/7960-font.xml
tftp-server system:/its/united_states/7960-dictionary.xml alias
English_United_States/7960-dictionary.xml
```

```
tftp-server system:/its/united_states/7960-kate.xml alias
English_United_States/7960-kate.xml
tftp-server system:/its/united_states/SCCP-dictionary.xml alias
English_United_States/SCCP-dictionary.xml
tftp-server system:/its/XMLDefault7960.cnf.xml alias SEP003094C2772E.cnf.xml
tftp-server system:/its/XMLDefault7960.cnf.xml alias SEP001201372DD1.cnf.xml
tftp-server system:/its/XMLDefault7960.cnf.xml alias SEPFFDD00000001.cnf.xml
tftp-server system:/its/XMLDefault7960.cnf.xml alias SEPFFDD00000002.cnf.xml
tftp-server system:/its/XMLDefault7960.cnf.xml alias SEPFFDD00000003.cnf.xml
tftp-server system:/its/XMLDefault7960.cnf.xml alias SEPFFDD00000004.cnf.xml
tftp-server system:/its/XMLDefault7960.cnf.xml alias SEPFFDD00000005.cnf.xml
tftp-server system:/its/XMLDefault7960.cnf.xml alias SEPFFDD00000006.cnf.xml
tftp-server system:/its/XMLDefault7960.cnf.xml alias SEPFFDD00000007.cnf.xml
tftp-server system:/its/XMLDefault7960.cnf.xml alias SEPFFDD00000008.cnf.xml
tftp-server system:/its/XMLDefault7960.cnf.xml alias SEPFFDD00000009.cnf.xml
tftp-server system:/its/XMLDefault7960.cnf.xml alias SEPFFDD0000000A.cnf.xml
tftp-server system:/its/XMLDefault7960.cnf.xml alias SEPFFDD0000000B.cnf.xml
tftp-server system:/its/XMLDefault7960.cnf.xml alias SEPFFDD0000000C.cnf.xml
tftp-server system:/its/XMLDefault7960.cnf.xml alias SEPFFDD0000000D.cnf.xml
tftp-server system:/its/XMLDefault7960.cnf.xml alias SEPFFDD0000000E.cnf.xml
tftp-server system:/its/XMLDefault7960.cnf.xml alias SEPFFDD0000000F.cnf.xml
tftp-server system:/its/XMLDefault7960.cnf.xml alias SEPFFDD00000010.cnf.xml
tftp-server system:/its/XMLDefault7960.cnf.xml alias SEPFFDD00000011.cnf.xml
tftp-server system:/its/XMLDefault7960.cnf.xml alias SEPFFDD00000012.cnf.xml
```

Setting Video Parameters for Cisco Unified SRST

Using the following procedure to set the maximum bit rate for all video-capable phones in a Cisco Unified SRST system.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dcall-manager-fallback**
4. **video**
5. **maximum bit-rate** *value*

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**dcall-manager-fallback** **Example:** `Router(config)# call-manager-fallback`	Enters call-manager-fallback configuration mode.

	Command or Action	Purpose
Step 4	**video** **Example:** `Router(config-call-manager-fallback)# video`	Enters call-manager-fallback video configuration mode.
Step 5	**maximum bit-rate** *value* **Example:** `Router(conf-cm-fallback-video)# maximum bit-rate 256`	Sets the maximum IP phone video bandwidth, in kbps. The range is 0 to 10000000. The default is 10000000.

Example

The following example shows the configuration for video with Cisco Unified SRST:

```
call-manager-fallback
video
maximum bit-rate 384
max-conferences 2 gain -6
transfer-system full-consult
ip source-address 10.0.1.1 port 2000
max-ephones 52
max-dn 110
dialplan-pattern 1 4084442... extension-length 4
transfer-pattern .T
keepalive 45
voicemail 6001
call-forward pattern .T
call-forward busy 6001
call-forward noan 6001 timeout 3
moh music-on-hold.au
time-format 24
date-format dd-mm-yy
!
```

Troubleshooting Video for Cisco Unified SRST

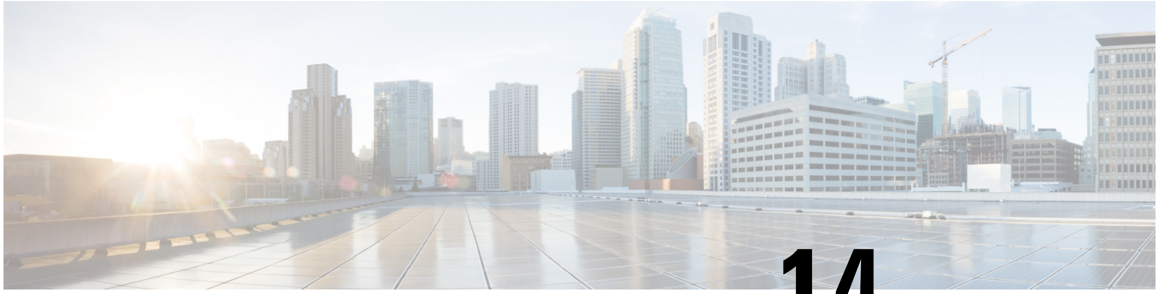
Use the following commands to troubleshoot Video for Cisco Unified SRST.

1. For SCCP endpoint troubleshooting, use the following debug commands:
    - Debug cch323 video: Enables the video debugging trace on the H.323 SPI.
    - Debug ephone detail: Debugs all Cisco Unified IP phones that are registered to the router and displays error and state levels.
    - Debug h225 asn1: Displays Abstract Syntax Notation One (ASN.1) contents of H.225 messages that are sent or received.
    - Debug h245 asn1: Displays ASN.1 contents of H.245 messages that are sent or received.
    - Debug VoIP CCAPI inout: Displays the execution path through the call-control-application programming interface (CPI).
2. For ephone troubleshooting, use the following debug commands:

- Debug ephone message: Enables message tracing between Cisco ephones.
- Debug ephone register: Sets registration debugging for ephones.
- Debug ephone video: Sets ephone video traces, which provide information about different video states for the call, including video capabilities selection, start, and stop.

3. For basic video-to-video call checking, use the following show commands:

- Show call active video: Displays call information for SCCP video CallsInProgress.
- Show ephone off hook: Displays information and packet counts for ephones that are currently off hook.
- Show VoIP RTP connections: Displays information about RTP named-event packets, such as caller ID number, IP address, and port, for both the local and remote endpoints.



CHAPTER 14

Monitoring and Maintaining Cisco Unified SRST

Monitoring and Maintaining Cisco Unified SRST

To monitor and maintain Cisco Unified Survivable Remote Site Telephony (SRST), use the following commands in privileged EXEC mode.

Command	Purpose
Router# **show call-manager-fallback all**	Displays the detailed configuration of all the Cisco Unified IP phones, voice ports, and dial peers of the Cisco Unified SRST Router.
Router# **show call-manager-fallback dial-peer**	Displays the output of the dial peers of the Cisco Unified SRST Router.
Router# **show call-manager-fallback ephone-dn**	Displays Cisco Unified IP Phone destination numbers when in Cisco Unified Communications Manager fallback mode.
Router# **show call-manager-fallback voice-port**	Displays output for the voice ports.
Router# **show dial-peer voice summary**	Displays a summary of all voice dial peers.
Router# **show ephone***phone*	Displays Cisco Unified IP Phone status.
Router# **show ephone offhook**	Displays Cisco Unified IP Phone status for all phones that are off hook.
Router# **show ephone registered**	Displays Cisco Unified IP Phone status for all phones that are currently registered.
Router# **show ephone remote**	Displays Cisco Unified IP Phone status for all nonlocal phones (phones that have no Address Resolution Protocol [ARP] entry).
Router# **show ephone ringing**	Displays Cisco Unified IP Phone status for all phones that are ringing.

Command	Purpose
Router# **show ephone summary**	Displays a summary of all Cisco Unified IP Phones.
Router# **show ephone telephone-number***phone-number*	Displays Unified IP Phone status for a specific phone number.
Router# **show ephone unregistered**	Displays Unified IP Phone status for all unregistered phones.
Router# **show ephone-dn***tag*	Displays Unified IP Phone destination numbers.
Router# **show ephone-dn summary**	Displays a summary of all Cisco Unified IP Phone destination numbers.
Router# **show ephone-dn loopback**	Displays Cisco Unified IP Phone destination numbers in loopback mode.
Router# **show running-config**	Display the configuration.
Router# **show sip-ua status registrar**	Display SIP registrar clients.
Router# **show voice port summary**	Displays a summary of all voice ports.
Router# **show voice register all**	Displays all SIP SRST configurations, SIP phone registrations, and dial peer information.
Router# **show voice register global**	Displays voice register global config.
Router# **show voice register pool all**	Displays all config SIP phone voice register Pool detail information.
Router# **show voice register pool** *tag*	Displays specific SIP phone voice register Pool detail information.
Router# **show voice register dial-peers**	Displays SIP-SRST created dial peer.
Router# **show voice register dn all**	Displays all config voice register directory number detail information.
Router# **show voice register dn** *tag*	Displays specific voice register directory number detail information.

APPENDIX A

Appendix A: Configuring Cisco Unified SIP SRST Features Using Redirect Mode

This chapter describes Cisco Unified Session Initiation Protocol (SIP) Survivable Remote Site Telephony (SRST) features using redirect mode.

Note This chapter applies to version 3.0 only.

Prerequisites for Cisco Unified SIP SRST Features Using Redirect Mode

Complete the prerequisites documented in the Cisco Unified SRST Feature Overview, on page 41 chapter.

Restrictions for Cisco Unified SIP SRST Features Using Redirect Mode

See the restrictions documented in the Cisco Unified SRST Feature Overview, on page 41 chapter.

Information About Cisco Unified SIP SRST Features Using Redirect Mode

Cisco Unified SIP SRST provides backup to an external SIP call control (IP-PBX) by providing basic registrar and redirect services. These services are used by a SIP IP phone if a WAN connection outage when the SIP phone is unable to communicate with its primary SIP proxy. The Cisco Unified SIP SRST device also provides PSTN gateway access for placing and receiving PSTN calls.

To make maximum use of the Cisco Unified SIP SRST service, the local SIP IP phones should support dual (concurrent) registration with both their primary SIP proxy or registrar and the Cisco Unified SIP SRST backup registrar. Cisco Unified SIP SRST works for the following types of calls:

- Local SIP IP phone to local SIP phone, if the main proxy is unavailable.
- Other services like class of restriction (COR) for local SIP IP phones to the outgoing PSTN. For example, to block outgoing 1-900 numbers.

How to Configure Cisco Unified SIP SRST Features Using Redirect Mode

Configuring Call Redirect Enhancements to Support Calls Between SIP IP Phones for Cisco Unified SIP SRST

The call redirect enhancement supports calls from a local SIP phone to another local SIP phone through the Cisco IOS voice gateway. Before this enhancement, an attempt by a SIP phone to contact another local SIP phone using the Cisco IOS voice gateway as if it were a SIP proxy or redirect server would fail. However, the Cisco IOS voice gateway can now act as a SIP redirect server. The voice gateway responds to the originator with a SIP Redirect message, allowing the SIP phone that originated the call to establish a call to its destination.

The **redirect ip2ip** (voice service) and **redirect ip2ip** (dial-peer) commands allow you to enable the SIP functionality, globally or on a specific inbound dial peer. The default application on Cisco Unified SIP SRST supports IP-to-IP redirection.

Configuring Audio and Video Codecs at the Dial Peer Level

To enable global IP-to-IP call redirection for all VoIP dial peers, use voice service configuration mode.

Note When IP-to-IP redirection is configured in dial-peer configuration mode, the configuration for the specific dial peer takes precedence over the global configuration entered under voice service configuration mode.

SUMMARY STEPS

1. **enable**

2. **configure terminal**
3. **voice service voip**
4. **redirect ip2ip**
5. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice service voip** **Example:** `Router(config)# voice service voip`	Enters voice service configuration mode.
Step 4	**redirect ip2ip** **Example:** `Router(config-voi-srv)# redirect ip2ip`	Configures a video codec at the dial peer level. Redirects SIP phone calls to SIP phone calls globally on a gateway using the Cisco IOS voice gateway.

Step 5	**end**
	Example:
	`Router(config-voi-srv)# end`
		Returns to privileged EXEC mode.

Configuring Call Redirect Enhancements to Support Calls On a Specific VoIP Dial Peer

To enable IP-to-IP call redirection for a specific VoIP dial peer, configure it on an inbound dial peer in dial-peer configuration mode. The default application on Cisco Unified SIP SRST supports IP-to-IP redirection.

Note When IP-to-IP redirection is configured in dial-peer configuration mode, the configuration for the specific dial peer takes precedence over the global configuration entered under voice service configuration mode.

Before you begin

The **redirect ip2ip** command must be configured on an inbound dial peer of the gateway.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *tag* **voip**

4. **application** *application-name*
5. **redirect ip2ip**
6. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**dial-peer voice** *tag* **voip** **Example:** `Router(config)# dial-peer voice 25 voip`	Enters dial-peer configuration mode. • *tag* : A number that uniquely identifies the dial peer (this number has local significance only). • VoIP: Indicates that this is a VoIP peer using voice encapsulation on the POTS network and is used for configuring redirect.
Step 4	**application** *application-name* **Example:** `Router(config-dial-peer)# application session`	Enables a specific application on a dial peer. • For SIP, the default Tool Command Language (Tcl) application (from the Cisco IOS image) is session and can be applied to both VoIP and POTS dial peers. • The application must support IP-to-IP redirection.
Step 5	**redirect ip2ip** **Example:** `Router(config-dial-peer)# redirect ip2ip`	Redirects SIP phone calls to SIP phone calls on a specific VoIP dial peer using the Cisco IOS voice gateway.

Step 6	**end**
	Example:
	`Router(config-dial-peer)# end`
		Returns to privileged EXEC mode.

Configuring Sending 300 Multiple Choice Support

Before Cisco IOS Release 12.2(15)ZJ, when a call was redirected, the SIP gateway would send a 302 Moved Temporarily message. The first longest match route on a gateway (dial-peer destination pattern) was used in the Contact header of the 302 message. With Release 12.2(15)ZJ, if multiple routes to a destination exist for a redirected number (multiple dial peers are matched), the SIP gateway sends a 300 Multiple Choice message, and the multiple routes in the Contact header are listed.

The configuration below allows users to choose the order in which the routes appear in the Contact header.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **sip**
5. **redirect contact order [best-match | longestmatch]**
6. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice service voip** **Example:** `Router(config)# voice service voip`	Enters voice service configuration mode.
Step 4	**sip** **Example:** `Router(config-voi-srv)# sip`	Enters SIP configuration mode.
Step 5	**redirect contact order [best-match | longestmatch]** **Example:** `Router(conf-serv-sip)# redirect contact order best-match`	Sets the order of contacts in the 300 Multiple Choice message. The keywords are defined as follows: • **best-match** : Uses the current system configuration to set the order of contacts. • **longestmatch** : Sets the contact order by using the destination pattern longest match first, and then the second longest match, the third longest match, and so on. This is the default.

Step 6	**end**
	Example:
	`Router(config-serv-sip)# end`
		Returns to privileged EXEC mode.

Configuration Examples for Cisco Unified SIP SRST Features Using Redirect Mode

This section provides the following configuration example:

Cisco Unified SIP SRST: Example

This section provides a configuration example to match the configuration tasks in the previous sections.

```
!
! Sets up the registrar server and enables IP-to-IP redirection and 300
! Multiple Choice support.
!
voice service voip
redirect ip2ip
sip
registrar server expires max 600 min 60
redirect contact order best-match
!
! Configures the voice-class codec with G.711uLaw and G729 codecs. The codecs are
! applied to the voice register pools.
!
voice class codec 1
codec preference 1 g711ulaw
codec preference 2 g729br8
!
! The voice register pools define various pools that are used to match
! incoming REGISTER requests and create corresponding dial peers.
!
voice register pool 1
id mac 0030.94C2.A22A
preference 5
cor incoming call91 1 91011
translate-outgoing called 1
proxy 10.2.161.187 preference 1 monitor probe icmp-ping
alias 1 94... to 91011 preference 8
voice-class codec 1
!
voice register pool 2
id ip 192.168.0.3 mask 255.255.255.255
preference 5
cor outgoing call95 1 91021
proxy 10.2.161.187 preference 1
voice-class codec 1
!
voice register pool 3
id network 10.2.161.0 mask 255.255.255.0
number 1 95... preference 1
preference 5
cor incoming call95 1 95011
cor outgoing call95 1 95011
proxy 10.2.161.187 preference 1 monitor probe icmp-ping
max registrations 5
voice-class codec 1
!
voice register pool 4
id network 10.2.161.0 mask 255.255.255.0
number 1 94... preference 1
```

```
preference 5
cor incoming everywhere default
cor outgoing everywhere default
proxy 10.2.161.187 preference 1
max registrations 2
voice-class codec 1
!
! Configures translation rules to be applied in the voice register pools.
!
translation-rule 1
Rule 0 94 91
!
! Sets up proxy monitoring.
!
call fallback active
!
dial-peer cor custom
name 95
name 94
name 91
!
! Configures COR values to be applied to the voice register pool.
!
dial-peer cor list call95
member 95
!
dial-peer cor list call94
member 94
!
dial-peer cor list call91
member 91
!
dial-peer cor list everywhere
member 95
member 94
member 91
!
! Configures a voice port and a POTS dial peer for calls to and from the PSTN endpoints.
voice-port 1/0/0
!
dial-peer voice 91500 pots
corlist incoming call91
corlist outgoing call91
destination-pattern 91500
port 1/0/0
!
```

APPENDIX B

Appendix B: Integrating Cisco Unified Communications Manager and Cisco Unified SRST to Use Cisco Unified SRST as a Multicast MOH Resource

This chapter describes how to configure Cisco Unified CM and Cisco Unified SRST to allow Cisco Unified CM to use Cisco Unified SRST gateways as multicast music-on-hold (MOH) resources during fallback and normal Cisco Unified CM operation. A distributed MOH design with local gateways providing MOH eliminates the need to stream MOH across a WAN and saves bandwidth.

Prerequisites for Using Cisco Unified SRST Gateways as a Multicast MOH Resource

- Multicast MOH for H.323 and MGCP is supported on Cisco Unified CM 3.1.1 and higher versions.
- Cisco Unified CM must be configured as follows:
  - With multicast MOH enabled.
  - With Media Resource Groups (MRGs) and Media Resource Group Lists (MRGLs) controlling which devices receive multicast MOH and which devices receive unicast MOH.
  - With Cisco Unified CM regions assigned so that G.711 is used whenever a Cisco Unified SRST multicast MOH resource is invoked.
- The Cisco Unified SRST gateways must run on Cisco Unified SRST 3.0 on Cisco IOS Release 12.2(15)ZJ2 or a later release.

- Cisco Unified SRST must be registered to Cisco Unified CM using protocol such as H.323, MGCP, or SIP.
- For branches that do not run Cisco Unified SRST, Cisco Unified CM multicast MOH packets must cross the WAN. To accomplish this, you must have multicast routing enabled in your network. For more information about multicast routing, see the "IP Multicast" section of Cisco IOS IP Configuration Guide, Release 12.4T.
- With Cisco IOS earlier than 12.3(14)T, configure Cisco Unified SRST as your MGCP gateway's fallback mode using the **ccm-manager fallback-mgcp** and **call application alternate** commands. With Cisco IOS releases after 12.3(14)T, the **ccm-manager fallback-mgcp** and **service** commands must be configured. Configuring these two commands allows Cisco Unified SRST to assume control over the voice port and over call processing on the MGCP gateway. A complete configuration describing setting up Cisco Unified SRST as your fallback mode is shown in Cisco Unified Communications Manager Administration Guide, Release 5.1(3) Survivable Remote Site Telephony Configuration.

Restrictions for Using Cisco Unified SRST Gateways as a Multicast MOH Resource

- Cisco Unified SRST multicast MOH does not support unicast MOH.
- Only a single Cisco Unified CM audio source can be used throughout the network. However, the audio files on each Cisco Unified SRST gateway's flash memory can be different.
- Cisco Unified SRST multicast MOH supports G.711 only.
- Unified SRST multicast MOH does not support co-location of tunnels on the same device.
- Multicast MOH support for H.323 is unavailable in all versions of Cisco Unified Communications Manager 3.3.2. For more information, see CSCdz00697 using the Bug Toolkit.
- In the Cisco IOS Release 12.2(15)ZJ image for Cisco 1700 series gateways, Cisco Unified SRST multicast MOH does not include support for H.323 mode.

Information About Using Cisco Unified SRST Gateways as a Multicast MOH Resource

To configure Cisco Unified SRST gateways as an MOH resource, you should understand the following concepts:

Cisco Unified SRST Gateways and Cisco Unified Communications Manager

Cisco Unified SRST gateways can be configured to multicast Real-Time Transport Protocol (RTP) packets from flash memory during fallback and normal Cisco Unified CM operation. To make this happen, Cisco Unified Communications Manager must be configured for multicast MOH so that the audio packets do not cross the WAN. Instead, audio packets are broadcast from the flash memory of Cisco Unified SRST gateways to the same multicast MOH IP address and port number configured for Cisco Unified Communications

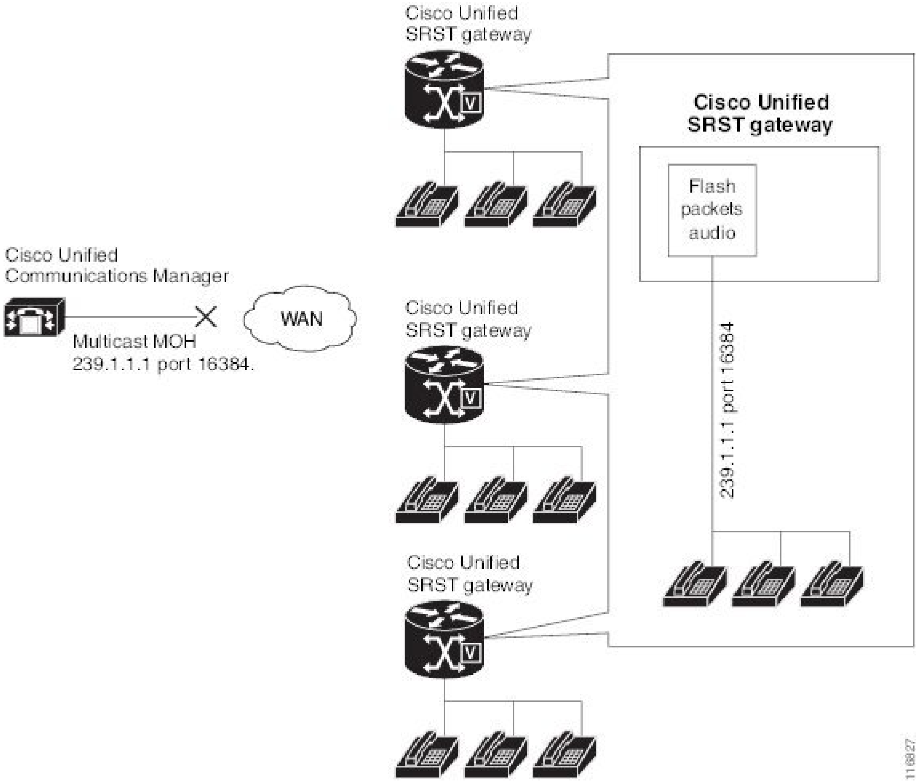
Manager multicast MOH. IP phones at remote sites are able to pick up RTP packets that are multicast from the local branch gateways instead of from the central Cisco Unified CM.

Multicast MOH for PSTN callers is supported when the Cisco Unified SRST router is used as the Cisco IOS voice gateway for Cisco Unified CM. In this state the Cisco Unified SRST function of the router remains in standby mode (no phones registered) with call control of the phones and gateway provided by Cisco Unified Communications Manager. This feature does not apply when the Cisco Unified SRST router is in fallback mode (phones are registered to Cisco Unified SRST). Instead, MOH is provided to PSTN callers via a direct internal path rather than through the multicast loopback interface.

The following figure shows a sample configuration in which all phones are configured by Cisco Unified Communications Manager to receive multicast MOH through port number 16384 and IP address 239.1.1.1. Cisco Unified CM is configured so that multicast MOH cannot reach the WAN, and local Cisco Unified SRST gateways are configured to send audio packets from their flash files to port number 16384 and IP address 239.1.1.1. Cisco Unified CM and the IP phones are spoofed and behave as if Cisco Unified CM were originating the multicast MOH.

Note Phone users at the central site would use multicast MOH from the central site.

Figure 9: Multicast MOH from Cisco Unified SRST Flash Memory

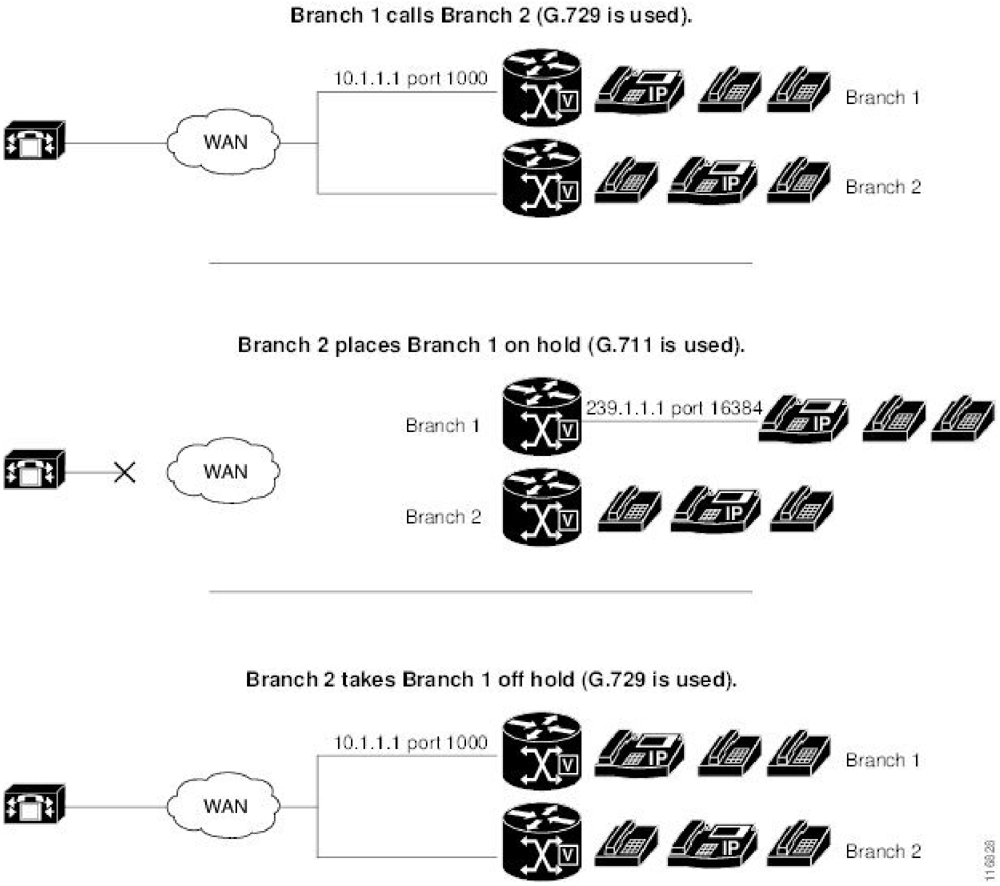


Codecs, Port Numbers, and IP Addresses

Cisco Unified SRST multicast MOH supports G.711 only. shows an example in which G.711 is the only codec used by a central Cisco Unified CM and three branches. In some cases, a Cisco Unified CM system may use additional codecs. For example, for bandwidth savings, Cisco Unified CM may use G.711 for multicast MOH and G.729 for phone conversations.

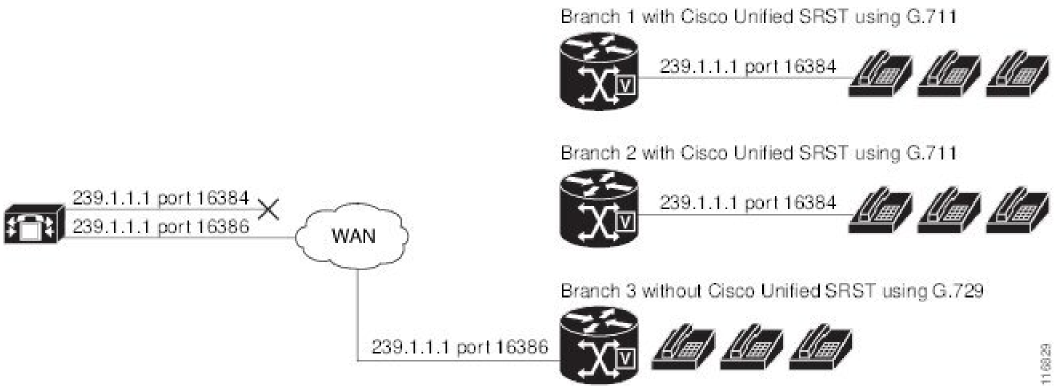
As shown in the example in, IP address 10.1.1.1 and port 1000 are used during phone conversations when G.729 is in use, and IP address 239.1.1.1 and port 16384 are used when a call is placed on hold and G.711 is in use.

Figure 10: IP Address and Port Usage for G.711 and G.729 Configuration


Branch 1 calls Branch 2 (G.729 is used).
10.1.1.1 port 1000
WAN
IP
Branch 1
IP
Branch 2
Branch 2 places Branch 1 on hold (G.711 is used).
Branch 1
239.1.1.1 port 16384
IP
WAN
Branch 2
IP
Branch 2 takes Branch 1 off hold (G.729 is used).
10.1.1.1 port 1000
WAN
IP
Branch 1
IP
Branch 2
116828

The figure 1 and figure 2 shows all branches using Cisco Unified SRST multicasting MOH.The figure 3 shows a case in which some gateways are configured with Cisco Unified SRST and other gateways are not. When the central site and Branch 3 phone users are put on hold by other IP phones in the Cisco Unified CM system, MOH is originated by Cisco Unified CM. When Branch 1 and Branch 2 phone users are put on hold by other phone users in the Cisco Unified CM system, MOH is originated by the Cisco Unified SRST gateways.

Figure 11: MOH Sources for Cisco Unified SRST and Other Unified SRST IP Phones Using MOH


Branch 1 with Cisco Unified SRST using G.711
239.1.1.1 port 16384
Branch 2 with Cisco Unified SRST using G.711
239.1.1.1 port 16384
239.1.1.1 port 16384
239.1.1.1 port 16386
WAN
Branch 3 without Cisco Unified SRST using G.729
239.1.1.1 port 16386
116829

To enable MOH audio packet transmission through two paths, the Cisco Unified CM MOH server must be configured with either one IP address and two different port numbers or one port address and two different IP multicast addresses so that one set of branches can use Cisco Unified SRST multicast MOH and the other can use Cisco Unified CM multicast MOH.

Multicast MOH Transmission

If Cisco Unified SRST multicast MOH is supported by all branches in a system, such as in the figure 1, Cisco Unified Communications Manager must be configured to keep all multicast MOH audio packets from reaching the WAN. When there is a mix of Cisco Unified SRST branches, as shown in the figure 3, one set of Cisco Unified Communications Manager MOH audio files must reach the WAN and another set must not. Audio packets from the central Cisco Unified Communications Manager must cross the WAN to reach branches running Cisco Unified Communications Manager. For branches running Cisco Unified SRST, the packets must not reach the WAN. For more information about Multicast MOH, see the Configuring Cisco Unified SRST for Multicast MOH from an Audio File section.

MOH from a Live Feed

MOH live feed is an SRST feature that provides MOH streams to IP phones from an audio device connected to a local E&M (ISR G2) or FXO (ISR G2/G3) port, or from a remote gateway. Live audio is fed continuously from a fixed source to the MOH playout buffer instead of being read from a flash file.

Live feed audio can also be streamed via multicast to compatible devices. For more information, see Configuring Cisco Unified SRST for MOH from a Live Feed section.

MOH from Flash Files

The MOH Multicast from Flash Files feature facilitates the continuous multicast of MOH audio feed from files in the flash memories of Cisco Unified SRST branch office routers during Cisco Unified Communications fallback and normal Cisco Unified Communications service. Multicasting MOH from individual branch routers saves WAN bandwidth by eliminating the need to stream MOH audio from central offices to remote branches.

The MOH Multicast from Flash Files feature can act as a backup mechanism to the MOH live feed feature. Using the Flash to backup the live-feed is the recommend method rather than using just the live feed feature alone.

Cisco Unified Communications Manager MOH audio files must reach the WAN and another set must not. Audio packets from the central Cisco Unified CM must cross the WAN to reach branches running Cisco Unified CM. For branches running Cisco Unified SRST, the packets must not reach the WAN.

The following table provides a summary of options for MOH.

Audio Source	Description	How to Configure
Flash memory	No external audio input is required.	Configuring Cisco Unified SRST for Multicast MOH from an Audio File
Live feed	The multicast audio stream has minimal delay for local IP phones. The MOH stream for PSTN callers is delayed by a few seconds. If the live feed audio input fails, callers on hold hear silence.	Configuring Cisco Unified SRST for MOH from a Live Feed
Live feed and flash memory	The live feed stream has a few seconds of delay for both PSTN and local IP phone callers. The flash MOH acts as backup for the live-feed MoH. We recommend this option if you want live-feed because it provides guaranteed MOH if the live-feed input is not found or fails.	Configuring Cisco Unified SRST for Multicast MOH from an Audio File and Configuring Cisco Unified SRST for MOH from a Live Feed

How to Use Cisco Unified SRST Gateways as a Multicast MOH Resource

For Cisco Unified CM 8.0 or later, see the Configuring MOH-groups for Cisco Unified SRST (fallback) section in the Cisco Unified Survivable Remote Site Telephony 8.0 Music On Hold Enhancement document.

To use Cisco Unified SRST gateways as a multicast MOH resource, perform the following tasks:

Configuring Cisco Unified Communications Manager for Cisco Unified SRST Multicast MOH

The following sections describe the Cisco Unified CM configuration tasks for Cisco Unified SRST multicast MOH:

- Configuring the MOH Audio Source to Enable Multicasting
- Enabling Multicast on the Cisco Unified Communications Manager MOH Server and Configuring Port Numbers and IP Addresses
- Creating an MRG and an MRGL, Enabling MOH Multicast, and Configuring Gateways
- Creating a Region for the MOH Server
- Verifying Cisco Unified Communications Manager Multicast MOH

To use Cisco Unified SRST gateways as multicast MOH resources, you must configure Cisco Unified Communications Manager to multicast MOH to the required branch sites. To accomplish this, you must configure IP addresses, port numbers, the MOH source, and the MOH audio server.

Even though the MOH routing is set up to prevent the Cisco Unified CM-sourced multicast MOH from actually reaching the WAN and the remote phones, the configured Cisco Unified CM MOH IP port and address information are still used by Cisco Unified CM to tell the phones which multicast IP address to listen to for MOH (for the MOH sourced by SRST).

Configuring the MOH server involves designating a maximum number of hops for the audio source. A configuration of one hop keeps Cisco Unified CM multicast MOH packets from reaching the WAN, thus spoofing Cisco Unified CM and allowing Cisco Unified SRST multicast MOH packets to be sent from Cisco Unified SRST gateways to their component phones. For cases in which Cisco Unified CM multicast must reach gateways that do not run Cisco Unified SRST, use the Cisco IOS **ip multicast boundary** command to control where multicast packets go.

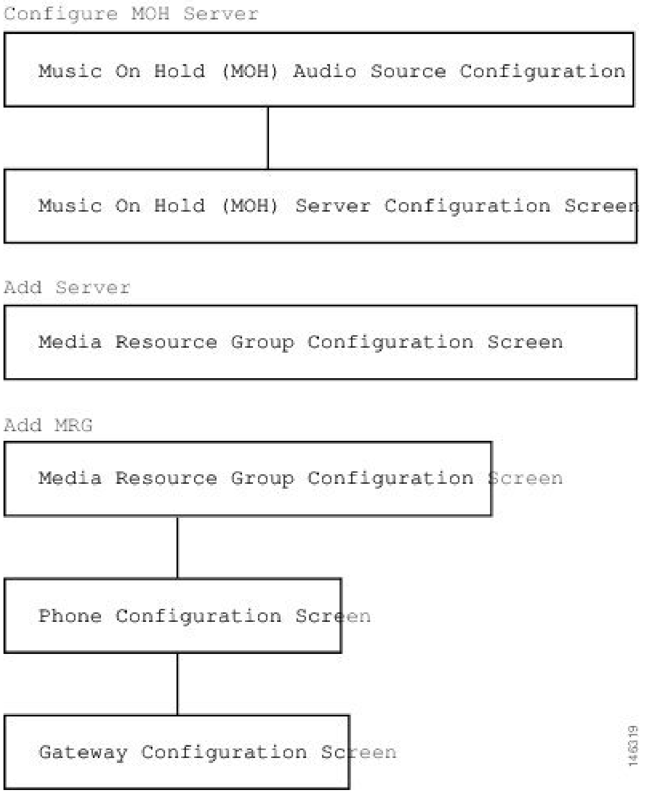
After the MOH server is configured, the MOH server must be added to a Media Resource Group (MRG); the MRG is added to a Media Resource Group List (MRGL); and the designated Cisco Unified CM branch gateways are configured to use the MRGL.

Five Cisco Unified CM windows are used to configure the MOH server, audio source, MRG, MRGL, and individual gateways. The figure 4 provides an overview of this process.

The last Cisco Unified CM configuration task involves creating an MOH region that assigns MOH G.711 codec usage for the central site or sites and branch office or offices.

Regions specify the codecs that are used for audio and video calls within a region and between existing regions. For information about regions, see the “Region Configuration” section in the *Cisco Unified Communications Manager Administration Guide*. From the Cisco Unified Communications Manager documentation directory, click **Maintain and Operate Guides** and select the required Cisco Unified Communications Manager version to locate the administration guide for your version.

Figure 12: Unified Communications Manager Screens for Configuring Multicast MOH


Configure MOH Server
Music On Hold (MOH) Audio Source Configuration
Music On Hold (MOH) Server Configuration Screen
Add Server
Media Resource Group Configuration Screen
Add MRG
Media Resource Group Configuration Screen
Phone Configuration Screen
Gateway Configuration Screen
146319

Configuring the MOH Audio Source to Enable Multicasting

The MOH audio source is a file from which Cisco Unified CM transmits RTP packets. You can create an audio file or use the default audio file. For Cisco Unified SRST multicast MOH, only one audio source can be used, even if, for example, one out of 500 sites uses Cisco Unified SRST multicast MOH. In addition, all Cisco Unified Communications Manager systems must use the same audio source for user and network MOH because Cisco Unified SRST multicast MOH can stream audio only to a single multicast IP address and port. For Cisco Unified SRST multicast MOH, the Cisco Unified Communications Manager audio source file must be configured for G.711 bandwidth.

Tip The simplest way to create an audio source is to use the default audio source.

Whether you use a default Cisco Unified CM MOH audio source or you create one, the MOH audio source must be configured for multicasting in the MOH Audio Source Configuration window.

Note that the MOH Audio Source File Status section shows that the MOH audio source file is configured for four codec formats. If you are planning to use several codecs, ensure that the audio source file accommodates them.

For further information about the creation of an MOH audio source, see the *Cisco Unified Communications Manager Administration Guide*. From the Cisco Unified Communications Manager documentation directory, click **Maintain and Operate Guides** and select the required Cisco Unified CM version.

Use this procedure to configure the MOH audio source to enable multicasting and continuous play.

Note These instructions assume that an MOH audio source file was already created.

SUMMARY STEPS

1. To enable multicast MOH for the MOH audio source, choose **Service** > **Media Resources** > **Music On Hold Audio Source** to display the MOH Audio Source Configuration window.
2. Double-click the required audio source listed in the MOH Audio Sources column.
3. In the MOH Audio Source Configuration window, check **Allow Multicasting**.
4. Click **Update**.

DETAILED STEPS

	Command or Action	Purpose
Step 1	To enable multicast MOH for the MOH audio source, choose **Service** > **Media Resources** > **Music On Hold Audio Source** to display the MOH Audio Source Configuration window.	
Step 2	Double-click the required audio source listed in the MOH Audio Sources column.	
Step 3	In the MOH Audio Source Configuration window, check **Allow Multicasting**.	
Step 4	Click **Update**.	

Enabling Multicast on the Cisco Unified Communications Manager MOH Server and Configuring Port Numbers and IP Addresses

Enter a base multicast IP address and port number in the Multicast Audio Source Information section of the MOH Server Configuration window. If you are using Cisco Unified CM multicast MOH and Cisco Unified SRST multicast MOH (see the Codecs, Port Numbers, and IP Addresses section and the Multicast MOH Transmission section), you must select a port and IP address increment method to configure for two sets of port numbers and IP address.

If the Increment Multicast on radio button is set to IP address, each MOH audio source and codec combination is multicast to different IP addresses but uses the same port number. If it is set to Port Number, each MOH audio source and codec combination is multicast to the same IP address but uses different destination port numbers.

Table 2 shows the difference between incrementing on an IP address and incrementing on a port number, using the base IP address of 239.1.1.1 and the base port number of 16384. The table also matches Cisco Unified Communications Manager audio sources and codecs to IP addresses and port numbers.

Table 4: Example of the Differences Between Incrementing Multicast on IP Address and Incrementing Multicast on Port Number

Audio Source	Codec	Increment Multicast on IP Address		Increment Multicast on Port Number	
		Destination IP Address	Destination Port	Destination IP Address	Destination Port
1	G.711 mu-law	239.1.1.1	16384	239.1.1.1	16384
1	G.711 a-law	239.1.1.2	16384	239.1.1.1	16386
1	G.729	239.1.1.3	16384	239.1.1.1	16388
1	Wideband	239.1.1.4	16384	239.1.1.1	16390
2	G.711 mu-law	239.1.1.5	16384	239.1.1.1	16392
2	G.711 a-law	239.1.1.6	16384	239.1.1.1	16394
2	G.729	239.1.1.7	16384	239.1.1.1	16396
2	Wideband	239.1.1.8	16384	239.1.1.1	16398

Note The lower destination port 16384 is assigned to the first multicast-enabled audio source ID, and the subsequent ports will be assigned to the subsequent multicast-enabled audio sources.

Incrementation is triggered by a change in codec usage. When codec usage changes, a new IP address or port number (depending on the incrementation selected) is assigned to the new codec type and is put intouse. The original codec keeps its IP address and port number. For example, as seen in Table 2, if your baseline IP address and port number are 239.1.1.1 and 16384 for a G.711 mu-law codec and the codec usage changes to G.729 (triggering an increment on the port number), the IP address and port number in use changes, or increment, to 239.1.1.1 and 16386. If G.711 usage resumes, the IP address and port number returns to 239.1.1.1 and 16384. If G.729 is in use again, the IP address and port goes back to 239.1.1.1 and 16386, and so forth.

It is important to configure a Cisco Unified CM port number and IP address that use a G.711 audio source for Cisco Unified SRST multicast MOH. If Cisco Unified CM multicast MOH is also being used on gateways that do not have Cisco Unified SRST and use a different codec, such as G.729, ensure that the additional or incremental port number or IP address uses the same audio source as the Cisco Unified SRST gateways and the required codec.

The MOH Server Configuration window is also where the multicast audio source for the MOH server is configured. For Cisco Unified SRST multicast MOH, the Cisco Unified CM MOH server can use only one audio source. An audio source is selected by inputting the audio source’s maximum number of hops.

The Max Hops configuration sets the length of the transmission of the audio source packets. Limiting the number of hops is one way to stop audio packets from reaching the WAN and thus spoofing Cisco Unified Communications Manager so Cisco Unified SRST can multicast MOH. If all of your branches run Cisco Unified SRST, use a low number of hops to prevent audio source packets from crossing the WAN. If your system configuration includes routers that do not run Cisco Unified SRST, enter a high number of hops to allow source packets to cross the WAN. Use the ip multicast bounder and access-list commands to keep resource packets from specific IP addresses from reaching the WAN.

Use this procedure to enable multicast and configure port numbers and IP addresses.

SUMMARY STEPS

1. Enable multicast MOH for Cisco Unified CM
2. Set the base IP address and port number.
3. Select whether Cisco Unified CM increments port numbers or IP addresses.
4. Enter a maximum number of hops.
5. Use Cisco IOS commands to stop Cisco Unified CM signals from crossing the WAN and reaching Cisco Unified SRST gateways.

DETAILED STEPS

	Command or Action	Purpose
Step 1	Enable multicast MOH for Cisco Unified CM	
Step 2	Set the base IP address and port number.	In the MOH Server Configuration window, enter an IP address in the Base Multicast IP Address field and enter a port number in the Base Multicast Port Number field. Ensure that the IP address and port number use the required audio source and codec. See Table 2.
Step 3	Select whether Cisco Unified CM increments port numbers or IP addresses.	In the MOH Server Configuration window, in the Increment Multicast on field, choose Port Number if you want port numbers to be incremented and the IP address to remain unchanged. Choose IP Address if you want IP addresses to be incremented and the port number to remain unchanged. • If all of your branches run Cisco Unified SRST and thus use G.711 for MOH, use either settingbecause incrementation does not take place and a selection does not matter. • If your system configuration includes routers that do not run Cisco Unified SRST and use a different codec, select an incrementation method. **Note** If your branches include routers that do not run Cisco Unified SRST and do use G.711, configure separate audio sources: one for the routers that run Cisco Unified SRST and one for the routers that do not.
Step 4	Enter a maximum number of hops.	In the MOH Server Configuration window, next to the Audio Source Name field, enter 1 in the Max Hops field if all of your branches run Cisco Unified SRST. If your system configuration includes routers that do not run Cisco Unified SRST, enter 16 in the Max Hops field.
Step 5	Use Cisco IOS commands to stop Cisco Unified CM signals from crossing the WAN and reaching Cisco Unified SRST gateways.	If all of your branches run Cisco Unified SRST, skip this step. If your system configuration includes routers that do not run Cisco Unified SRST and use a different codec, enter

	Command or Action	Purpose
		the following Cisco IOS commands starting from global configuration mode on the central site router:

Creating an MRG and an MRGL, Enabling MOH Multicast, and Configuring Gateways

The next task involves configuring individual gateways to use an MOH server that can transport the required MOH audio source to their IP phones on hold. This is accomplished by creating a Media Resource Group (MRG). An MRG references media resources, such as MOH servers. The MRG is then added to a Media Resource Group List (MRGL), and the MRGL is added to the phone and gateway configurations.

MRGs are created in the Media Resource Group Configuration window. MRGLs are created in the Media Resource Group List Configuration window. Phones are configured in the Phone Configuration window. Gateways are configured in the Gateway Configuration window.

Note The Gateway Configuration window for an H.323 gateway is similar for MGCP gateways.

Add MRGL to a gateway or IP phone configuration by adding the MRGL to a device pool configuration. For further information about device pools, see *Cisco Unified Communications Manager Administration Guide*. From the Cisco Unified Communications Manager documentation directory, click **Maintain and Operate Guides** and select the required Cisco Unified CM version.

Use the following procedure to create an MRG and MRGL, to enable MOH multicast, and to configure gateways.

SUMMARY STEPS

1. Create an MRG with a multicast MOH media resource.
2. Create an MRGL that contains the newly created MRG.
3. Add the MRGL to the required IP phones.
4. Add the MRGL to the required gateway.

DETAILED STEPS

	Command or Action	Purpose
Step 1	Create an MRG with a multicast MOH media resource.	
Step 2	Create an MRGL that contains the newly created MRG.	
Step 3	Add the MRGL to the required IP phones.	
Step 4	Add the MRGL to the required gateway.	

Creating a Region for the MOH Server

To ensure that the MOH server uses G.711 for Cisco Unified SRST gateways, you must create a separate region for the MOH server. For more information about codecs, see the Codecs, Port Numbers, and IP Addresses section. For information about regions, see Cisco Unified Communications Manager Administration Guide. From the Cisco Unified Communications Manager documentation directory, click **Maintain and Operate Guides** and select the required Cisco Unified Communications Manager version.

Configure the Region Configuration window. If the Cisco Unified CM system uses G.711 only, all of the central sites and their constituent branches for the MOH region must be set to G.711. If a Cisco Unified CM system has a combination of branches that do and do not run Cisco Unified SRST multicast MOH and the branches that do not run Cisco Unified SRST require a different codec for Cisco Unified Communications Manager multicast MOH, they must be configured accordingly.

A Region Configuration window where the "MOH Server" region is configured to use the G.711 and G.729 codecs might look like this:

- G.711 is used for Branch 1 because its gateway is configured to run Cisco Unified SRST multicast MOH, which requires G.711.
- G.729 is used for Branch 2 because its gateway doe not run Cisco Unified SRST and it is configured to use a port and IP address that use G.729.
- G.711 is configured for the central site and the MOH server region.

Use the following procedure to create a region for the MOH server.

SUMMARY STEPS

1. Create an MOH server region.
2. Create other regions as needed for different codecs.

DETAILED STEPS

	Command or Action	Purpose
Step 1	Create an MOH server region.	
Step 2	Create other regions as needed for different codecs.	

Verifying Cisco Unified Communications Manager Multicast MOH

The Cisco Unified CM multicast MOH configuration must run correctly for Cisco Unified SRST multicast MOH to work. Verification of Cisco Unified Communications Manager multicast MOH differs for configurations using a WAN with multicast enabled and a WAN with multicast disabled.

You must verify that the Cisco Unified CM multicast MOH is provided through multicasting and not unicasting. Because unicast MOH is enabled by default, it is easy to mistakenly conclude that multicast MOH is working when it is not.

SUMMARY STEPS

1. Verify that Cisco Unified CM system's multicast MOH is heard on a remote gateway.
2. Verify that the Cisco Unified CM system's MOH is multicast, not unicast.

DETAILED STEPS

	Command or Action	Purpose
Step 1	Verify that Cisco Unified CM system's multicast MOH is heard on a remote gateway.	

	Command or Action	Purpose
Step 2	Verify that the Cisco Unified CM system's MOH is multicast, not unicast.	

Configuring Cisco Unified SRST for Multicast MOH from an Audio File

Note Use the steps in this section only when you are using Microsoft Windows to run Cisco Unified Communications Manager version 4.3 or below. Use the RTMT (Real-Time Monitoring Tool) in Cisco Unified Communications Manager version 5.0 and later versions on the Linux operating system to monitor MOH activity in Cisco Unified CM version. See Cisco Unified Communications Serviceability System Guide, Release 4.0(1) for more information about RTMT.

Use the following procedures to configure Cisco Unified SRST for multicast MOH from an audio file.

Prerequisites

- The Cisco Unified SRST gateways must run Cisco IOS Release 12.2(15)ZJ2 or a later release.
- The flash memory in each of the Cisco Unified SRST gateways must have an MOH audio file. The MOH file can be in .wav or .au file format, but must contain 8-bit 8-kHz data, such as an a-law or mu-law data format. A known working MOH audio file (music-on-hold.au) is included in the program .zip files that can be downloaded from http://www.cisco.com/cgi-bin/tablebuild.pl/ip-key. Or the music-on-hold.au file can be downloaded from http://www.cisco.com/cgi-bin/tablebuild.pl/ip-iostsp and copied to the flash memory on your Cisco Unified SRST router.

Note The MOH file packaged with the SRST software is completely royalty free.

- For Cisco Unified CM versions 4.3 or earlier versions running on Windows, download MOH files by copying one of the MOH files, such as SampleAudioSource.ULAW.wav, from C:\Program Files\Cisco\MOH on Cisco Unified CM.

Note During the copying process, four files are added to each router's flash automatically. One of the files must use a mu-law format as indicated by the extension.ULAW.wav.

- You must configure a loopback interface and include its IP addresses in the Cisco Unified SRST multicast MOH configuration. This configuration allows multicast MOH to be heard on POTS ports on the gateway. The loopback interface does not have to bind to either H.323 or MGCP.
- Configure at least one ephone and directory number (DN), even if the gateway is not used for Cisco Unified SRST. Cisco Unified SRST multicast MOH streaming never starts without an ephone and directory number.

Enabling Multicast MOH on the Cisco Unified SRST Gateway

No multicast MOH routing configuration is required for Cisco Unified SRST gateways because each Cisco Unified SRST gateway is configured to act as a host running an application that streams multicast MOH packets from the network. The multicast moh command declares the Cisco Unified Communications Manager multicast MOH address and port number and allows Cisco Unified SRST gateways to route MOH from flash memory to up to four IP addresses. If no route IP addresses are configured, the flash MOH is sent through the IP address configured in the Cisco Unified SRST ip source-address command.

SUMMARY STEPS

1. **ccm-manager music-on-hold**
2. **interface loopback** *number*
3. **ip address** *ip-address mask*
4. **exit**
5. **interface fastethernet** *slot/port*
6. **ip address** *ip-address mask*
7. **exit**
8. **call-manager-fallback**
9. **ip source-address** *ip-address* [**port** *port*
10. **max-ephones** *max-phones*
11. **max-dn** *max-directory-number*
12. **moh** *filename*
13. **multicasting-enabled**
14. **multicast moh** *multicast-address***port** *port* [**route** *ip-address-list*]
15. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**ccm-manager music-on-hold** **Example:** `Router(config)# ccm-manager music-on-hold`	Enables the multicast MOH feature on a voice gateway.
Step 2	**interface loopback** *number* **Example:** `Router(config)# interface loopback 1`	Configures an interface type and enters theinterface configuration mode. *number* —Loopback interface number. The range is from 0 to 2147483647.
Step 3	**ip address** *ip-address mask* **Example:** `Router(config-if)# ip address 10.1.1.1 255.255.255.255`	Sets a primary IP address for an interface. • *ip-address*—IP address. • *mask*—Mask for the associated IP subnet.
Step 4	**exit** **Example:** `Router(config-if)# exit`	Exits interface configuration mode.

	Command or Action	Purpose
Step 5	**interface fastethernet** *slot/port* **Example:** `Router(config)# interface fastethernet 0/0`	(Optional if the route keyword is not used in the **multicast moh** command. See Step 9 and Step 13.) Configures an interface type and enters interface configuration mode.
Step 6	**ip address** *ip-address mask* **Example:** `Router(config-if)# ip-address 172.21.51.143 255.255.255.192`	(Optional if the route keyword is not used in the **multicast moh** command. See Step 9 and Step 13.) Sets a primary IP address for an interface.
Step 7	**exit** **Example:** `Router(config-if)# exit`	(Optional if the route keyword is not used in the **multicast moh** command. See Step 9 and Step 13.) Exits interface configuration mode.
Step 8	**call-manager-fallback** **Example:** `Router(config)# call-manager-fallback`	Enters call-manager-fallback configuration mode.
Step 9	**ip source-address** *ip-address* [**port** *port* **Example:** `Router(config-cm-fallback)# ip source-address 172.21.51.143 port 2000`	(Optional if the route keyword is not used in the **multicast moh** command. See Step 13.) Enables a router to receive messages from Cisco Unified IP phones through the specified IP addresses and ports. • *ip-address*—The pre-existing router IP address, typically one of the addresses of the Ethernet port of the router. • **port** *port*—(Optional) The port to which the gateway router connects to receive messages from the Cisco Unified IP phones. The port number range is from 2000 to 9999. The default port number is 2000.
Step 10	**max-ephones** *max-phones* **Example:** `Router(config-cm-fallback)# max-ephones 1`	Configures the maximum number of Cisco Unified IP phones that can be supported by a router. *max-phones*—Maximum number of Cisco IP phones supported by the router. The maximum number is platform-dependent. The default is 0.
Step 11	**max-dn** *max-directory-number* **Example:** `Router(config-cm-fallback)# max-dn 1`	Sets the maximum possible number of virtual voice ports that can be supported by a router. *max-directory-number* —Maximum number of directory numbers or virtual voice ports supported by the router. The maximum possible number is platform-dependent. The default is 0.
Step 12	**moh** *filename* **Example:** `Router(config-cm-fallback)# moh music-on-hold.au`	Enables use of an MOH file. *filename*—Filename of the music file. The music file must reside in flash memory.

	Command or Action	Purpose
Step 13	**multicasting-enabled**	Selects the multicast-enabled MOH audio source in the User Hold MOH Audio Source field on the Phone Configuration page in Cisco Unified CM Administration GUI.

Step 14	**multicast moh** *multicast-address***port** *port* [**route** *ip-address-list*]
	Example:
	`Router(config-cm-fallback)# multicast moh 239.1.1.1`
	`port 16386 route 239.1.1.2 239.1.1.3 239.1.1.4 239.1.1.5`
		Enables multicast of MOH from a branch office flash MOH file to IP phones in the branch office.
		• *multicast-address***port** *port* —Declares the IP address and port number of MOH packets that are to be multicast. The multicast IP address and port must match the IP address and the port number that Cisco Unified CM is configured to use for multicast MOH. If you are using different codecs for MOH, these might not be the base IP address and port, instead an incremented IP address or port number. See the Configuring the MOH Audio Source to Enable Multicasting section. If you have multiple audio sources configured on Cisco Unified CM, ensure that you are using the audio sources's correct IP address and port number.
		• **route** —(Optional) List of explicit router interfaces for the IP multicast packets.
		• *ip-address-list*—(Optional) List of up to four explicit routes for multicast MOH. The default is that the MOH multicast stream is automatically output on the interfaces that correspond to the address that was configured with the ip source-address command.

Step 15	**exit** **Example:** `Router(config-cm-fallback)# exit`	Exits call-manager-fallback configuration mode.

Verifying Basic Cisco Unified SRST Multicast MOH Streaming

Use the following procedure to verify that multicast MOH packets are configured with the **multicast moh** command.

SUMMARY STEPS

1. **debug ephone moh**
2. **show interfaces fastethernet**
3. **show ephone summary**

DETAILED STEPS

	Command or Action	Purpose

Step 1	**debug ephone moh**
	Example:
	`Router# debug ephone moh`
	`!`
	`MOH route If FastEthernet0/0 ETHERNET 172.21.51.143 via ARP`
	`MOH route If Loopback0 46 172.21.51.98 via 172.21.51.98`
	`!`
		This command sets debugging for MOH. You can use this command to show that the Cisco Unified SRST gateway is multicasting MOH out of Loopback 0 and Fast Ethernet 0/0.

Step 2	**show interfaces fastethernet**
	Example:
	`Router# show interfaces fastethernet 0/0`
	`!`
	`30 second output rate 86000 bits/sec, 50 packets/sec`
	`!`
		Use this command to confirm that the interface output rates match one G.711 stream, which the show interfaces fastethernet output displays as 50 packets/sec and 80 kbps or more.

Step 3	**show ephone summary**
	Example:
	`Router# show ephone summary`
	`!`
	`File music-on-hold.au type AU Media_Payload_G.711Ulaw64k 160 bytes`
	`!`
		Use this command to verify that the Cisco IOS software was able to read the MOH audio file successfully.

Verifying Cisco Unified SRST MOH to PSTN

Use the following procedure to verify Cisco Unified CM control of MOH (the WAN link is up) and that multicast MOH packets transmit over a public switched telephone network (PSTN).

Note This feature does not apply when the Cisco Unified SRST router is in fallback mode.

SUMMARY STEPS

1. Verify that a PSTN caller hears MOH when placed on hold by an IP phone caller. Use a Cisco Unified SRST gateway IP phone to call a PSTN phone, and put the PSTN caller on hold. The PSTN caller should hear MOH.
2. **show ccm-manager music-on-hold**
3. **debug h245 asn**
4. **show call active voice**

DETAILED STEPS

	Command or Action	Purpose
Step 1	Verify that a PSTN caller hears MOH when placed on hold by an IP phone caller. Use a Cisco Unified SRST gateway IP phone to call a PSTN phone, and put the PSTN caller on hold. The PSTN caller should hear MOH.	

Step 2	**show ccm-manager music-on-hold**
	Example:
	`Router# show ccm-manager music-on-hold`
	`Current active multicast sessions : 1`
	`Multicast RTP port Packets    Call Codec Incoming`
	`Address   number   in/out     id         Interface`
	`==================================================`
	`239.1.1.1 16384    326/326    42         G.711ulaw`
	` Lo0`
		Use this command to verify that the MOH is multicast if you are using Windows and Cisco Unified CM version 4.3 or an earlier version.
		Note that the **show ccm-manager music-on-hold** command displays information about PSTN connections on hold only. It does not display information about multicast streams going to IP phones on hold. The following is an example of **show ccm-manager music-on-hold** command output.
		If the PSTN caller hears MOH, and the **show ccm-manager music-on-hold** command displays no active multicast streams, the MOH is unicast. Confirm this by checking the MOH performance counters as discussed in the Verifying Cisco Unified Communications Manager Multicast MOH section.

Step 3	**debug h245 asn**
	Example:
	`Router# debug h245 asn`
	`*Mar 1 04:20:19.227: H245 MSC INCOMING PDU ::=`
	`value MultimediaSystemControlMessage ::= response`
	` :`
	`openLogicalChannelAck :`
	`{`
	`forwardLogicalChannelNumber 6`
	`forwardMultiplexAckParameters`
	`h2250LogicalChannelAckParameters :`
	`{`
	`sessionID 1`
	`mediaChannel unicastAddress : iPAddress :`
	`{`
	`network 'EF010101'H`
	`tsapIdentifier 16384`
	`}`
	`mediaControlChannel unicastAddress : iPAddress :`
	`{`
	`network 'EF010101'H`
	`tsapIdentifier 16385`
	`}`
	`}`
	`}`
		Use this command if H.323 is being used and no multicast address appears in the **show ccm-manager music-on-hold** command output to verify the H.323 handshaking between Cisco Unified Communications Manager and the Cisco Unified SRST gateway. When a PSTN caller is placed on hold, Cisco Unified Communications Manager sends an H.245 closeLogicalChannel, followed by an openLogicalChannel. Verify that the final openLogicalChannelAck from Cisco Unified Communications Manager to the Cisco Unified SRST gateway contains the expected multicast IP address and port number. In the following example, the IP address is EF010101 (239.1.1.1) and the port number is 16384.

Step 4	**show call active voice**
	Example:
	`Router# show call active voice | include`
	`RemoteMedia`
		Use this command with the debug h245 asn command to further verify the H.323 handshaking between Cisco Unified Communications Manager and the Cisco Unified SRST gateway.

	Command or Action	Purpose

	`RemoteMediaIPAddress=239.1.1.1`
	`RemoteMediaPort=16384`
		The IP address and port number displayed must match the IP address and port number displayed by the debug h245 asn command. If the RemoteMediaIPAddress field displays 0.0.0.0, you probably have encountered caveat CSCdz00697. For more information, see the Cisco Bug ToolKit and the Restrictions for Using Cisco Unified SRST Gateways as a Multicast MOH Resource section.

Verifying Cisco Unified SRST Multicast MOH to IP Phones

To verify that Cisco Unified CM is signaling the IP phone to receive Cisco Unified SRST multicast MOH correctly, perform the following steps.

SUMMARY STEPS

1. Verify that an IP phone caller hears MOH when placed on hold by an IP phone caller.
2. Check the MOHMulticastResourceActive and MOHUnicastResourceActive counters.

DETAILED STEPS

	Command or Action	Purpose
Step 1	Verify that an IP phone caller hears MOH when placed on hold by an IP phone caller.	Use an IP phone to call a second IP phone, and put the second caller on hold. The second caller should hear MOH.
Step 2	Check the MOHMulticastResourceActive and MOHUnicastResourceActive counters.	Use the Performance window to check the MOHMulticastResourceActive and MOHUnicastResourceActive counters under the Cisco MOH Device performance object. See Step 2 in the Verifying Cisco Unified Communications Manager Multicast MOH section. For Cisco Unified SRST multicasting MOH to work, the multicast counter must increment.

Troubleshooting Tips

If no MOH is heard and the Cisco Unified SRST MOH signaling is multicasting, connect a sniffer to the PC port on the back of IP phone. If the IP phone and Cisco Unified SRST gateway are connected to the same subnet, multicast RTP packets must be detected at all times, even when the IP phone was not placed on hold. If the IP phone and the Cisco Unified SRST gateway are not connected to the same subnet, multicast RTP packets are detected only when the IP phone is placed on hold and sends an Internet Group Management Protocol (IGMP) Join to the closest router.

Configuring Cisco Unified SRST for MOH from a Live Feed

To configure MOH from a live feed, establish a voice port and dial peer for the call and then create a “dummy” phone or directory number. The dummy number allows for making and receiving calls, and the number is not assigned to a physical phone. It is that number that the MOH system autodials to establish the MOH feed.

The **moh-live** command allocates one of the virtual voice ports from the pool of virtual voice ports created by the **max-dn** command. The virtual voice port places an outgoing call to the dummy number; that is, the directory number specified in the **moh-live** command. The audio stream obtained from the MOH call provides the music-on-hold audio stream.

We recommend that the interface for live-feed MOH is an analog E&M port because it requires the minimum number of external components. Connect a line-level audio feed (standard audio jack) directly to pins 3 and 6 of an E&M RJ-45 connector. The E&M WAN interface card (WIC) has a built-in audio transformer that provides appropriate electrical isolation for the external audio source. (An audio connection on an E&M port does not require loop current.) The **signal immediate** and **auto-cut-through** commands disable E&M signaling on this voice port. A G.711 audio packet stream is generated by a digital signal processor (DSP) on the E&M port.

In Cisco IOS Release 12.4(15)T and later releases, you can directly connect a live-feed source to an FXO port if the **signal loop-start live-feed** command is configured on the voice port; otherwise, the port must connect through an external third-party adapter to provide a battery feed. An external adapter must supply normal telephone company (telco) battery voltage with the correct polarity to the tip and ring leads of the FXO port and it must provide transformer-based isolation between the external audio source and the tip and ring leads of the FXO port.

Music from a live feed is continuously fed into the MOH playout buffer instead of being read from a flash file, so there is typically a 2-second delay. An outbound call to an MOH live-feed source is attempted (or reattempted) every 30 seconds until the connection is made by the directory number that was configured for MOH. If the live-feed source is shut down for any reason, the flash memory source automatically activates.

A live-feed MOH connection is established as an automatically connected voice call that is made by the Cisco Unified SRST MOH system itself or by an external source directly calling in to the live-feed MOH port. An MOH call can be from or to the PSTN or can proceed via VoIP with voice activity detection (VAD) disabled. The call is assumed to be an incoming call unless the **out-call** keyword is used with the **moh-live** command during configuration.

The Cisco Unified SRST router uses the audio stream from the call as the source for the MOH stream, displacing any audio stream that is available from a flash file. An example of an MOH stream received over an incoming call is an external H.323-based server device that calls the directory number to deliver an audio stream to the Cisco Unified SRST router.

The following sections describe the configuration tasks for Cisco Unified SRST MOH live feed:

Prerequisites

Cisco Unified SRST for multicast MOH, as described in the Configuring Cisco Unified SRST for Multicast MOH from an Audio File section, is not required for the MOH live-feed configuration. However, MOH live feed is designed to work in conjunction with multicast MOH.

Restrictions

- An FXO port can be used for a live feed if the port is supplied with an external third-party adapter to provide a battery feed.
- An FXS port cannot be used for a live feed.
- For a live feed from VoIP, VAD must be disabled.
- MOH is supplied to PSTN and VoIP G.711 calls. Some versions of Cisco Unified SRST provide MOH to local phones. On Cisco Unified SRST that do not support MOH for local IP phones, callers hear a repeating tone on hold for reassurance that they are still connected.

- Conditions may occur within your network that is caused by brief spikes of a higher CPU usage. Small spikes in CPU usage can temporarily affect the quality of the MOH heard by parties connected via TDM (FXO / PRI / S) interfaces.

Setting Up the Voice Port on the Cisco Unified SRST Gateway

Use the following procedure to activate MOH from a live feed and to set up and connect the physical voice port.

SUMMARY STEPS

1. **voice-port** *port*
2. **input gain** *decibels*
3. **auto-cut-through**
4. **operation 4-wire**
5. **signal immediate**
6. **no shutdown**
7. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**voice-port** *port* **Example:** `Router(config)# voice-port 1/1/0`	Enters voice-port configuration mode to set up the physical voice port. To find the correct definition of the port argument for your router, see Cisco IOS Survivable Remote Site Telephony Version 3.2 Command Reference. .
Step 2	**input gain** *decibels* **Example:** `Router(config-voice-port)# input gain 0`	Specifies, in decibels, the amount of gain to be inserted at the receiver side of the interface. Acceptable values are integers from –6 to 14.
Step 3	**auto-cut-through** **Example:** `Router(config-voiceport)# auto-cut-through`	(E&M ports only) Enables call completion when a PBX does not provide an M-lead response. MOH requires that you use this command with E&M ports.
Step 4	**operation 4-wire** **Example:** `Router(config-voiceport)# operation 4-wire`	(E&M ports only) Selects the 4-wire cabling scheme. MOH requires that you specify 4-wire operation with this command for E&M ports.
Step 5	**signal immediate** **Example:** `Router(config-voiceport)# signal immediate`	(E&M ports only) For E&M tie trunk interfaces, directs the calling side to seize a line by going off-hook on its E-lead and to send address information as DTMF digits.
Step 6	**no shutdown** **Example:** `Router(config-voiceport)# no shutdown`	Activates the voice port.

	Command or Action	Purpose
Step 7	**exit** **Example:** `Router(config-voiceport)# exit`	Exits voice-port configuration mode.

Setting Up the Directory Numbers on the Cisco Unified SRST Gateway

After setting up the voice port, create a dial peer and give the voice port a directory number with the **destination-pattern** command. The directory number is the number that the system uses to access the MOH.

SUMMARY STEPS

1. **dial-peer voice** *tag***pots**
2. **destination-pattern** *string*
3. **port** *port*
4. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**dial-peer voice** *tag***pots** **Example:** `Router(config)# dial-peer voice 7777 pots`	Enters dial-peer configuration mode.
Step 2	**destination-pattern** *string* **Example:** `Router(config-dial-peer)# destination-pattern 7777`	Specifies the directory number that the system uses to create MOH. This command specifies either the prefix or the full E.164 telephone number to be used for a dial peer.
Step 3	**port** *port* **Example:** `Router(config-dial-peer)# port 1/1/0`	Associates the dial peer with the voice port that was specified in the Setting Up the Voice Port on the Cisco Unified SRST Gateway section.
Step 4	**exit** **Example:** `Router(config-dial-peer)# exit`	Exits dial-peer configuration mode.

Establishing the MOH Feed

Use the following procedure to establish the MOH feed and connect the music source, such as a CD player, to autodial the directory number.

SUMMARY STEPS

1. **call-manager-fallback**
2. **max-dn** *max-directory-number*

3. **multicast moh** *multicast-address***port***port* [**route** *ip-address-list*]
4. **moh-live dn-number** *calling-number***out-call***outcall-number*
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**call-manager-fallback** **Example:** `Router(config)# call-manager-fallback`	Enters call-manager-fallback configuration mode.
Step 2	**max-dn** *max-directory-number* **Example:** `Router(config-cm-fallback)# max-dn 1`	Sets the maximum possible number of virtual voice ports that can be supported by a router. • *max-directory-number*—Maximum number of directory numbers or virtual voice ports supported by the router. The maximum possible number is platform-dependent. The default is 0.

Step 3	**multicast moh** *multicast-address***port***port* [**route** *ip-address-list*]
	Example:
	`Router(config-cm-fallback)# multicast moh 239.1.1.1 port 16386 route 239.1.1.2 239.1.1.3 239.1.1.4 239.1.1.5`
	Example:
	`Router(config-cm-fallback)# multicast moh`
		Enables multicast of MOH from a branch office flash MOH file to IP phones in the branch office.
		Note This command must be used to source live feed MOH to multicast Cisco Unified CM mode. It is not required in strict SRST mode.
		• *multicast-address* and **port** *port* —Declares the IP address and port number of MOH packets that are to be multicast. The multicast IP address and port must match the IP address and the port number that Cisco Unified Communications Manager is configured to use for multicast MOH. If you are using different codecs for MOH, these might not be the base IP address and port, but an incremented IP address or port number. See the Configuring the MOH Audio Source to Enable Multicasting section. If you have multiple audio sources configured on Cisco Unified CM, ensure that you are using the audio sources' correct IP address and port number.
		• **route** *ip-address-list* —(Optional) Declares the IP address or addresses from which the flash MOH packets can be transmitted. A maximum of four IP address entries are allowed. If a **route** keyword is not configured, the Cisco Unified SRST system uses the **ip source-address** command value configured for Cisco Unified SRST.

Step 4	**moh-live dn-number** *calling-number***out-call***outcall-number*	Specifies that this telephone number is to be used for an outgoing call that is to be the source for an MOH stream.

	Command or Action	Purpose
	Example: `Router(config-cm-fallback)# moh-live dn-number 3333 out-call 7777`	- **dn-number** *calling-number* — Sets the MOH telephone number. The *calling-number* argument is a sequence of digits that represent a telephone number. - **out-call** *outcall-number* —Indicates that the router is calling out for a live feed that is to be used for MOH and specifies the number to be called. The *outcall-number* argument is a sequence of digits that represent a telephone number, typically of an E&M port. The **outcall** keyword makes a connection to the local router voice port that was specified in the the Setting Up the Voice Port on the Cisco Unified SRST Gateway section.
Step 5	**exit** **Example:** `Router(config-cm-fallback)# exit`	Exits call-manager-fallback configuration mode.

Verifying Cisco Unified SRST MOH Live Feed

To verify MOH live feed, use the **debug ephone moh** command and the other commands described in the Verifying Basic Cisco Unified SRST Multicast MOH Streaming section.

Configurations Examples for Cisco Unified SRST Gateways

This section provides the following configuration examples for Cisco Unified SRST gateways:

MOH Routed to Two IP Addresses: Example

The following example declares the Cisco Unified CM multicast MOH IP address 239.1.1.1 and port number 16384 and streams music-on-hold.au audio file packets out the interfaces that are configured with the IP addresses 10.1.1.1 and 172.21.51.143:

```
ccm-manager music-on-hold
 interface Loopback0
  ip address 10.1.1.1. 255.255.255.255

interface FastEthernet0/0
 ip address 172.21.51.143 255.255.255.192

call-manager-fallback
 ip source-address 172.21.51.143 port 2000
 max-ephones 1
 max-dn 1
 moh music-on-hold.au
 multicast moh 239.1.1.1 port 16384 route 172.21.51.143 10.1.1.1
```

Note The multicast IP address and port must match the IP address and the port number that Cisco Unified CM is configured to use for multicast MOH. If you are using different codecs for MOH, these might not be the base IP address and port, but an incremented IP address or port number. See the the Configuring the MOH Audio Source to Enable Multicasting section. If you have multiple audio sources configured on Cisco Unified CM, ensure that you are using the audio source's correct IP address and port number.

MOH Live Feed: Example

The following example configures MOH from a live feed. Note that the dial peer references the E&M port that was set with the **voice-port** command and that the dial peer number (7777) matches the outcall number configured with the **out-call** keyword of the **moh-live** command.

```
voice-port 1/0/0
 input gain 3
 auto-cut-through
 operation 4-wire
 signal immediate
!
dial-peer voice 7777 pots
 destination-pattern 7777
 port 1/0/0
!
!
call-manager-fallback
 max-conferences 8
 max-dn 1
 moh-live dn-number 3333 out-call 7777
!
.
.
.
```

Feature Information for Cisco Unified SRST as a Multicast MOH Resource

The Feature Information for Cisco Unified SRST as a Multicast MOH Resource table lists the enhancements to the Cisco Unified SRST as a Mulitcast MOH Resource feature by version.

To determine hardware and software compatibility, see the Cisco Unified CM Compatibility Information page at the following URL: http://www.cisco.com/en/US/products/sw/voicesw/ps556/products_device_support_tables_list.html

See also the Cisco Unified CM Documentation Roadmaps at the following URL: http://www.cisco.com/en/US/products/sw/voicesw/ps556/products_documentation_roadmaps_list.htm.

Use Cisco Feature Navigator to find information about platform support and software image support. Cisco Feature Navigator enables you to determine which Cisco IOS software images support a specific software release, feature set, or platform. To access Cisco Feature Navigator, go to http://www.cisco.com/go/cfn. An account on Cisco.com is not required.

Note The Feature Information for Cisco Unified SRST as a Multicast MOH Resource table lists the Cisco Unified SRST version that introduced support for a given feature. Unless noted otherwise, subsequent versions of Cisco Unified SRST software also support that feature.

Table 5: Feature Information for Cisco Unified SRST as a Multicast MOH Resource

Feature Name	Releases	Feature Information
Cisco Unified SRST as a Multicast MOH Resource	3.0	The MOH-live feature was added.

Where to Go Next

For additional information, see the Related Documents and References, on page 58 section in the Cisco Unified SRST Feature Overview, on page 41 chapter.