Cisco Unified SCCP and SIP 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.

For the most up-to-date information about Cisco Unified IP Phone support, the maximum number of Cisco Unified IP Phones, the maximum number of directory numbers (DNs) or virtual voice ports, and memory requirements for Cisco Unified SRST and Cisco Unified SIP SRST, see Cisco Unified SRST Supported Firmware, Platforms, Memory, and Voice Products.

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Cisco Unified SCCP SRST

Information About SCCP SRST

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. Prior to 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 back 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.

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.

Note The time it takes for a Cisco Unified IP Phone to fallback 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 fallback to SRST mode.

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. Assuming that 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 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.

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; see the "Device Pool Configuration Settings" chapter in the appropriate Cisco Unified CM Administration Guide. 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.

Cisco Unified SRST supports the following call combinations:

- SCCP phone to SCCP phone

- SCCP phone to PSTN/router voice-port

- SCCP phone to WAN VoIP using SIP or H.323

- 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

The figure shows a branch office with several Cisco Unified IP phones connected to a Cisco Unified SRST router. The router provides connections to both a WAN link and the PSTN. Typically, the Cisco Unified IP phones connect to their primary Cisco Unified Communications Manager at the central office via the WAN link. When the WAN connection is down, the Cisco Unified IP phones use the Cisco Unified SRST router as a fallback for their primary Cisco Unified Communications Manager. The branch office Cisco Unified IP phones are connected to the PSTN through the Cisco Unified SRST router and are able to make and receive off-net calls.

Figure 1: Branch Office Cisco Unifed IP Phones Connected to a Remote Central Cisco Unified Communications Manage Operating in SRST Mode
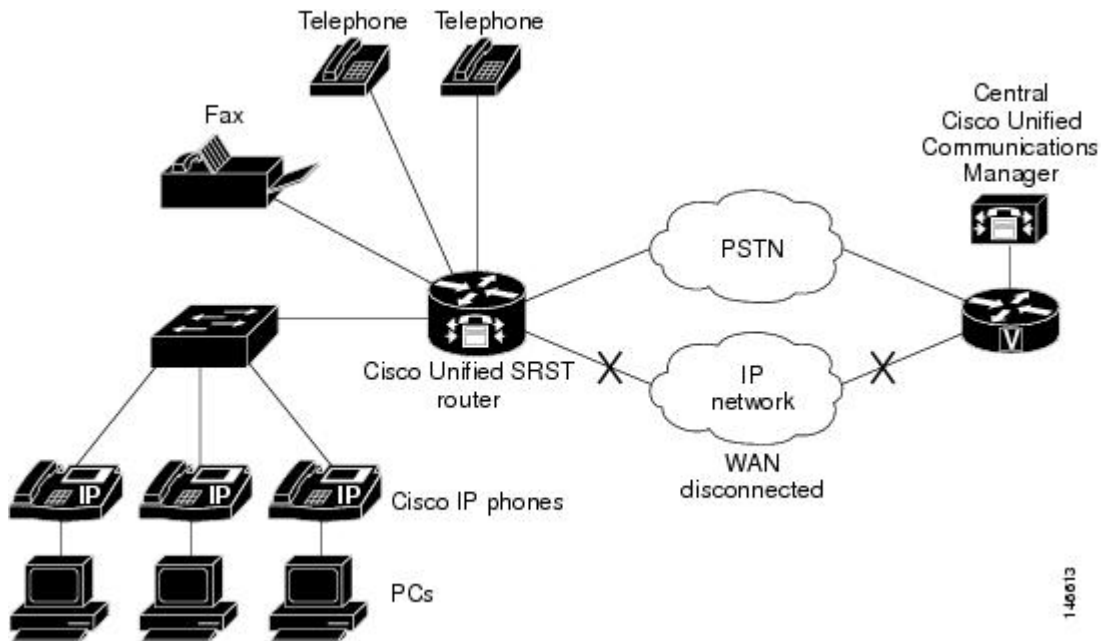
On H.323 gateways for SCCP SRST, when the WAN link fails, active calls from Cisco Unified IP phones to the PSTN are not maintained by default. Call preservation may work with the **no h225 timeout keepalive** command.

Under default configuration, the H.323 gateway maintains a keepalive signal with Cisco Unified Communications Manager and terminates H.323-to-PSTN calls if the keepalive signal fails, for example, if the WAN link fails. To disable this behavior and help preserve existing calls from local Cisco Unified IP phones, you can use the **no h225 timeout keepalive** command. Disabling the keepalive mechanism only affects calls that will be torn down as a result of the loss of the H.225 keepalive signal. For information regarding disconnecting a call when an inactive condition is detected, see the Media Inactive Call Detection document.

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.

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 .

Installing Cisco Unified SCCP SRST

Installing Cisco Unified SRST V3.0 and Later Versions

Install the Cisco IOS software release image containing the Cisco SRST or Cisco Unified SRST version that is compatible with your Cisco Unified Communications Manager version. See the . Cisco IOS software can be downloaded from the Cisco Software Center at http://www.cisco.com/public/sw-center/ http://www.cisco.com/public/sw-center/.

Cisco SRST and Cisco Unified SRST can be configured to support continuous multicast output of music-on-hold (MOH) from a flash MOH file in flash memory. For more information, see the If you plan to use MOH, go to the Technical Support Software Download site at http://www.cisco.com/cgi-bin/tablebuild.pl/ ip-iostsp and copy the music-on-hold.au file to the flash memory on your Cisco SRST or Cisco Unified SRST router.

Installing Cisco Unified SRST V2.0 and V2.1

Download and install Cisco SRST V2.0 or Cisco SRST V2.1 from the Cisco Software Center at http://www.cisco.com/public/sw-center/.

Installing Cisco Unified SRST V1.0

Cisco SRST V1.0 runs with Cisco Communications Manager V3.0.5 only. It is recommended that you upgrade to the latest Cisco Unified Communications Manager and Cisco Unified SRST versions.

Integrating Cisco Unified SCCP SRST with Cisco Unified Communications Manager

There are two procedures for integrating Cisco Unified SRST with Cisco Unified Communications Manager. Procedure selection depends on the Cisco Unified Communications Manager version that you have.

If You Have Cisco Communications Manager V3.3 or Later Versions

If you have Cisco Communications Manager V3.3 or later versions, 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.

1. Create an SRST Reference

    - From any page in Cisco Unified Communications Manager, click **System** and **SRST**.

    - On the Find and List SRST References page, click **Add a New SRST Reference**.

    - On the SRST Reference Configuration page, enter a name in the SRST Reference Name field and the IP address of the Cisco SRST router in the IP Address field.

    - Click **Insert**.

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

- From any page in Cisco Unified Communications Manager, click **System** and **Device Pool**.

- On the Device Pool Configuration page, click on the required device pool icon.

- On the Device Pool Configuration page, choose an SRST reference or Use Default Gateway from the SRST Reference field's menu.

If You Have Cisco Unified Communications Manager Version Prior to V3.3

If you have firmware versions that enable Cisco Unified SRST by default, no additional configuration is required on Cisco Unified Communications Manager to support Cisco Unified SRST. If your firmware versions disable Cisco Unified SRST by default, you must enable Cisco Unified SRST for each phone configuration.

1. Go to the Cisco Unified Communications Manager Phone Configuration page.

    - From any page in Cisco Unified Communications Manager, click **Device** and **Phone**.

    - In the Find and List Phones page, click **Find**.

    - After a list of phones appears, click on the required device name.

    - The Phone Configuration appears.

2. In the Phone Configuration page, go to the Product Specific Configuration section at the end of the page, choose **Enabled** from the Cisco Unified SRST field's menu, and click **Update**.

3. Go to the Phone Configuration page for the next phone and choose Enabled from the Cisco Unified SRST field's menu by repeating Step 1 and Step 2.

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).

Interface Support for Cisco Unified Communications Manager Express and Cisco Unified 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)

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 on page 126.

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.

IPv6 Support for Unified SRST SIP IP Phones

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 .

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 .

For an example of configuring IPv6 Support on Unified SRST, see .

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.

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, page 36.

- 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 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 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.

	Command or Action	Purpose
Step 6	**no anat** **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-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.

	Command or Action	Purpose
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 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
```

Support for Cisco Unified IP Phones and Platforms

Support for Cisco Unified IP Phones and Platforms

The following sections provide information about Cisco Feature Navigator and the histories of Cisco Unified IP Phone, platform, and Cisco Unified CM support from Cisco SRST Version 1.0 to the present version of Cisco Unified SRST.

-
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-
-
-
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Finding Cisco IOS Software Releases That Support Cisco Unified SRST

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

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

See Cisco Unified CME and Cisco IOS Software Version Compatibility Matrix for related compatibility information.

Cisco Unified IP Phone Support

For the most up-to-date information about Cisco Unified IP Phone support, see Compatibility Information.

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 passthrough 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.

During Cisco Unified CM 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 available as of Cisco 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.

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.

Platform and Memory Support

Platform and Memory Support

For the most up-to-date information about Platform and Memory Support, see Compatibility Information.

Determining Platform Support Through 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.

Availability of Cisco IOS Software Images

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.

For the most up-to-date information about Cisco IOS software images, see Compatibility Information.

Cisco Unified Communications Manager Compatibility

See Cisco Unified Communications Manager Compatibility Matrix.

Signal Support

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

Language Support

See Cisco Unified Communications Manager Express Cisco Unified CME Localization Matrix.

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)

Multicast Music On Hold

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, page 13.

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*
	
	 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`
		Enables music on hold using the specified file.
		
		 • 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.

	Command or Action	Purpose
		• 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 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

Additional 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.