Cisco Unified SIP SRST 4.0 System Administrator Guide

Cisco IOS Release
12.4(4)XC
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CONTENTS

Cisco Unified SIP SRST Feature Roadmap

Note Prior to version 4.0, the name of this product was Cisco SIP SRST.

This chapter contains a summary of Cisco Unified Session Initiation Protocol (SIP) Survivable Remote Site Telephony (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.

Note The Cisco IOS Voice Configuration Library includes a standard library preface, a glossary, and feature and troubleshooting documents and is located at http://www.cisco.com/univercd/cc/td/doc/product/software/ios123/123cgcr/vcl.htm.

Contents

Documentation Organization

This book consists of the following chapters as shown in Table 1.

Table 1 Cisco Unified SIP SRST Configuration Sequence

Chapter or Appendix	Description
Cisco Unified SIP SRST Feature Overview	Gives a brief description of Cisco Unified SIP 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.
Getting Started	Describes the two versions of Cisco Unified SIP SRST. This chapter gives a brief overview of each version. In addition, Version 3.4 requires a few changes and new configurations as compared to the setup that was required for Version 3.0. This chapter includes the following tasks: • Disabling Call Redirection • Enabling SIP-to-SIP Connection Capabilities
Configuring the SIP Registrar	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 proxy server 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. This chapter includes the following tasks: • Configuring the SIP Registrar • Configuring Backup Registrar Service to SIP Phones • Configuring Backup Registrar Service to SIP Phones (Using Optional Commands) • Verifying SIP Registrar Configuration • Verifying Proxy Dial-Peer Configuration
Configuring Cisco Unified SIP SRST Features Using Redirect Mode (for Version 3.0 Only)	Describes features using redirect mode. This chapter includes the following tasks: • Configuring Call Redirect Enhancements to Support Calls Between SIP IP Phones for Cisco Unified SIP SRST • Configuring Sending 300 Multiple Choice Support
Configuring Cisco Unified SIP SRST Features Using Back-to-Back User Agent Mode (for Version 3.4 and Version 4.0 Only)	Describes features using back-to-back user agent mode. Features include Cisco Unified SIP SRST support for standardized RFC 3261 SIP phones. This chapter includes the following tasks: • Configuring SIP Phone Features • Configuring SIP-to-SIP Call Forwarding • Configuring Call Blocking Based on Time of Day, Day of Week, or Date

Feature Roadmap

Table 2 provides a summary of Cisco Unified SIP SRST features by release.

Table 2 Cisco Unified SIP SRST Features by Cisco IOS Release

Cisco SIP SRST Version	Cisco IOS Release	Modifications
Version 4.0	12.4(4)XC	—
Version 3.4	12.4(4)T	Cisco SIP SRST 3.4 includes the following features: • Getting Started • Configuring Cisco Unified SIP SRST Features Using Redirect Mode (for Version 3.0 Only) (formerly *SIP SRST*) • Configuring Cisco Unified SIP SRST Features Using Back-to-Back User Agent Mode (for Version 3.4 and Version 4.0 Only)
Version 3.2	12.3(11)T	The SIP SRST feature was updated to include additional prerequisite information, including phone and memory requirements.
Version 3.1	12.3(7)T	The SIP SRST feature was integrated into Cisco IOS Release 12.3(7)T.
Version 3.0	12.2(15)ZJ 12.3(4)T	The SIP SRST feature was introduced.

Cisco Unified SIP SRST Feature Overview

Note Prior to version 4.0, the name of this product was Cisco SIP SRST.

This chapter includes information about supported Cisco IP phones and platforms. It also includes information on Cisco Unified Session Initiation Protocol (SIP) Survivable Remote Site Telephony (SRST) 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 DNs or virtual voice ports, and memory requirements for Cisco Unified SRST and Cisco Unified SIP SRST, see the *Cisco Unified SRST 4.0 Supported Firmware, Platforms, Memory, and Voice Products* at http://www.cisco.com/en/US/customer/products/sw/voicesw/ps2169/prod_installation_guide09186a00805f6f1b.html.

Contents

Cisco Unified SIP SRST Description

This book describes Survivable Remote Site Telephony (SRST) functionality for Session Initiation Protocol (SIP) networks. Cisco Unified SIP SRST provides backup to an external SIP proxy server 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.

Cisco Unified SIP SRST supports the following call combinations:

- 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

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.

Figure 1 shows that when the WAN is up, dual registration occurs. The phone registers with the SIP proxy server and the SIP registrar (B2BUA router). But any calls from the SIP phone go to the SIP proxy server through the WAN and out to the PSTN.

Figure 1 Dual Registration When WAN Is Up

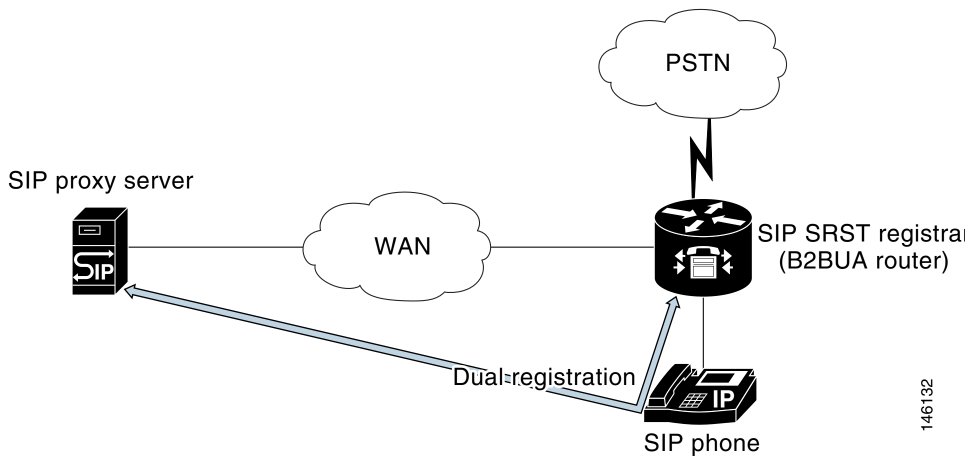
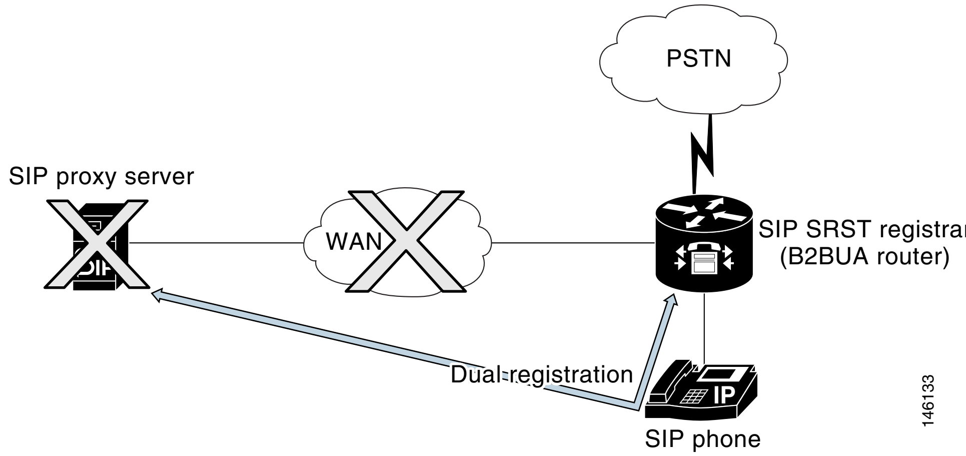


Figure 2 shows that when the WAN or SIP proxy server goes down, the call from the SIP phone cannot get to the SIP proxy server and instead goes through the B2BUA router out to the PSTN.

Figure 2 Call Proceeds with Cisco Unified SIP SRST, When WAN Is Down



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 and platform support from Cisco SRST 3.0 to the present version.

Finding Cisco IOS Software Releases That Support Cisco Unified SRST

The tables in this chapter list only the Cisco IOS software releases that first introduce new features to Cisco Unified SRST. Other Cisco IOS software releases may subsequently inherit versions of Cisco Unified SRST. To get a list of Cisco IOS software releases that support a particular version of Cisco Unified SRST, use Cisco Feature Navigator.

Cisco Feature Navigator is a web-based tool that enables you to determine which Cisco IOS software images support a specific set of features and which features are supported in a specific Cisco IOS image. You can search by feature or release. Under the release section, you can compare releases side by side to display both the features unique to each software release and the features in common.

To access Cisco Feature Navigator, you must have an account on Cisco.com. If you have forgotten or lost your account information, send a blank e-mail to cco-locksmith@cisco.com. An automatic check will verify that your e-mail address is registered with Cisco.com. If the check is successful, account details with a new random password will be e-mailed to you. Qualified users can establish an account on Cisco.com by following the directions found at this URL:

http://tools.cisco.com/RPF/register/register.do

Cisco Feature Navigator is updated regularly when major Cisco IOS software releases and technology releases occur. For the most current information, go to the Cisco Feature Navigator home page at the following URL:

http://www.cisco.com/go/fn

Cisco Unified IP Phone Support

For the most up-to-date information about Cisco Unified IP Phone support, see *Cisco Unified SRST 4.0 Supported Firmware, Platforms, Memory, and Voice Products* at http://www.cisco.com/en/US/customer/products/sw/voicesw/ps2169/prod_installation_guide09186a00805f6f1b.html

Cisco UnifiedIP Phone 7940G and Cisco Unified IP Phone 7960G are fully supported if dual registration is enabled. Dual registration means that the SIP phone is capable of registering with the main SIP proxy and the Cisco Unified SIP SRST device (redirect server or back-to-back user agent) at the same time. If this requirement is not met, the Cisco Unified SIP SRST device may not be capable of routing incoming calls to the SIP phone until the SIP phone registers with the Cisco Unified SIP SRST device. Cisco Unified IP Phone 7940G and Cisco Unified IP Phone 7960G,l beginning with phone load POS3-04-2-00.bin, are capable of dual registration of the phone's primary phone line. Additional lines are not registered by the phone for Cisco Unified SIP SRST. To enable dual registration for the primary line, you must set backup proxy information such as proxy_backup and proxy_backup_port in the SIP phone's configuration file. For configuration instructions, see the *Cisco SIP IP Phone 7960 Administrator Guide*, Version 5.1.

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.

Platform and Memory Support

For the most up-to-date information about platform and memory support, see the *Cisco Unified SRST 4.0 Supported Firmware, Platforms, Memory, and Voice Products* at http://www.cisco.com/en/US/customer/products/sw/voicesw/ps2169/prod_installation_guide09186a00805f6f1b.html.

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. Please contact your account representative if you have further questions.
- Cisco Unified IP Phone 7940G and Cisco IP Phone 7960G are fully supported if dual registration is enabled. Dual registration means that the SIP phone is capable of registering with the main SIP proxy and the Cisco Unified SIP SRST device (redirect server or back-to-back user agent) at the same time. If this requirement is not met, the Cisco Unified SIP SRST device may not be capable of routing incoming calls to the SIP phone until the SIP phone registers with the Cisco Unified SIP SRST device. Cisco Unified IP Phone 7940G and Cisco Unified IP Phone 7960G, beginning with phone load POS3-04-2-00.bin, are capable of dual registration of the phone's primary phone line. Additional lines are not registered by the phone for Cisco Unified SIP SRST. To enable dual registration for the primary line, you must set backup proxy information such as proxy_backup and proxy_backup_port in the SIP phone's configuration file. For configuration instructions, see the *Cisco SIP IP Phone 7960 Administrator Guide*, Version 5.1.

Note When the WAN goes down, for each outgoing call the SIP phone continues to send the SIP proxy server up to seven Invite messages. If the Invite messages are not acknowledged, the SIP phone switches to Cisco Unified SIP SRST to route the call. Thus, there may be a few seconds delay before SIP SRST takes over call processing from the SIP proxy server. If your network is designed to return an ICMP host unreachable indication to the phone in response to an outgoing SIP Invite message when the WAN is down, the phone responds by switching to the Cisco Unified SIP SRST router more rapidly.

Dual registration is not supported on the Cisco Unified IP Phone 7905G, Cisco Unified IP Phone 7912G, or Cisco Analog Telephone Adaptor (ATA) series with a SIP image. Therefore auto registration to the SIP SRST router is not available.

- If the WAN is down, and you reboot your Cisco Unified SIP SRST router, when the router reloads it will have no database of SIP phone registrations. The SIP phones will have to register again, which could take several minutes, because SIP phones do not use a keepalive functionality. To shorten the time before the phones re-register, the registration expiry can be adjusted with the **registrar server** command. The default expiry is 3600 seconds; an expiry of 600 seconds is recommended.

Restrictions for Configuring Cisco Unified SIP SRST

Table 3 provides a history of restrictions from Cisco SIP SRST 3.0 to the present version.

Table 3 *History of Restrictions from Cisco SIP SRST Version 3.0 to the Present Version*

Cisco 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** • Music on hold (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).

Where to Go Next

The next chapters of this book describe how to configure Cisco Unified SIP SRST. As shown in Table 4, 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. For instructions, see the "Prerequisites for Configuring Cisco Unified SIP SRST" section on page 8.

Table 4 Cisco Unified SRST Configuration Sequence

Task	Where Task Is Described
1. If you are upgrading to Version 3.4 or using Cisco Unified SIP SRST for the first time, this chapter describes procedures to get you started.	"Getting Started" chapter
2. This chapter describes how to provide a backup to an external SIP proxy server by providing basic registrar services.	"Configuring the SIP Registrar" chapter
3. This chapter describes basic Cisco Unified SIP SRST and local SIP phone configurations that were introduced in Version 3.0.	"Configuring Cisco Unified SIP SRST Features Using Redirect Mode (for Version 3.0 Only)" chapter
4. This chapter describes global phone configurations and additional features, such as call forwarding, that were introduced in Version 3.0.	"Configuring Cisco Unified SIP SRST Features Using Back-to-Back User Agent Mode (for Version 3.4 and Version 4.0 Only)" chapter

Additional References

The following sections provide additional references related to Cisco Unified SIP SRST:

- Related Documents, page 11
- Standards, page 12
- MIBs, page 12
- RFCs, page 12
- Technical Assistance, page 13

Related Documents

Related Topic	Documents
Cisco Unified SRST commands and specifications	• *Cisco Unified SRST and Cisco Unified SIP SRST Command Reference (All Versions)* • *Cisco Unified SRST 4.0 Supported Firmware, Platforms, Memory, and Voice Products* a
Cisco Unified SRST administration	• *Cisco Unified SRST 4.0 System Administrator Guide*

Related Topic	Documents
Cisco Unified IP Phones	• *Cisco IP Phone 7902 Quick Start Guide* • *Cisco IP Phone 7902G Quick Start Guide* • *At a Glance Cisco IP Phone 7912G* • *Cisco IP Phone 7960 and 7940 Series User Guide* • *Cisco IP Phone 7970 Guide* • *Cisco SIP IP Phone 7960 Administrator Guide*, Version 5.1
Cisco SIP functionality	• *Cisco IOS SIP Configuration Guide*
Command reference information for voice and telephony commands	• *Cisco IOS Voice Command Reference* • *Cisco IOS Debug Command Reference*
Standard preface	• *Cisco IOS Voice Configuration Library Preface*
Standard glossary	• *Cisco IOS Voice Configuration Library Glossary*

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
RFC 2543	*SIP: Session Initiation Protocol*
RFC 3261	*SIP: Session Initiation Protocol*

Technical Assistance

Description	Link
The Cisco Technical Support 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

Getting Started

Note Prior to version 4.0, the name of this product was Cisco SIP SRST.

This chapter describes the main tasks necessary for the following:

- Running Cisco Session Initiation Protocol (SIP) Survivable Remote Site Telephony (SRST) 3.0 for the first time
- Running Cisco Unified SIP SRST 4.0 for the first time
- Upgrading from Cisco SIP SRST 3.0 to Cisco Unified SIP SRST 4.0

Note that upgrades from Cisco SIP SRST 3.4 to Cisco Unified SIP SRST 4.0 are not impacted by the issues discussed in this chapter.

Note The Cisco IOS Voice Configuration Library includes a standard library preface, glossary, and feature and troubleshooting documents and is located at http://www.cisco.com/univercd/cc/td/doc/product/software/ios123/123cgcr/vcl.htm.

Contents

Comparison of Cisco SIP SRST 3.0 and Cisco Unified SIP SRST 4.0

Cisco SIP SRST 3.0, Cisco IOS Release 12.2(15)ZJ Through Cisco IOS Release 12.4

Cisco SIP SRST 3.0 was a predecessor to Cisco Unified SIP SRST 4.0. In Cisco SIP SRST 3.0, you could configure a Cisco IOS voice gateway to act as a SIP redirect server. The voice gateway would respond to the originator of a call with a SIP Redirect message, and the Redirect message allowed the SIP phone that originated the call to establish a call to its destination. In addition, several commands in voice register pool configuration mode were introduced that allowed registration permission control.

Cisco Unified SIP SRST V4.0, Cisco IOS Release 12.4(4)XC

With Cisco Unified SIP SRST 4.0, a SIP redirect server is not necessary. Instead, a back-to-back user agent (B2BUA) server routes the call as desired. A B2BUA is a separate call agent that has more features than a redirect server, which can accept and forward calls only. With a B2BUA you can also configure call blocking and call forwarding. In call forwarding, the B2BUA forwards calls on behalf of the phone, while maintaining a presence as call middleman in the call path.

Configuration and Upgrade Tasks

The table below lists the high-level steps you need to take to upgrade to Cisco Unified SIP SRST 4.0. It also lists the high-level steps you need to take in order to run Version 3.0 or Version 4.0.

Cisco Unified SIP SRST Version	Instructions and Procedures
If you are interested in Cisco SIP SRST 3.0 (using a redirect server), complete these procedures.	Cisco SIP SRST Version 3.0 provides a backup to an external SIP proxy server by providing basic registrar and redirect services. The following chapters provide full Version 3.0 information, including basic voice register pool configurations. • Configuring the SIP Registrar • Configuring Backup Registrar Service to SIP Phones • Configuring Cisco Unified SIP SRST Features Using Redirect Mode (for Version 3.0 Only)

<table>
<tr><th>Cisco Unified SIP SRST Version</th><th>Instructions and Procedures</th></tr>
<tr><td>If you are interested in Cisco Unified SIP SRST 4.0 (using a B2BUA) and have never used Cisco Unified SIP SRST in the past, complete these procedures.</td><td>VoIP-to-VoIP connections permit the termination and reorigination of transferred and forwarded calls over the VoIP network. The following task describes how to allow SIP connections:
• Enabling SIP-to-SIP Connection Capabilities
SIP registrar functionality in Cisco IOS software is a required part of Cisco Unified SIP SRST. 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. The following task describes how to configure the SIP registrar:
• Configuring the SIP Registrar
Configure a basic voice register pool:
• Configuring Backup Registrar Service to SIP Phones
You are now ready to configure Version 4.0 features such as call blocking and call forwarding. The following chapter describes the call blocking and call forwarding configurations:
• Configuring Cisco Unified SIP SRST Features Using Back-to-Back User Agent Mode (for Version 3.4 and Version 4.0 Only)</td></tr>
<tr><td>If you are currently running Version 3.0 and want to upgrade to Version 4.0, complete these procedures.</td><td>Since Version 4.0 uses a B2BUA and not a redirect server, call redirection must be disabled as described in the following task:
• Disabling Call Redirection
VoIP-to-VoIP connections permit the termination and reorigination of transferred and forwarded calls over the VoIP network. The following task describes how to allow SIP connections:
• Enabling SIP-to-SIP Connection Capabilities
You are now ready to configure Version 4.0 features such as call blocking and call forwarding. The following chapter describes the call blocking and call forwarding configurations:
• Configuring Cisco Unified SIP SRST Features Using Back-to-Back User Agent Mode (for Version 3.4 and Version 4.0 Only)</td></tr>
</table>

How to Upgrade from Cisco SIP SRST 3.0 to Cisco Unified SIP SRST 4.0

This section contains the following procedures:

- Disabling Call Redirection, page 18 (required)
- Enabling SIP-to-SIP Connection Capabilities, page 21 (required)

Disabling Call Redirection

Because Version 4.0 uses a B2BUA and not a redirect server, call redirection must be disabled if it was previously enabled. Complete the following tasks as required, depending on whether call redirection was enabled globally or on a dial-peer basis.

- Disabling Call Redirection Globally, page 18
- Disabling Call Redirection on a Specific VoIP Dial Peer, page 19

Disabling Call Redirection Globally

To disable 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. **no 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.

	Command or Action	Purpose

Step 3	`voice service voip`
	
	Example:
	`Router(config)# voice service voip`
		Enters voice service configuration mode.

Step 4	`no redirect ip2ip`
	
	Example:
	`Router(config-voi-srv)# no redirect ip2ip`
		Disables redirection of SIP phone calls to SIP phone calls globally using the Cisco IOS voice gateway.

Step 5	`end`
	
	Example:
	`Router(config-voi-srv)# end`
		Returns to privileged EXEC mode.

Disabling Call Redirection on a Specific VoIP Dial Peer

To disable IP-to-IP call redirection for a specific VoIP dial peer, disable it on the inbound dial peer where it was originally enabled.

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. **dial-peer voice** *tag* **voip**
4. **no 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.

	Command or Action	Purpose
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	**no redirect ip2ip** **Example:** `Router(config-dial-peer)# no redirect ip2ip`	Disables redirection of SIP phone calls to SIP phone calls on a specific VoIP dial peer using the Cisco IOS voice gateway.
Step 5	**end** **Example:** `Router(config-dial-peer)# end`	Returns to privileged EXEC mode.

Enabling SIP-to-SIP Connection Capabilities

VoIP-to-VoIP connections permit the termination and reorigination of transferred and forwarded calls over the VoIP network. For Cisco Unified SIP SRST 4.0 we enable SIP-to-SIP connections for hairpin call routing. The B2BUA that routes the call uses the SIP-to-SIP connection. Because VoIP-to-VoIP connections are disabled on the router by default, they must be explicitly enabled to use call routing.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **allow-connections sip to sip**
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 to establish global call transfer and forwarding parameters.
Step 4	**allow-connections sip to sip** **Example:** `Router(config-voi-srv)# allow-connections sip to sip`	Enables VoIP-to-VoIP call connections. Use the **no** form of the command to disable VoIP-to-VoIP connections, which is the default.
Step 5	**end** **Example:** `Router(conf-voi-serv)# end`	Returns to privileged EXEC mode.

What to Do Next

SIP registrar functionality in Cisco IOS software is a required part of Cisco Unified SIP SRST. By default, Cisco Unified SIP SRST is not enabled and cannot accept SIP register messages. To configure the SIP registrar to accept incoming SIP Register messages, see the "Configuring the SIP Registrar" chapter.

To configure a basic voice register pool, see "Configuring Backup Registrar Service to SIP Phones" section on page 26.

To configure call forwarding or call blocking, see the "Configuring Cisco Unified SIP SRST Features Using Back-to-Back User Agent Mode (for Version 3.4 and Version 4.0 Only)" chapter.

Configuring the SIP Registrar

Note Prior to version 4.0, the name of this product was Cisco SIP 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 2543, 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.

Note The Cisco IOS Voice Configuration Library includes a standard library preface, glossary, and feature and troubleshooting documents and is located at http://www.cisco.com/univercd/cc/td/doc/product/software/ios123/123cgcr/vcl.htm.

Contents

This section contains the following procedures:

Prerequisites for Configuring the SIP Registrar

Complete the prerequisites documented in the "Prerequisites for Configuring Cisco Unified SIP SRST" section in the "Cisco Unified SIP 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 SIP SRST Feature Overview" chapter.

Information About Configuring the SIP Registrar

Cisco Unified SIP SRST provides backup to an external SIP proxy server by providing basic registrar and redirect 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.

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.
- 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

This section contains the following procedures:

- Configuring the SIP Registrar, page 24 (required)
- Configuring Backup Registrar Service to SIP Phones, page 26 (required)
- Configuring Backup Registrar Service to SIP Phones (Using Optional Commands), page 30 (optional)
- Verifying SIP Registrar Configuration, page 33 (optional)
- Verifying Proxy Dial-Peer Configuration, page 34 (optional)

Configuring the SIP Registrar

The local SIP gateway that becomes the SIP registrar acts as a backup SIP proxy or redirector and accepts SIP Register messages from SIP phones. It becomes a location database of local SIP IP phones that are set up for dual registration. Dual registration allows SIP IP phones to simultaneously register with both their primary and their fallback registrar devices. That is, when a SIP IP phone registers with a Cisco Unified SIP SRST gateway, it simultaneously registers with the main proxy and SIP redirect server for coverage in case of a WAN failure.

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.

Prerequisites

The SIP endpoints (IP phones) must support dual concurrent registration, which is registering with the main SIP proxy and the Cisco Unified SIP SRST device (redirect server) at the same time. If this requirement is not met, the Cisco Unified SIP SRST device cannot route incoming calls to the SIP phone. For configuration instructions, see the Cisco IP Phone Documentation for Session Initiation Protocol (SIP).

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **sip**
5. **registrar server** [**expires** [**max** *sec*] [**min** *sec*]]
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.

	Command or Action	Purpose
Step 5	**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. • **min** *sec*: (Optional) Minimum expiration time for a registration, in seconds. The range is from 60 to 3600. The default is 60.
Step 6	**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 “Configuring Backup Registrar Service to SIP Phones” section on page 26.

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 (using VoIP-to-VoIP dial-peer redirect)

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.

Note For command-level information, see the appropriate command page in the *Cisco Unified SRST and Cisco Unified SIP SRST Command Reference (All Versions)*.

Prerequisites

- The SIP registrar must be configured before a voice register pool is set up. See the “Configuring the SIP Registrar” section on page 24 for complete instructions.

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.

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. **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.

	Command or Action	Purpose
Step 3	**call fallback active** **Example:** `Router(config)# call fallback active`	(Optional) 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 the 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.

	Command or Action	Purpose
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`	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** 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.
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	**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 "Configuring Backup Registrar Service to SIP Phones (Using Optional Commands)" section on page 30 for these commands.

Configuring 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.

Prerequisites

- Prerequisites as described in the "Configuring Backup Registrar Service to SIP Phones" section on page 26.
- Configuration of the required commands as described in the "Configuring Backup Registrar Service to SIP Phones" section on page 26.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register pool** *tag*
4. **translate-outgoing** {**called** | **calling**} *rule-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. **max registrations** *value*
9. **number** *tag number-pattern* {**preference** *value*} [**huntstop**]
10. **dtmf-relay** [**cisco-rtp**] [**rtp-nte**] [**sip-notify**]
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 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	**`translate-outgoing`** {**`called`** | **`calling`**} *`rule-tag`* **Example:** `Router(config-register-pool)# translate-outgoing called 1`	Allows explicit setting of translation rules on the VoIP dial peer to modify a phone number dialed by a Cisco Unified IP Phone user. • The *rule-tag* argument is the reference number of the translation rule. Valid entries are 1 to 2147483647.
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*: (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.

	Command or Action	Purpose
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. • **-** : (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.

	Command or Action	Purpose
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. • **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.

Examples

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

Step 1 **debug voice register errors**

Use this command to debug errors that happen during registration, for 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
```

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**

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.

```
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 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 in the debug, which means there is a preexisting VoIP dial peer.

Step 3 **show sip-ua status registrar**

Use this command to display all the SIP endpoints currently registered with the contact address.

```
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
```

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** *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

Step 1 **configure terminal**

Use this command to enter global configuration mode.

```
Router# configure terminal
```

Step 2 **voice register pool** *tag*

Use this command to enter voice register pool configuration mode.

```
Router(config)# voice register pool 1
```

Step 3 **proxy** *ip-address* [**preference** *value*] [**monitor probe** {**icmp-ping** | **rtr**} [*alternate-ip-address*]]

Set the **proxy** command to monitor with **icmp-ping**:

```
Router(config-register-pool)# proxy 10.2.161.187 preference 1 monitor probe icmp-ping
```

Step 4 **end**

Returns to privileged EXEC mode.

```
Router(config-register-pool)# end
```

Step 5 **show voice register dial-peers**

Use this command to verify dial-peer configurations, and notice that **icmp-ping** monitoring is set.

```
Router# show voice register dial-peers

dial-peer voice 40035 voip
preference 5
destination-pattern 91011
redirect ip2ip
session target ipv4:192.168.0.2
session protocol sipv2
voice-class codec 1

dial-peer voice 40036 voip
preference 1
destination-pattern 91011
redirect ip2ip
session target ipv4:10.2.161.187
session protocol sipv2
voice-class codec 1
monitor probe icmp-ping 10.2.161.187
```

Step 6 **show dial-peer voice**

Finally, 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.

```
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:
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
CAS=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,
redirect ip2ip = enabled
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.
```

What to Do Next

To configure Cisco Unified SIP SRST redirect mode, features see the "Configuring Cisco Unified SIP SRST Features Using Redirect Mode (for Version 3.0 Only)" chapter.

To configure Cisco Unified SIP SRST call forwarding and call blocking features, see the "Configuring Cisco Unified SIP SRST Features Using Back-to-Back User Agent Mode (for Version 3.4 and Version 4.0 Only)" chapter.

Configuring Cisco Unified SIP SRST Features Using Redirect Mode (for Version 3.0 Only)

Note Prior to version 4.0, the name of this product was Cisco SIP SRST.

This chapter describes Cisco Unified Session Initiation Protocol (SIP) Survivable Remote Site Telephony (SRST) features using redirect mode.

Note The Cisco IOS Voice Configuration Library includes a standard library preface, glossary, and feature and troubleshooting documents and is located at
http://www.cisco.com/univercd/cc/td/doc/product/software/ios123/123cgcr/vcl.htm.

Contents

Prerequisites for Cisco Unified SIP SRST Features Using Redirect Mode

Complete the prerequisites documented in the “Prerequisites for Configuring Cisco Unified SIP SRST” section in the “Cisco Unified SIP SRST Feature Overview” chapter.

Restrictions for Cisco Unified SIP SRST Features Using Redirect Mode

See the restrictions documented in the "Restrictions for Configuring Cisco Unified SIP SRST" section in the "Cisco Unified SIP SRST Feature Overview" chapter.

Information About Cisco UnifiedSIP SRST Features Using Redirect Mode

Cisco Unified SIP SRST provides backup to an external SIP proxy server by providing basic registrar and redirect 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.

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.
- 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 Cisco Unified SIP SRST Features Using Redirect Mode

This section contains the following procedures:

- Configuring Call Redirect Enhancements to Support Calls Between SIP IP Phones for Cisco Unified SIP SRST, page 41 (required)
- Configuring Sending 300 Multiple Choice Support, page 43 (required)

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. Prior to 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, now the Cisco IOS voice gateway can 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 Call Redirect Enhancements to Support Calls Globally, page 41
- Configuring Call Redirect Enhancements to Support Calls on a Specific VoIP Dial Peer, page 42

Configuring Call Redirect Enhancements to Support Calls Globally

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.

	Command or Action	Purpose
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`	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.

Restrictions

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

Prior to 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** | **longest-match**]
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`** `|` **`longest-match`**`]` **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**—(Optional) Uses the current system configuration to set the order of contacts. • **longest-match**—(Optional) 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

Note IP addresses and hostnames in examples are fictitious.

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
!
```

Where to Go Next

After configuring basic Cisco Unified SIP SRST, the “Configuring Cisco Unified SIP SRST Features Using Back-to-Back User Agent Mode (for Version 3.4 and Version 4.0 Only)” chapter describes additional configurations to increase SIP phone functionality.

Configuring Cisco Unified SIP SRST Features Using Back-to-Back User Agent Mode (for Version 3.4 and Version 4.0 Only)

Note Prior to version 4.0, the name of this product was Cisco SIP SRST.

This chapter describes Cisco Unified Cisco Unified Survivable Remote Site Telephony (SRST) support for standardized RFC 3261 features for SIP phones. Features include call blocking and call forwarding.

Note The Cisco IOS Voice Configuration Library includes a standard library preface, glossary, and feature and troubleshooting documents and is located at http://www.cisco.com/univercd/cc/td/doc/product/software/ios123/123cgcr/vcl.htm.

Contents

Prerequisites for Cisco Unified 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 SIP SRST Feature Overview" chapter.

- Complete the necessary tasks found in the "Getting Started" chapter. Specific tasks include the required task that is documented in the "Enabling SIP-to-SIP Connection Capabilities" section on page 21.
- 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 "Configuring the SIP Registrar" section on page 24.

Restrictions for Cisco Unified 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 SIP SRST Feature Overview" chapter.

Information About Cisco Unified 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 gives 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

See Figure 1 on page 6 and Figure 2 on page 7 for an illustration of Cisco Unified SIP SRST using a B2BUA.

Cisco Unified SIP SRST and Cisco SIP CallManager Express Feature Crossover

Cisco Unified SIP SRST uses is a voice register dn configuration mode. However, in a typical Cisco Unified SIP SRST setup, **voice register dn** commands are not used, so they are not discussed in this book. Although you are not restricted from using **voice register dn** commands, they are not likely to be needed in a Cisco Unified SIP SRST environment. The **voice register dn** commands are most likely to be used in a Cisco Unified SIP CallManager Express (CME) environment. If you work in a Cisco Unified SIP CME environment and would like to know which commands are also applicable to Cisco Unified SIP SRST, Table 5 lists Version 3.4 commands for CME and SRST. Commands marked under the column "Cisco (SIP) CME Mode Only" show up if **mode** *cme* is configured in voice register global configuration mode; these commands apply to Cisco CME only.

Procedures for configuring Cisco Unified SIP CME and complete descriptions of all CME and voice register dn commands are found in the Cisco CallManager Express Version 3.4 documentation.

Note Table 5 is not all-inclusive; additional commands may exist.

Table 5 Version 3.4 New or Enhanced Commands for Cisco Unified SRST and Cisco Unified CME (Sorted by Configuration Mode)

Function—>Command	Dial Peer	Voice Register Mode	Configurable for Cisco Unified (SIP) CME and Cisco Unified SIP SRST	Applicable to Cisco Unified (SIP) CME Only
after-hour exempt	X	dn	X	—
auto-answer	—	dn	—	X
call forward	X	dn	X	—
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
timezone	—	global	—	X
upgrade	—	global	—	X
url	—	global	—	X

Table 5 Version 3.4 New or Enhanced Commands for Cisco Unified SRST and Cisco Unified CME (Sorted by Configuration Mode) (continued)

Function—>Command	Dial Peer	Voice Register Mode	Configurable for Cisco Unified (SIP) CME and Cisco Unified SIP SRST	Applicable to Cisco Unified (SIP) CME 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
translate-outgoing	X	pool	X	—
type	—	pool	—	X
username	—	pool	—	X
vad	X	pool	X	—
anonymous	—	template	—	X
caller-id	—	template	—	X
conference	—	template	—	X
dnd-control	—	template	—	X
forward	—	template	—	X
transfer	—	template	—	X

How to Configure Cisco Unified SIP SRST

This section contains the following procedures:

- Configuring SIP Phone Features, page 51 (optional)
- Configuring SIP-to-SIP Call Forwarding, page 53 (required)
- Configuring Call Blocking Based on Time of Day, Day of Week, or Date, page 55 (required)
- SIP Call Hold and Resume, page 58 (no confguration necessary)

Configuring SIP Phone Features

Once a voice register pool has been set, this procedure adds optional features to increase functionality. Some features can be made 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 **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.

	Command or Action	Purpose
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`	Sets the maximum number of SIP voice register pools that are supported 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 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 a 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 used 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 to be 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 seem to be disconnected. You may prefer to set **no vad** on the SIP phone pool.

	Command or Action	Purpose
Step 10	**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 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 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. For more information on setting the **allow-connections** command, see the "Enabling SIP-to-SIP Connection Capabilities" section on page 21. 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 **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 CallManager Express (CME) environment. Detailed procedures for configuring the **call-forward b2bua mailbox** command are found in Cisco CallManager Express Version 3.4 documentation.

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 extension: • *directory-number*—Telephone number to which calls are forwarded. Represents a fully qualified E.164 number. Maximum length of the telephone 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*—Telephone number to which calls are forwarded. Represents a fully qualified E.164 number. Maximum length of the telephone number is 32.

	Command or Action	Purpose
Step 6	**call-forward b2bua mailbox** *directory-number* **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*—Telephone number to which calls are forwarded when the forwarded destination is busy or does not answer. Represents a fully qualified E.164 number. Maximum length of the telephone 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*—Telephone number to which calls are forwarded. Represents a fully qualified E.164 number. Maximum length of the telephone 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.

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

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.

	Command or Action	Purpose
Step 5	**after-hours day** *day start-time stop-time* **Example:** `Router(config-cm-fallback)# after-hours day mon 19:00 07:00`	Defines a recurring time period based on 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 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 its outgoing calls are blocked even though call blocking is enabled.
Step 10	**end** **Example:** `Router(config-register-pool)# end`	Returns to privileged EXEC mode.

Examples

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
```

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 regarding a specific pool.
- **debug ccsip message—**Debugs basic B2BUA calls.

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. 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.

Note Music on hold (MOH) is not supported for call hold invoked from a SIP phone. A caller hears only silence when placed on hold by a SIP phone.

Configuration Examples for Cisco Unified SIP SRST Features Using Back-to-Back User Agent Mode

This section provides the following configuration example.

- Cisco Unified SIP SRST: Example

Note IP addresses and hostnames in examples are fictitious.

Cisco Unified SIP SRST: Example

This section provides a configuration example to match the configuration tasks in the previous sections.

```
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 all 2412
 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
```

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