



## **Cisco IOS SIP SRST Version 3.4 System Administrator Guide**

Cisco IOS Release  
12.4(4)T  
October 2005

### **Corporate Headquarters**

Cisco Systems, Inc.  
170 West Tasman Drive  
San Jose, CA 95134-1706  
USA  
<http://www.cisco.com>  
Tel: 408 526-4000  
800 553-NETS (6387)  
Fax: 408 526-4100



THE SPECIFICATIONS AND INFORMATION REGARDING THE PRODUCTS IN THIS MANUAL ARE SUBJECT TO CHANGE WITHOUT NOTICE. ALL STATEMENTS, INFORMATION, AND RECOMMENDATIONS IN THIS MANUAL ARE BELIEVED TO BE ACCURATE BUT ARE PRESENTED WITHOUT WARRANTY OF ANY KIND, EXPRESS OR IMPLIED. USERS MUST TAKE FULL RESPONSIBILITY FOR THEIR APPLICATION OF ANY PRODUCTS.

THE SOFTWARE LICENSE AND LIMITED WARRANTY FOR THE ACCOMPANYING PRODUCT ARE SET FORTH IN THE INFORMATION PACKET THAT SHIPPED WITH THE PRODUCT AND ARE INCORPORATED HEREIN BY THIS REFERENCE. IF YOU ARE UNABLE TO LOCATE THE SOFTWARE LICENSE OR LIMITED WARRANTY, CONTACT YOUR CISCO REPRESENTATIVE FOR A COPY.

The Cisco implementation of TCP header compression is an adaptation of a program developed by the University of California, Berkeley (UCB) as part of UCB's public domain version of the UNIX operating system. All rights reserved. Copyright © 1981, Regents of the University of California.

NOTWITHSTANDING ANY OTHER WARRANTY HEREIN, ALL DOCUMENT FILES AND SOFTWARE OF THESE SUPPLIERS ARE PROVIDED "AS IS" WITH ALL FAULTS. CISCO AND THE ABOVE-NAMED SUPPLIERS DISCLAIM ALL WARRANTIES, EXPRESSED OR IMPLIED, INCLUDING, WITHOUT LIMITATION, THOSE OF MERCHANTABILITY, FITNESS FOR A PARTICULAR PURPOSE AND NONINFRINGEMENT OR ARISING FROM A COURSE OF DEALING, USAGE, OR TRADE PRACTICE.

IN NO EVENT SHALL CISCO OR ITS SUPPLIERS BE LIABLE FOR ANY INDIRECT, SPECIAL, CONSEQUENTIAL, OR INCIDENTAL DAMAGES, INCLUDING, WITHOUT LIMITATION, LOST PROFITS OR LOSS OR DAMAGE TO DATA ARISING OUT OF THE USE OR INABILITY TO USE THIS MANUAL, EVEN IF CISCO OR ITS SUPPLIERS HAVE BEEN ADVISED OF THE POSSIBILITY OF SUCH DAMAGES.

CCSP, CCVP, the Cisco Square Bridge logo, Follow Me Browsing, and StackWise are trademarks of Cisco Systems, Inc.; Changing the Way We Work, Live, Play, and Learn, and iQuick Study are service marks of Cisco Systems, Inc.; and Access Registrar, Aironet, ASIST, BPX, Catalyst, CCDA, CCDP, CCIE, CCIP, CCNA, CCNP, Cisco, the Cisco Certified Internetwork Expert logo, Cisco IOS, Cisco Press, Cisco Systems, Cisco Systems Capital, the Cisco Systems logo, Cisco Unity, Empowering the Internet Generation, Enterprise/Solver, EtherChannel, EtherFast, EtherSwitch, Fast Step, FormShare, GigaDrive, GigaStack, HomeLink, Internet Quotient, IOS, IP/TV, iQ Expertise, the iQ logo, iQ Net Readiness Scorecard, LightStream, Linksys, MeetingPlace, MGX, the Networkers logo, Networking Academy, Network Registrar, *Packet*, PIX, Post-Routing, Pre-Routing, ProConnect, RateMUX, ScriptShare, SlideCast, SMARTnet, StrataView Plus, TeleRouter, The Fastest Way to Increase Your Internet Quotient, and TransPath are registered trademarks of Cisco Systems, Inc. and/or its affiliates in the United States and certain other countries.

All other trademarks mentioned in this document or Website are the property of their respective owners. The use of the word partner does not imply a partnership relationship between Cisco and any other company. (0502R)

*Cisco IOS SIP SRST Version 3.4 System Administrator Guide*  
Copyright © 2005 Cisco Systems, Inc. All rights reserved.



<b>Cisco IOS SIP SRST Feature Roadmap</b>	<b>1</b>
Contents	1
Documentation Organization	1
Feature Roadmap	3
<b>Cisco IOS SIP SRST Feature Overview</b>	<b>5</b>
Contents	5
Cisco SIP SRST Description	5
Support for Cisco IP Phones and Platforms	7
Finding Cisco IOS Software Releases That Support Cisco SRST	7
Cisco IP Phone Support	8
Platform and Memory Support	8
Prerequisites for Configuring Cisco SIP SRST	8
Restrictions for Configuring Cisco SIP SRST	10
Where to Go Next	11
Additional References	11
Related Documents	11
Standards	12
MIBs	12
RFCs	12
Technical Assistance	13
<b>Upgrading from Cisco IOS SIP SRST Version 3.0 to Version 3.4</b>	<b>15</b>
Contents	15
Comparison of Cisco SIP SRST Version 3.0 and Version 3.4	15
Procedures Necessary to Run Version 3.0 or Version 3.4	16
How to Upgrade from Cisco SIP SRST Version 3.0 to Version 3.4	17
Disabling Call Redirection	17
Enabling SIP-to-SIP Connection Capabilities	20
<b>Configuring the SIP Registrar</b>	<b>23</b>
Contents	23

Prerequisites for Configuring the SIP Registrar	23
Restrictions for Configuring the SIP Registrar	23
Information About Configuring the SIP Registrar	24
How to Configure the SIP Registrar	24
Configuring the SIP Registrar	24
Configuring Backup Registrar Service to SIP Phones	26
Configuring Backup Registrar Service to SIP Phones (Using Optional Commands)	30
Verifying SIP Registrar Configuration	33
Verifying Proxy Dial-Peer Configuration	34
<b>Configuring Cisco SIP SRST Version 3.0 Features Using Redirect Mode</b>	<b>39</b>
Contents	39
Prerequisites for Cisco SIP SRST Version 3.0 Features Using Redirect Mode	39
Restrictions for Cisco SIP SRST Version 3.0 Features Using Redirect Mode	39
Information About Cisco SIP SRST Version 3.0 Features Using Redirect Mode	40
How to Configure Cisco SIP SRST Version 3.0 Features Using Redirect Mode	40
Configuring Call Redirect Enhancements to Support Calls Between SIP IP Phones for Cisco SIP SRST	41
Configuring Sending 300 Multiple Choice Support	43
Configuration Examples for Cisco SIP SRST Version 3.0 Features Using Redirect Mode	45
Cisco SIP SRST Version 3.0: Example	45
Where to Go Next	46
<b>Configuring Cisco SIP SRST Version 3.4 Features Using Back-to-Back User Agent Mode</b>	<b>47</b>
Contents	47
Prerequisites for Cisco SIP SRST Version 3.4 Features Using Back-to-Back User Agent Mode	47
Restrictions for Cisco SIP SRST Version 3.4 Features Using Back-to-Back User Agent Mode	48
Information About Cisco SIP SRST Version 3.4 Features Using Back-to-Back User Agent Mode	48
Cisco SIP SRST and Cisco SIP CallManager Express Feature Crossover	48
How to Configure Cisco SIP SRST Version 3.4	50
Configuring SIP Phone Features	51
Configuring SIP-to-SIP Call Forwarding	53
Configuring Call Blocking Based on Time of Day, Day of Week, or Date	55
SIP Call Hold and Resume	58
Configuration Examples for Cisco SIP SRST Version 3.4 Features Using Back-to-Back User Agent Mode	59

Cisco SIP SRST Version 3.4: Example 59

---

**INDEX**





# Cisco IOS SIP SRST Feature Roadmap

---

This chapter contains a summary of Cisco IOS 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, page 1](#)
- [Feature Roadmap, page 3](#)

## Documentation Organization

This book consists of the following chapters as shown in [Table 1](#).

**Table 1** Cisco SIP SRST Configuration Sequence

Chapter or Appendix	Description
<a href="#">Cisco IOS SIP SRST Feature Overview</a>	Gives a brief description of Cisco SIP SRST and provides information on the supported platforms and Cisco IP phones. In addition, it describes any prerequisites or restrictions that should be addressed before Cisco SIP SRST is configured.
<a href="#">Upgrading from Cisco IOS SIP SRST Version 3.0 to Version 3.4</a>	Describes the two versions of Cisco 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: <ul style="list-style-type: none"> <li>• <a href="#">Disabling Call Redirection</a></li> <li>• <a href="#">Enabling SIP-to-SIP Connection Capabilities</a></li> </ul>
<a href="#">Configuring the SIP Registrar</a>	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: <ul style="list-style-type: none"> <li>• <a href="#">Configuring the SIP Registrar</a></li> <li>• <a href="#">Configuring Backup Registrar Service to SIP Phones</a></li> <li>• <a href="#">Configuring Backup Registrar Service to SIP Phones (Using Optional Commands)</a></li> <li>• <a href="#">Verifying SIP Registrar Configuration</a></li> <li>• <a href="#">Verifying Proxy Dial-Peer Configuration</a></li> </ul>
<a href="#">Configuring Cisco SIP SRST Version 3.0 Features Using Redirect Mode</a>	Describes features necessary for Version 3.0. This chapter includes the following tasks: <ul style="list-style-type: none"> <li>• <a href="#">Configuring Call Redirect Enhancements to Support Calls Between SIP IP Phones for Cisco SIP SRST</a></li> <li>• <a href="#">Configuring Sending 300 Multiple Choice Support</a></li> </ul>
<a href="#">Configuring Cisco SIP SRST Version 3.4 Features Using Back-to-Back User Agent Mode</a>	Describes features available in Version 3.4. Features include Cisco SIP SRST 3.4 support for standardized RFC 3261 SIP phones. This chapter includes the following tasks: <ul style="list-style-type: none"> <li>• <a href="#">Configuring SIP Phone Features</a></li> <li>• <a href="#">Configuring SIP-to-SIP Call Forwarding</a></li> <li>• <a href="#">Configuring Call Blocking Based on Time of Day, Day of Week, or Date</a></li> </ul>



# Feature Roadmap

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

**Table 2** Cisco SIP SRST Features by Cisco IOS Release

Cisco SIP SRST Version	Cisco IOS Release	Modifications
Version 3.4	12.4(4)T	The <i>Cisco SIP SRST Version 3.4 System Administrator Guide</i> was created. It includes the following features: <ul style="list-style-type: none"> <li>• <a href="#">Upgrading from Cisco IOS SIP SRST Version 3.0 to Version 3.4</a></li> <li>• <a href="#">Configuring Cisco SIP SRST Version 3.0 Features Using Redirect Mode (formerly SIP SRST)</a></li> <li>• <a href="#">Configuring Cisco SIP SRST Version 3.4 Features Using Back-to-Back User Agent Mode</a></li> </ul>
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 IOS SIP SRST Feature Overview

---

This chapter includes information about supported Cisco IP phones and platforms. It also includes information on Cisco IOS SIP SRST specifications, features, prerequisites, restrictions, and where to find additional reference documents.



## Note

For the most up-to-date information about Cisco IP phone support, the maximum number of Cisco IP phones, the maximum number of DNs or virtual voice ports, and memory requirements for Cisco SRST and Cisco SIP SRST, see the *Cisco Survivable Remote Site Telephony (SRST) 3.4 Specifications for Cisco IOS Release 12.4(4)T* at the following URL:

<http://www.cisco.com/univercd/cc/td/doc/product/voice/srst/srst34/srs34spc.htm>

---

## Contents

- [Cisco SIP SRST Description, page 5](#)
- [Support for Cisco IP Phones and Platforms, page 7](#)
- [Prerequisites for Configuring Cisco SIP SRST, page 8](#)
- [Restrictions for Configuring Cisco SIP SRST, page 10](#)
- [Where to Go Next, page 11](#)
- [Additional References, page 11](#)

## Cisco SIP SRST Description

This book describes Survivable Remote Site Telephony (SRST) functionality for Session Initiation Protocol (SIP) networks. Cisco 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 SIP SRST can support SIP phones with standard RFC 3261 feature support locally and across SIP WAN networks. With Cisco SIP SRST 3.4, SIP phones can place calls across SIP networks in the same way as SCCP phones.

Cisco 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 SIP SRST functionality on a SIP PSTN gateway provides service reliability for SIP-based IP phones in the event of a WAN outage. Cisco 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*

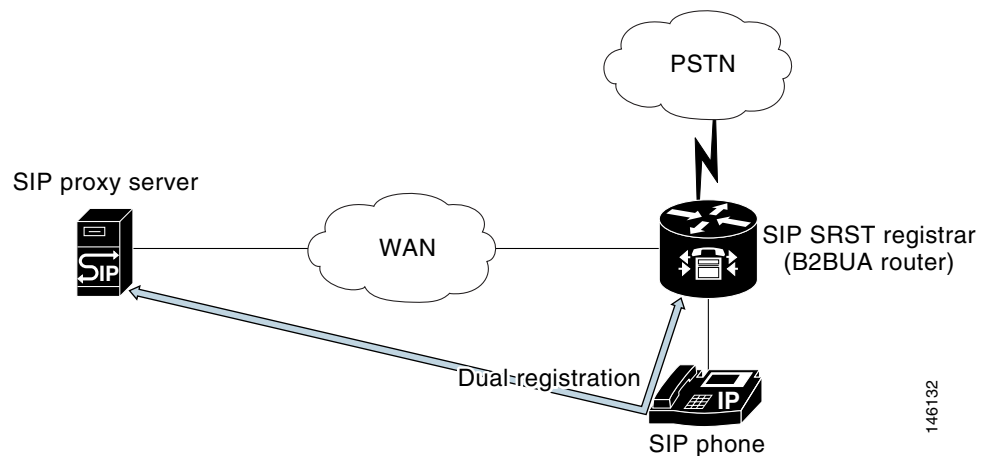
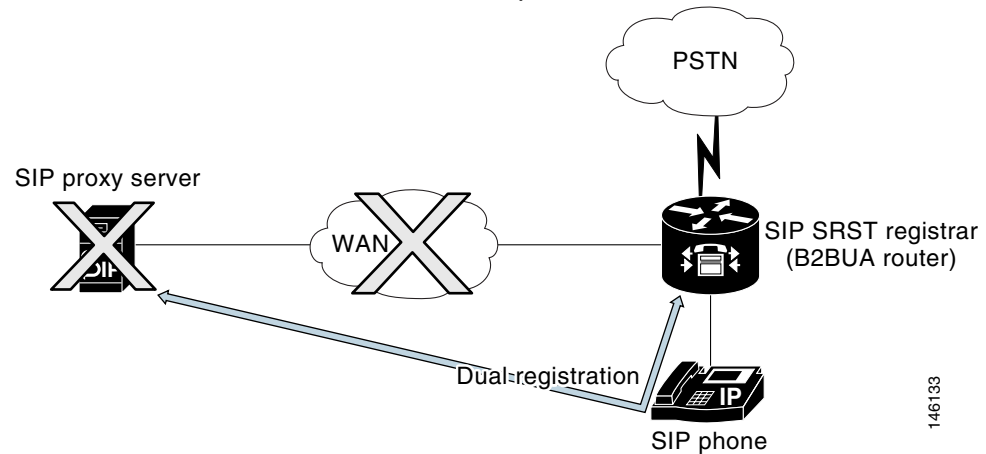


Figure 2 shows that when the WAN or SIP proxy server goes down, the call from the SIP phone cannot get to the SIP proxy server and instead goes through the B2BUA router out to the PSTN.

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



146133

## Support for Cisco IP Phones and Platforms

The following sections provide information about Cisco Feature Navigator and the histories of Cisco IP phone and platform support from Cisco SRST Version 3.0 to the present version.

- [Finding Cisco IOS Software Releases That Support Cisco SRST, page 7](#)
- [Cisco IP Phone Support, page 8](#)
- [Platform and Memory Support, page 8](#)

## Finding Cisco IOS Software Releases That Support Cisco SRST

The tables in this chapter list only the Cisco IOS software releases that first introduce new features to Cisco SRST. Other Cisco IOS software releases may subsequently inherit versions of Cisco SRST. To get a list of Cisco IOS software releases that support a particular version of Cisco 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](mailto: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 IP Phone Support

For the most up-to-date information about Cisco IP phone support, see the *Cisco Survivable Remote Site Telephony (SRST) 3.4 Specifications for Cisco IOS Software Release 12.4(4)T* at the following URL:

<http://www.cisco.com/univercd/cc/td/doc/product/voice/srst/srst34/srs34spc.htm>

Cisco 7940G and Cisco 7960G IP phones 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 SIP SRST device (redirect server or back-to-back user agent) at the same time. If this requirement is not met, the Cisco SIP SRST device may not be capable of routing incoming calls to the SIP phone until the SIP phone registers with the Cisco SIP SRST device. Cisco IP 7940G and 7960G phones 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 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 IP phones 7905G, Cisco 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 SIP SRST.

## Platform and Memory Support

For the most up-to-date information about platform and memory support, see the *Cisco Survivable Remote Site Telephony (SRST) 3.4 Specifications for Cisco IOS Software Release 12.4(4)T* at the following URL:

<http://www.cisco.com/univercd/cc/td/doc/product/voice/srst/srst34/srs34spc.htm>

## Prerequisites for Configuring Cisco SIP SRST

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

- An SRST feature license is required to enable the Cisco SIP SRST feature. Please contact your account representative if you have further questions.
- Cisco 7940G and Cisco 7960G IP phones 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 SIP SRST device (redirect server or back-to-back user agent) at the same time. If this requirement is not met, the Cisco SIP SRST device may not be capable of routing incoming calls to the SIP phone until the SIP phone registers with the Cisco SIP SRST device. Cisco IP 7940G and 7960G phones 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 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 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 SIP SRST router more rapidly.


Dual registration is not supported on the Cisco IP phone 7905G, Cisco 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 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.
- For Cisco SIP SRST Version 3.0, SIP phones must support 300 or 302 Redirect messages.

# Restrictions for Configuring Cisco SIP SRST

Table 3 provides a history of restrictions from Cisco SIP SRST Version 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 3.4	12.4(4)T	<p><b>Not Supported</b></p> <ul style="list-style-type: none"> <li>• 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.</li> <li>• 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.</li> <li>• SIP-NAT is not supported.</li> <li>• Cisco Unity Express is not supported.</li> <li>• Transcoding is not supported.</li> </ul> <p><b>Phone Features</b></p> <ul style="list-style-type: none"> <li>• For call waiting to work on the Cisco ATA and Cisco IP Phones 7912 and 7905G with a 1.0(2) build, the incoming call leg should be configured with the G.711 codec.</li> </ul> <p> <b>Note</b> Cisco IP phones 7905G, Cisco 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 SIP SRST.</p> <hr/> <p><b>General</b></p> <ul style="list-style-type: none"> <li>• Call detail records (CDRs) are only supported by standard IOS RADIUS support; CDRs are not supported otherwise.</li> <li>• All calls must use the same codec, either G.729r8 or G.711.</li> <li>• Calls that have been transferred cannot be transferred a second time.</li> <li>• URL dialing is not supported. Only number dialing is supported.</li> <li>• The SIP registrar functionality provided by Cisco SIP SRST provides no security or authentication services.</li> <li>• 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 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).</li> <li>• SIP-phone-to-SIP-trunk support requires Refer and 302/300 Redirection to be supported by the SIP trunk (Version 3.0).</li> </ul>
Version 3.2	12.3(11)T	
Version 3.1	12.3(7)T	
Version 3.0	12.2(15)ZJ 12.3(4)T	



## Where to Go Next

The next chapters of this book describe how to configure Cisco 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 SRST is to ensure that the basic software and hardware in your system are configured correctly for Cisco SRST. For instructions, see the [“Prerequisites for Configuring Cisco SIP SRST” section on page 8](#).

**Table 4** Cisco SRST Configuration Sequence

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

## Additional References

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

- [Related Documents, page 11](#)
- [Standards, page 12](#)
- [MIBs, page 12](#)
- [RFCs, page 12](#)
- [Technical Assistance, page 13](#)

## Related Documents

Related Topic	Documents
SRST commands and specifications	<ul style="list-style-type: none"> <li>• <a href="#">Cisco IOS Survivable Remote Site Telephony Version 3.4 Command Reference</a></li> <li>• <a href="#">Cisco Survivable Remote Site Telephony (SRST) 3.4 Specifications for Cisco IOS Release 12.4(4)T</a></li> </ul>
SRST administration	<ul style="list-style-type: none"> <li>• <a href="#">Cisco IOS Survivable Remote Site Telephony Version 3.4 Administrator Guide</a></li> </ul>

Related Topic	Documents
Cisco IP phones	<ul style="list-style-type: none"> <li>• <a href="#">Cisco IP Phone 7902 Quick Start Guide</a></li> <li>• <a href="#">Cisco IP Phone 7902G Quick Start Guide</a></li> <li>• <a href="#">At a Glance Cisco IP Phone 7912G</a></li> <li>• <a href="#">Cisco IP Phone 7960 and 7940 Series User Guide</a></li> <li>• <a href="#">Cisco IP Phone 7970 Guide</a></li> <li>• <a href="#">Cisco SIP IP Phone 7960 Administrator Guide, Version 5.1</a></li> </ul>
Cisco SIP functionality	<ul style="list-style-type: none"> <li>• <a href="#">Cisco IOS SIP Configuration Guide</a></li> </ul>
Command reference information for voice and telephony commands	<ul style="list-style-type: none"> <li>• <a href="#">Cisco IOS Voice Command Reference</a></li> <li>• <a href="#">Cisco IOS Debug Command Reference</a></li> </ul>
Standard preface	<ul style="list-style-type: none"> <li>• <a href="#">Cisco IOS Voice Configuration Library Preface</a></li> </ul>
Standard glossary	<ul style="list-style-type: none"> <li>• <a href="#">Cisco IOS Voice Configuration Library Glossary</a></li> </ul>

## 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.	<p>To locate and download MIBs for selected platforms, Cisco IOS releases, and feature sets, use Cisco MIB Locator found at the following URL:</p> <p><a href="http://www.cisco.com/go/mibs">http://www.cisco.com/go/mibs</a></p>

## RFCs

RFC	Title
RFC 2543	<a href="#">SIP: Session Initiation Protocol</a>
RFC 3261	<a href="#">SIP: Session Initiation Protocol</a>

## 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.	<a href="http://www.cisco.com/techsupport">http://www.cisco.com/techsupport</a>





# Upgrading from Cisco IOS SIP SRST Version 3.0 to Version 3.4

---

This chapter describes the changes necessary when upgrading from Version 3.0 to Version 3.4.



**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 Version 3.0 and Version 3.4, page 15](#)
- [Procedures Necessary to Run Version 3.0 or Version 3.4, page 16](#)
- [How to Upgrade from Cisco SIP SRST Version 3.0 to Version 3.4, page 17](#)

## Comparison of Cisco SIP SRST Version 3.0 and Version 3.4

### **Cisco IOS SIP SRST V3.0, Cisco IOS Release 12.2(15)ZJ Through Cisco IOS Release 12.4**

Cisco IOS SIP SRST V3.0 was the predecessor to Cisco SIP SRST V3.4. In Cisco SIP SRST Version 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 IOS SIP SRST V3.4, Cisco IOS Release 12.4(4)T**

With Cisco IOS SIP SRST V3.4, a SIP redirect server is no longer 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.

## Procedures Necessary to Run Version 3.0 or Version 3.4

The table below lists the high-level steps you need to take to upgrade to Cisco SIP SRST Version 3.4. It also lists the high-level steps you need to take in order to run Version 3.0 or Version 3.4.

Cisco SIP SRST Version	Instructions and Procedures
<p>If you are currently running Version 3.0 and want to upgrade to Version 3.4, complete these procedures.</p>	<p>Since Version 3.4 uses a B2BUA and not a redirect server, call redirection must be disabled as described in the following task:</p> <ul style="list-style-type: none"> <li>• <a href="#">Disabling Call Redirection</a></li> </ul> <p>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:</p> <ul style="list-style-type: none"> <li>• <a href="#">Enabling SIP-to-SIP Connection Capabilities</a></li> </ul> <p>You are now ready to configure Version 3.4 features such as call blocking and call forwarding. The following chapter describes the call blocking and call forwarding configurations:</p> <ul style="list-style-type: none"> <li>• <a href="#">Configuring Cisco SIP SRST Version 3.4 Features Using Back-to-Back User Agent Mode</a></li> </ul>
<p>If you are interested in Cisco SIP SRST Version 3.0 (using a redirect server), complete these procedures.</p>	<p>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.</p> <ul style="list-style-type: none"> <li>• <a href="#">Configuring the SIP Registrar</a></li> <li>• <a href="#">Configuring Backup Registrar Service to SIP Phones</a></li> <li>• <a href="#">Configuring Cisco SIP SRST Version 3.0 Features Using Redirect Mode</a></li> </ul>

Cisco SIP SRST Version	Instructions and Procedures
<p>If you are interested in Cisco SIP SRST Version 3.4 (using a B2BUA) and have never used Cisco SIP SRST in the past, complete these procedures.</p>	<p>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:</p> <ul style="list-style-type: none"> <li>• <a href="#">Enabling SIP-to-SIP Connection Capabilities</a></li> </ul> <p>SIP registrar functionality in Cisco IOS software is a required part of Cisco 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:</p> <ul style="list-style-type: none"> <li>• <a href="#">Configuring the SIP Registrar</a></li> </ul> <p>Configure a basic voice register pool:</p> <ul style="list-style-type: none"> <li>• <a href="#">Configuring Backup Registrar Service to SIP Phones</a></li> </ul> <p>You are now ready to configure Version 3.4 features such as call blocking and call forwarding. The following chapter describes the call blocking and call forwarding configurations:</p> <ul style="list-style-type: none"> <li>• <a href="#">Configuring Cisco SIP SRST Version 3.4 Features Using Back-to-Back User Agent Mode</a></li> </ul>

## How to Upgrade from Cisco SIP SRST Version 3.0 to Version 3.4

This section contains the following procedures:

- [Disabling Call Redirection, page 17](#) (required)
- [Enabling SIP-to-SIP Connection Capabilities, page 20](#) (required)

### Disabling Call Redirection

Because Version 3.4 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 18](#)

## 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	<b>enable</b>  <b>Example:</b> Router> enable	Enables privileged EXEC mode. <ul style="list-style-type: none"> <li>• Enter your password if prompted.</li> </ul>
Step 2	<b>configure terminal</b>  <b>Example:</b> Router# configure terminal	Enters global configuration mode.
Step 3	<b>voice service voip</b>  <b>Example:</b> Router(config)# voice service voip	Enters voice service configuration mode.
Step 4	<b>no redirect ip2ip</b>  <b>Example:</b> 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	<b>end</b>  <b>Example:</b> 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**

	<b>Command or Action</b>	<b>Purpose</b>
<b>Step 1</b>	<b>enable</b>  <b>Example:</b> Router> enable	Enables privileged EXEC mode. <ul style="list-style-type: none"> <li>• Enter your password if prompted.</li> </ul>
<b>Step 2</b>	<b>configure terminal</b>  <b>Example:</b> Router# configure terminal	Enters global configuration mode.
<b>Step 3</b>	<b>dial-peer voice <i>tag</i> voip</b>  <b>Example:</b> Router(config)# dial-peer voice 25 voip	Enters dial-peer configuration mode. <ul style="list-style-type: none"> <li>• <i>tag</i>—A number that uniquely identifies the dial peer (this number has local significance only).</li> <li>• <b>voip</b>—Indicates that this is a VoIP peer using voice encapsulation on the POTS network and is used for configuring redirect.</li> </ul>
<b>Step 4</b>	<b>no redirect ip2ip</b>  <b>Example:</b> 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.
<b>Step 5</b>	<b>end</b>  <b>Example:</b> 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 SIP SRST Version 3.4 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	<b>enable</b>  <b>Example:</b> Router> enable	Enables privileged EXEC mode. <ul style="list-style-type: none"> <li>• Enter your password if prompted.</li> </ul>
Step 2	<b>configure terminal</b>  <b>Example:</b> Router# configure terminal	Enters global configuration mode.
Step 3	<b>voice service voip</b>  <b>Example:</b> Router(config)# voice service voip	Enters voice service configuration mode to establish global call transfer and forwarding parameters.
Step 4	<b>allow-connections sip to sip</b>  <b>Example:</b> Router(config-voi-srv)# allow-connections sip to sip	Enables VoIP-to-VoIP call connections. Use the <b>no</b> form of the command to disable VoIP-to-VoIP connections, which is the default.
Step 5	<b>end</b>  <b>Example:</b> Router(conf-voi-srv)# end	Returns to privileged EXEC mode.

## What to Do Next

SIP registrar functionality in Cisco IOS software is a required part of Cisco SIP SRST. By default, Cisco 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 SIP SRST Version 3.4 Features Using Back-to-Back User Agent Mode”](#) chapter.





## Configuring the SIP Registrar

---

SIP registrar functionality in Cisco IOS software is an essential part of both Version 3.0 and Version 3.4 Cisco IOS SIP 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, page 23](#)
- [Restrictions for Configuring the SIP Registrar, page 23](#)
- [Information About Configuring the SIP Registrar, page 24](#)
- [How to Configure the SIP Registrar, page 24](#)

## Prerequisites for Configuring the SIP Registrar

Complete the prerequisites documented in the “[Prerequisites for Configuring Cisco SIP SRST](#)” section in the “[Cisco IOS SIP SRST Feature Overview](#)” chapter.

## Restrictions for Configuring the SIP Registrar

See the restrictions documented in the “[Restrictions for Configuring Cisco SIP SRST](#)” section in the “[Cisco IOS SIP SRST Feature Overview](#)” chapter.

# Information About Configuring the SIP Registrar

Cisco 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 SIP SRST device also provides PSTN gateway access for placing and receiving PSTN calls.

To make maximum use of the Cisco SIP SRST service, the local SIP IP phones should support dual (concurrent) registration with both their primary SIP proxy or registrar and the Cisco SIP SRST backup registrar. Cisco 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 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 SIP SRST functionality), DNS servers may not be available.

SIP registrar functionality is enabled with the following configuration. By default, Cisco 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 SIP SRST device (redirect server) at the same time. If this requirement is not met, the Cisco 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	<b>enable</b>  <b>Example:</b> Router> enable	Enables privileged EXEC mode. <ul style="list-style-type: none"> <li>• Enter your password if prompted.</li> </ul>
Step 2	<b>configure terminal</b>  <b>Example:</b> Router# configure terminal	Enters global configuration mode.
Step 3	<b>voice service voip</b>  <b>Example:</b> Router(config)# voice service voip	Enters voice service configuration mode.
Step 4	<b>sip</b>  <b>Example:</b> Router(config-voi-srv)# sip	Enters SIP configuration mode.

	Command or Action	Purpose
Step 5	<pre>registrar server [expires [max sec] [min sec]]</pre> <p><b>Example:</b> Router(conf-serv-sip)# registrar server expires max 600 min 60</p>	<p>Enables SIP registrar functionality. The keywords and arguments are defined as follows:</p> <ul style="list-style-type: none"> <li>• <b>expires:</b> (Optional) Sets the active time for an incoming registration.</li> <li>• <b>max sec:</b> (Optional) Maximum expiration time for a registration, in seconds. The range is from 600 to 86400. The default is 3600.</li> <li>• <b>min sec:</b> (Optional) Minimum expiration time for a registration, in seconds. The range is from 60 to 3600. The default is 60.</li> </ul>
Step 6	<pre>end</pre> <p><b>Example:</b> Router(conf-serv-sip)# end</p>	<p>Returns to privileged EXEC mode.</p>

## 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 SIP SRST device and which can be rejected. Registrations that match this pool create VoIP SIP dial peers with the dial-peer attributes set to these configurations. Although only the **id** command is mandatory, this configuration example shows basic functionality.

For command-level information, see the appropriate command page in the *Cisco IOS SIP SRST Version 3.4 Command Reference*.

## 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 SIP SRST. When a SIP phone registers to Cisco 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	<b>enable</b>  <b>Example:</b> Router> enable	Enables privileged EXEC mode.  • Enter your password if prompted.
Step 2	<b>configure terminal</b>  <b>Example:</b> Router# configure terminal	Enters global configuration mode.

	Command or Action	Purpose
Step 3	<p><b>call fallback active</b></p> <p><b>Example:</b> Router(config)# call fallback active</p>	<p>Enables a call request to fall back to alternate dial peers in case of network congestion.</p> <ul style="list-style-type: none"> <li>This command is useful to monitor the proxy dial peer and fallback to the next preferred dial peer. For full information on the <b>call fallback active</b> command, see the <a href="#">PSTN Fallback Feature</a>.</li> </ul>
Step 4	<p><b>voice register pool tag</b></p> <p><b>Example:</b> Router(config)# voice register pool 12</p>	<p>Enters voice register pool configuration mode for SIP phones.</p> <ul style="list-style-type: none"> <li>Use this command to control which registrations are accepted or rejected by a Cisco SIP SRST device.</li> </ul>
Step 5	<p><b>id {network address mask mask   ip address mask mask   mac address}</b></p> <p><b>Example:</b> Router(config-register-pool)# id network 172.16.0.0 mask 255.255.0.0</p>	<p>Explicitly identifies a locally available individual or set of SIP IP phones. The keywords and arguments are defined as follows:</p> <ul style="list-style-type: none"> <li><b>network address mask mask:</b> The <b>network address mask mask</b> keyword/argument combination is used to accept SIP Register messages for the indicated phone numbers from any IP phone within the indicated IP subnet.</li> <li><b>ip address mask mask:</b> The <b>ip address mask mask</b> keyword/argument combination is used to identify an individual phone.</li> <li><b>mac address:</b> MAC address of a particular Cisco IP phone.</li> </ul>
Step 6	<p><b>preference preference-order</b></p> <p><b>Example:</b> Router(config-register-pool)# preference 2</p>	<p>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.</p> <ul style="list-style-type: none"> <li>The preference must be greater (lower priority) than the preference configured with the <b>preference</b> keyword in the <b>proxy</b> command.</li> </ul>

	Command or Action	Purpose
<b>Step 7</b>	<p><b>proxy</b> <i>ip-address</i> [<b>preference</b> <i>value</i>] [<b>monitor probe</b> {<b>icmp-ping</b>   <b>rtr</b>} [<i>alternate-ip-address</i>]]</p> <p><b>Example:</b>  Router(config-register-pool)# proxy  10.2.161.187 preference 1</p>	<p>Autogenerates additional VoIP dial peers to reach the main SIP proxy whenever a Cisco SIP IP phone registers with a SIP SRST gateway. The keywords and arguments are defined as follows:</p> <ul style="list-style-type: none"> <li>• <i>ip-address</i>: IP address of the SIP proxy.</li> <li>• <b>preference value</b>: (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 <b>preference</b> command. Range is from 0 to 10. The highest preference is 0. There is no default.</li> <li>• <b>monitor probe</b>: (Optional) Enables monitoring of proxy dial peers.</li> <li>• <b>icmp-ping</b>: Enables monitoring of proxy dial peers using ICMP ping.</li> </ul> <p> <b>Note</b> 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.</p> <ul style="list-style-type: none"> <li>• <b>rtr</b>: Enables monitoring of proxy dial peers using RTR probes.</li> <li>• <i>alternate-ip-address</i>: (Optional) Enables monitoring of alternate IP addresses other than the proxy address. For example, to monitor a gateway front end to a SIP proxy.</li> </ul>
<b>Step 8</b>	<p><b>voice-class codec</b> <i>tag</i></p> <p><b>Example:</b>  Router(config-register-pool)# voice-class codec  15</p>	<p>Sets the voice class codec parameters. The <i>tag</i> argument is a codec group number between 1 and 10000.</p>
<b>Step 9</b>	<p><b>application</b> <i>application-name</i></p> <p><b>Example:</b>  Router(config-register-pool)# application  SIP.App</p>	<p>Selects the session-level application on the VoIP dial peer. Use the <i>application-name</i> argument to define a specific interactive voice response (IVR) application.</p>
<b>Step 10</b>	<p><b>end</b></p> <p><b>Example:</b>  Router(config-register-pool)# end</p>	<p>Returns to privileged EXEC mode.</p>

## 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	<p><b>enable</b></p> <p><b>Example:</b> Router&gt; enable</p>	<p>Enables privileged EXEC mode.</p> <ul style="list-style-type: none"> <li>Enter your password if prompted.</li> </ul>
Step 2	<p><b>configure terminal</b></p> <p><b>Example:</b> Router# configure terminal</p>	<p>Enters global configuration mode.</p>
Step 3	<p><b>voice register pool tag</b></p> <p><b>Example:</b> Router(config)# voice register pool 12</p>	<p>Enters voice register pool configuration mode.</p> <ul style="list-style-type: none"> <li>Use this command to control which registrations are accepted or rejected by a Cisco SIP SRST device.</li> </ul>
Step 4	<p><b>translate-outgoing {called   calling} rule-tag</b></p> <p><b>Example:</b> Router(config-register-pool)# translate-outgoing called 1</p>	<p>Allows explicit setting of translation rules on the VoIP dial peer to modify a phone number dialed by a Cisco IP phone user.</p> <ul style="list-style-type: none"> <li>The <i>rule-tag</i> argument is the reference number of the translation rule. Valid entries are 1 to 2147483647.</li> </ul>
Step 5	<p><b>alias tag pattern to target [preference value]</b></p> <p><b>Example:</b> Router(config-register-pool)# alias 1 94... to 91011 preference 8</p>	<p>Allows Cisco 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:</p> <ul style="list-style-type: none"> <li><i>tag</i>: Number from 1 to 5 and the distinguishing factor when there are multiple <b>alias</b> commands.</li> <li><i>pattern</i>: The prefix number; matches the incoming telephone number and may include wildcards.</li> <li><b>to</b>: Connects the tag number pattern to the alternate number.</li> <li><i>target</i>: The target number; an alternate telephone number to route incoming calls to match the number pattern.</li> <li><b>preference value</b>: (Optional) Assigns a dial-peer preference value to the alias. The <i>value</i> argument is the value of the associated dial peer, and the range is from 1 to 10. There is no default.</li> </ul>

Command or Action	Purpose
<p><b>Step 6</b></p> <pre>cor {incoming   outgoing} cor-list-name {cor-list-number starting-number [- ending-number]   default}</pre> <p><b>Example:</b> Router(config-register-pool)# cor incoming call91 1 91011</p>	<p>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:</p> <ul style="list-style-type: none"> <li>• <b>incoming:</b> COR list to be used by incoming dial peers.</li> <li>• <b>outgoing:</b> COR list to be used by outgoing dial peers.</li> <li>• <i>cor-list-name:</i> COR list name.</li> <li>• <i>cor-list-number:</i> COR list identifier. The maximum number of COR lists that can be created is four, comprised of incoming or outgoing dial peers.</li> <li>• <i>starting-number:</i> Start of a directory number range, if an ending number is included. Can also be a standalone number.</li> <li>• <b>- :</b> (Optional) Indicator that a full range is configured.</li> <li>• <i>ending-number:</i> (Optional) End of a directory number range.</li> <li>• <b>default:</b> Instructs the router to use an existing default COR list.</li> </ul>
<p><b>Step 7</b></p> <pre>incoming called-number [number]</pre> <p><b>Example:</b> Router(config-register-pool)# incoming called-number 308</p>	<p>Applies incoming called parameters to dynamically created dial peers. The <i>number</i> argument is optional and indicates a sequence of digits that represent a phone number prefix.</p>
<p><b>Step 8</b></p> <pre>number tag number-pattern [preference value] [huntstop]</pre> <p><b>Example:</b> Router(config-register-pool)# number 1 50.. preference 2</p>	<p>Indicates the E.164 phone numbers that the registrar permits to handle the Register message from the Cisco SIP IP phone. The keywords and arguments are defined as follows:</p> <ul style="list-style-type: none"> <li>• <i>tag:</i> Number from 1 to 10 and the distinguishing factor when there are multiple <b>number</b> commands.</li> <li>• <i>number-pattern:</i> Phone numbers (including wildcards and patterns) that are permitted by the registrar to handle the Register message from the SIP IP phone.</li> <li>• <b>preference value:</b> (Optional) Defines the number list preference order.</li> <li>• <b>huntstop:</b> (Optional) Stops hunting if the dial peer is busy.</li> </ul>

	Command or Action	Purpose
Step 9	<b>dtmf-relay</b> [ <b>cisco-rtp</b> ] [ <b>rtp-nte</b> ] [ <b>sip-notify</b> ]  <b>Example:</b> 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: <ul style="list-style-type: none"> <li>• <b>cisco-rtp</b>: (Optional) Forwards DTMF tones by using Real-Time Transport Protocol (RTP) with a Cisco proprietary payload type.</li> <li>• <b>rtp-nte</b>: (Optional) Forwards DTMF tones by using RTP with the Named Telephone Event (NTE) payload type.</li> <li>• <b>sip-notify</b>: (Optional) Forwards DTMF tones using SIP NOTIFY messages.</li> </ul>
Step 10	<b>end</b>  <b>Example:</b> 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 SIP SRST Version 3.0 features such as redirect mode, see the [“Configuring Cisco SIP SRST Version 3.0 Features Using Redirect Mode”](#) chapter.

To configure Cisco SIP SRST Version 3.4 features such as call forwarding or call blocking, see the [“Configuring Cisco SIP SRST Version 3.4 Features Using Back-to-Back User Agent Mode”](#) chapter.





# Configuring Cisco SIP SRST Version 3.0 Features Using Redirect Mode

---

This chapter describes Survivable Remote Site Telephony (SRST) functionality for Session Initiation Protocol (SIP) networks that was introduced in Version 3.0.



## 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 SIP SRST Version 3.0 Features Using Redirect Mode, page 39](#)
- [Restrictions for Cisco SIP SRST Version 3.0 Features Using Redirect Mode, page 39](#)
- [Information About Cisco SIP SRST Version 3.0 Features Using Redirect Mode, page 40](#)
- [How to Configure Cisco SIP SRST Version 3.0 Features Using Redirect Mode, page 40](#)
- [Configuration Examples for Cisco SIP SRST Version 3.0 Features Using Redirect Mode, page 45](#)
- [Where to Go Next, page 46](#)

## Prerequisites for Cisco SIP SRST Version 3.0 Features Using Redirect Mode

Complete the prerequisites documented in the “[Prerequisites for Configuring Cisco SIP SRST](#)” section in the “[Cisco IOS SIP SRST Feature Overview](#)” chapter.

## Restrictions for Cisco SIP SRST Version 3.0 Features Using Redirect Mode

See the restrictions documented in the “[Restrictions for Configuring Cisco SIP SRST](#)” section in the “[Cisco IOS SIP SRST Feature Overview](#)” chapter.

# Information About Cisco SIP SRST Version 3.0 Features Using Redirect Mode

Cisco 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 SIP SRST device also provides PSTN gateway access for placing and receiving PSTN calls.

To make maximum use of the Cisco SIP SRST service, the local SIP IP phones should support dual (concurrent) registration with both their primary SIP proxy or registrar and the Cisco SIP SRST backup registrar. Cisco 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 SIP SRST Version 3.0 Features Using Redirect Mode

This section contains the following procedures:

- [Configuring Call Redirect Enhancements to Support Calls Between SIP IP Phones for Cisco 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 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 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	<b>enable</b>  <b>Example:</b> Router> enable	Enables privileged EXEC mode.  • Enter your password if prompted.
Step 2	<b>configure terminal</b>  <b>Example:</b> Router# configure terminal	Enters global configuration mode.

	Command or Action	Purpose
Step 3	<code>voice service voip</code>  <b>Example:</b> Router(config)# <code>voice service voip</code>	Enters voice service configuration mode.
Step 4	<code>redirect ip2ip</code>  <b>Example:</b> Router(config-voi-srv)# <code>redirect ip2ip</code>	Redirects SIP phone calls to SIP phone calls globally on a gateway using the Cisco IOS voice gateway.
Step 5	<code>end</code>  <b>Example:</b> Router(config-voi-srv)# <code>end</code>	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 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	<b>enable</b>  <b>Example:</b> Router> enable	Enables privileged EXEC mode. <ul style="list-style-type: none"> <li>Enter your password if prompted.</li> </ul>
Step 2	<b>configure terminal</b>  <b>Example:</b> Router# configure terminal	Enters global configuration mode.
Step 3	<b>dial-peer voice tag voip</b>  <b>Example:</b> Router(config)# dial-peer voice 25 voip	Enters dial-peer configuration mode. <ul style="list-style-type: none"> <li><i>tag</i>—A number that uniquely identifies the dial peer (this number has local significance only).</li> <li><b>voip</b>—Indicates that this is a VoIP peer using voice encapsulation on the POTS network and is used for configuring redirect.</li> </ul>
Step 4	<b>application application-name</b>  <b>Example:</b> Router(config-dial-peer)# application session	Enables a specific application on a dial peer. <ul style="list-style-type: none"> <li>For SIP, the default Tool Command Language (Tcl) application (from the Cisco IOS image) is <b>session</b> and can be applied to both VoIP and POTS dial peers.</li> <li>The application must support IP-to-IP redirection.</li> </ul>
Step 5	<b>redirect ip2ip</b>  <b>Example:</b> 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	<b>end</b>  <b>Example:</b> 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	<b>enable</b>  <b>Example:</b> Router> enable	Enables privileged EXEC mode. <ul style="list-style-type: none"> <li>• Enter your password if prompted.</li> </ul>
Step 2	<b>configure terminal</b>  <b>Example:</b> Router# configure terminal	Enters global configuration mode.
Step 3	<b>voice service voip</b>  <b>Example:</b> Router(config)# voice service voip	Enters voice service configuration mode.
Step 4	<b>sip</b>  <b>Example:</b> Router(config-voi-srv)# sip	Enters SIP configuration mode.
Step 5	<b>redirect contact order [best-match   longest-match]</b>  <b>Example:</b> 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: <ul style="list-style-type: none"> <li>• <b>best-match</b>—(Optional) Uses the current system configuration to set the order of contacts.</li> <li>• <b>longest-match</b>—(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.</li> </ul>
Step 6	<b>end</b>  <b>Example:</b> Router(config-serv-sip)# end	Returns to privileged EXEC mode.

# Configuration Examples for Cisco SIP SRST Version 3.0 Features Using Redirect Mode

This section provides the following configuration example.

- [Cisco SIP SRST Version 3.0: Example](#)



**Note**

IP addresses and hostnames in examples are fictitious.

## Cisco SIP SRST Version 3.0: 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 SIP SRST, the [“Configuring Cisco SIP SRST Version 3.4 Features Using Back-to-Back User Agent Mode”](#) chapter describes additional configurations to increase SIP phone functionality.



# Configuring Cisco SIP SRST Version 3.4 Features Using Back-to-Back User Agent Mode

---

This chapter describes Cisco SRST 3.4 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 SIP SRST Version 3.4 Features Using Back-to-Back User Agent Mode](#), page 47
- [Restrictions for Cisco SIP SRST Version 3.4 Features Using Back-to-Back User Agent Mode](#), page 48
- [Information About Cisco SIP SRST Version 3.4 Features Using Back-to-Back User Agent Mode](#), page 48
- [How to Configure Cisco SIP SRST Version 3.4](#), page 50
- [Configuration Examples for Cisco SIP SRST Version 3.4 Features Using Back-to-Back User Agent Mode](#), page 59

## Prerequisites for Cisco SIP SRST Version 3.4 Features Using Back-to-Back User Agent Mode

- Complete the prerequisites documented in the “[Prerequisites for Configuring Cisco SIP SRST](#)” section in the “[Cisco IOS SIP SRST Feature Overview](#)” chapter.
- Complete the necessary tasks found in the “[Upgrading from Cisco IOS SIP SRST Version 3.0 to Version 3.4](#)” chapter. Specific tasks include the required task that is documented in the “[Enabling SIP-to-SIP Connection Capabilities](#)” section on page 20.

- 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 SIP SRST Version 3.4 Features Using Back-to-Back User Agent Mode

See the restrictions documented in the “[Restrictions for Configuring Cisco SIP SRST](#)” section in the “[Cisco IOS SIP SRST Feature Overview](#)” chapter.

## Information About Cisco SIP SRST Version 3.4 Features Using Back-to-Back User Agent Mode

A Cisco SRST system can now support SIP phones with standard-based RFC 3261 feature support locally and across SIP WAN networks. With Cisco SIP SRST 3.4, 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 SIP SRST 3.4 also uses a back-to-back user agent (B2BUA), which is a separate call agent that has more features than Cisco SIP SRST version 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 SIP SRST using a B2BUA.

## Cisco SIP SRST and Cisco SIP CallManager Express Feature Crossover

In Version 3.4 of Cisco SIP SRST there is a voice register dn configuration mode. However, in a typical Cisco 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 SIP SRST environment. The **voice register dn** commands are most likely to be used in a Cisco SIP CallManager Express (CME) environment. If you work in a Cisco SIP CME environment and would like to know which commands are also applicable to Cisco 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 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 SRST and Cisco CME (Sorted by Configuration Mode)

Function—>Command	Dial Peer	Voice Register Mode	Configurable for Cisco (SIP) CME and Cisco SIP SRST	Applicable to Cisco (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
voicemail	—	global	—	X
after-hour exempt	X	pool	X	—

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

Function—>Command	Dial Peer	Voice Register Mode	Configurable for Cisco (SIP) CME and Cisco SIP SRST	Applicable to Cisco (SIP) CME Only
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 SIP SRST Version 3.4

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 configuration 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 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	<b>enable</b>  <b>Example:</b> Router> enable	Enables privileged EXEC mode.  • Enter your password if prompted.
Step 2	<b>configure terminal</b>  <b>Example:</b> Router# configure terminal	Enters global configuration mode.
Step 3	<b>voice register global</b> <i>tag</i>  <b>Example:</b> 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 SIP SRST environment.

	Command or Action	Purpose
Step 4	<p><b>max-pool</b> <i>max-voice-register-pools</i></p> <p><b>Example:</b> Router(config-register-global)# max-pool 10</p>	<p>Sets the maximum number of SIP voice register pools that are supported in a Cisco SIP SRST environment. The <i>max-voice-register-pools</i> argument represents the maximum number of SIP voice register pools supported by the Cisco 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.</p>
Step 5	<p><b>application</b> <i>application-name</i></p> <p><b>Example:</b> Router(config-register-global)# application global_app</p>	<p>Selects the session-level application for all dial peers associated with SIP phones. Use the <i>application-name</i> argument to define a specific interactive voice response (IVR) application.</p>
Step 6	<p><b>external-ring</b> {<b>bellcore-dr1</b>   <b>bellcore-dr2</b>   <b>bellcore-dr3</b>   <b>bellcore-dr4</b>   <b>bellcore-dr5</b>}</p> <p><b>Example:</b> Router(config-register-global)# external-ring bellcore-dr1</p>	<p>Specifies the type of ring sound used on Cisco SIP or Cisco SCCP IP phones for external calls. Each <b>bellcore-dr 1-5</b> keyword supports standard distinctive ringing patterns as defined in the standard GR-506-CORE, <i>LSSGR: Signaling for Analog Interfaces</i>.</p>
Step 7	<p><b>exit</b></p> <p><b>Example:</b> Router(config-register-global)# exit</p>	<p>Exits voice register global configuration mode.</p>
Step 8	<p><b>voice register pool</b> <i>tag</i></p> <p><b>Example:</b> Router(config)# voice register pool 20</p>	<p>Enters voice register pool configuration mode for SIP phones.</p> <ul style="list-style-type: none"> <li>Use this command to control which phone registrations are to be accepted or rejected by a Cisco SIP SRST device.</li> </ul>
Step 9	<p><b>no vad</b></p> <p><b>Example:</b> Router(config-register-pool)# no vad</p>	<p>Disables voice activity detection (VAD) on the VoIP dial peer.</p> <ul style="list-style-type: none"> <li>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 <b>no vad</b> on the SIP phone pool.</li> </ul>

	Command or Action	Purpose
Step 10	<pre>codec codec-type [bytes]</pre> <p><b>Example:</b> Router(config-register-pool)# codec g729r8</p>	<p>Specifies the codec supported by a single SIP phone or a VoIP dial peer in a Cisco SIP SRST environment. The <i>codec-type</i> argument specifies the preferred codec and can be one of the following:</p> <ul style="list-style-type: none"> <li>• <b>g711alaw</b>—G.711 a-law 64,000 bps.</li> <li>• <b>g711ulaw</b>—G.711 mu-law 64,000 bps.</li> <li>• <b>g729r8</b>—G.729 8000 bps (default).</li> </ul> <p>The <i>bytes</i> argument is optional and specifies the number of bytes in the voice payload of each frame</p>
Step 11	<pre>end</pre> <p><b>Example:</b> Router(config-register-pool)# end</p>	<p>Returns to privileged EXEC mode.</p>

## Configuring SIP-to-SIP Call Forwarding

With Cisco SIP SRST Version 3.4, 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 20. 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 SIP SRST setup, the **call-forward b2bua mailbox** command is not used; however it is likely to be used in a Cisco 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	<b>enable</b>  <b>Example:</b> Router> enable	Enables privileged EXEC mode. <ul style="list-style-type: none"> <li>• Enter your password if prompted.</li> </ul>
Step 2	<b>configure terminal</b>  <b>Example:</b> Router# configure terminal	Enters global configuration mode.
Step 3	<b>voice register pool</b> <i>tag</i>  <b>Example:</b> Router(config)# voice register pool 15	Enters voice register pool configuration mode. <ul style="list-style-type: none"> <li>• Use this command to control which phone registrations are accepted or rejected by a Cisco SIP SRST device.</li> </ul>
Step 4	<b>call-forward b2bua all</b> <i>directory-number</i>  <b>Example:</b> 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: <ul style="list-style-type: none"> <li>• <i>directory-number</i>—Telephone number to which calls are forwarded. Represents a fully qualified E.164 number. Maximum length of the telephone number is 32.</li> </ul>
Step 5	<b>call-forward b2bua busy</b> <i>directory-number</i>  <b>Example:</b> 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. <ul style="list-style-type: none"> <li>• <i>directory-number</i>—Telephone number to which calls are forwarded. Represents a fully qualified E.164 number. Maximum length of the telephone number is 32.</li> </ul>

	Command or Action	Purpose
Step 6	<pre>call-forward b2bua mailbox directory-number</pre> <p><b>Example:</b> Router(config-register-pool)# call-forward b2bua mailbox 5007</p>	<p>Controls the specific voice-mail box selected in a voice-mail system at the end of a call forwarding exchange.</p> <ul style="list-style-type: none"> <li><i>directory-number</i>—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.</li> </ul>
Step 7	<pre>call-forward b2bua noan directory-number timeout seconds</pre> <p><b>Example:</b> Router(config-register-pool)# call-forward b2bua noan 5010 timeout 10</p>	<p>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.</p> <p>This command is used if a phone is registered with a Cisco SIP SRST router, but the phone is not reachable because there is no IP connectivity (there is no response to Invite requests).</p> <ul style="list-style-type: none"> <li><i>directory-number</i>—Telephone number to which calls are forwarded. Represents a fully qualified E.164 number. Maximum length of the telephone number is 32.</li> <li><b>timeout seconds</b>—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.</li> </ul>
Step 8	<pre>end</pre> <p><b>Example:</b> Router(config-register-pool)# end</p>	<p>Returns to privileged EXEC mode.</p>

## 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 SIP SRST 3.4 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 SIP SRST.

The commands used for SIP phone call blocking are the same commands that are used for SCCP phones on your Cisco 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 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	<b>enable</b>  <b>Example:</b> Router> enable	Enables privileged EXEC mode. <ul style="list-style-type: none"> <li>• Enter your password if prompted.</li> </ul>
Step 2	<b>configure terminal</b>  <b>Example:</b> Router# configure terminal	Enters global configuration mode.
Step 3	<b>call-manager-fallback</b>  <b>Example:</b> Router(config)# call-manager-fallback	Enters call-manager-fallback configuration mode.
Step 4	<b>after-hours block pattern</b> <i>tag pattern</i> [7-24]  <b>Example:</b> 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. <ul style="list-style-type: none"> <li>• If the <b>7-24</b> keyword is specified, the pattern is always blocked, 7 days a week, 24 hours a day.</li> <li>• If the <b>7-24</b> keyword is not specified, the pattern is blocked during the days and dates that are defined using the <b>after-hours day</b> and <b>after-hours date</b> commands.</li> </ul>

Command or Action	Purpose
<p><b>Step 5</b>    <code>after-hours day day start-time stop-time</code></p> <p><b>Example:</b>                      Router(config-cm-fallback)# after-hours day mon                      19:00 07:00</p>	<p>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 <b>after-hours block pattern</b> command.</p> <ul style="list-style-type: none"> <li>• <i>day</i>—Day of the week abbreviation. The following are valid day abbreviations: <b>sun, mon, tue, wed, thu, fri, sat</b>.</li> <li>• <i>start-time stop-time</i>—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.”</li> </ul> <p>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.</p>
<p><b>Step 6</b>    <code>after-hours date month date start-time stop-time</code></p> <p><b>Example:</b>                      Router(config-cm-fallback)# after-hours date                      jan 1 00:00 00:00</p>	<p>Defines a recurring time period based on month and date during which calls are blocked to outgoing dial patterns that are defined using the <b>after-hours block pattern</b> command.</p> <ul style="list-style-type: none"> <li>• <i>month</i>—Month abbreviation. The following are valid month abbreviations: <b>jan, feb, mar, apr, may, jun, jul, aug, sep, oct, nov, dec</b>.</li> <li>• <i>date</i>—Date of the month. Range is from 1 to 31.</li> <li>• <i>start-time stop-time</i>—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.</li> </ul> <p>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.</p>
<p><b>Step 7</b>    <code>exit</code></p> <p><b>Example:</b>                      Router(config-cm-fallback)# exit</p>	<p>Exits call-manager-fallback configuration mode.</p>
<p><b>Step 8</b>    <code>voice register pool tag</code></p> <p><b>Example:</b>                      Router(config)# voice register pool 12</p>	<p>Enters voice register pool configuration mode.</p> <ul style="list-style-type: none"> <li>• Use this command to control which registrations are accepted or rejected by a Cisco SIP SRST device.</li> </ul>

	Command or Action	Purpose
Step 9	<code>after-hour exempt</code>  <b>Example:</b> Router(config-register-pool)# <code>after-hour exempt</code>	Specifies that for a particular voice register pool, none its outgoing calls are blocked even though call blocking is enabled.
Step 10	<code>end</code>  <b>Example:</b> Router(config-register-pool)# <code>end</code>	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 SRST 3.4 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 SIP SRST Version 3.4 Features Using Back-to-Back User Agent Mode

This section provides the following configuration example.

- [Cisco SIP SRST Version 3.4: Example](#)

**Note**

IP addresses and hostnames in examples are fictitious.

### Cisco SIP SRST Version 3.4: 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
```





---

## A

- after-hour exempt command [58](#)
- after-hours block pattern command [56](#)
- after-hours date command [57](#)
- after-hours day command [57](#)
- alias command [30](#)
- allow-connections command [20](#)
- application command [27, 51](#)

---

## C

- call blocking configuration [55](#)
- call-forward b2bua all command [54](#)
- call-forward b2bua busy command [54](#)
- call-forward b2bua mailbox command [54](#)
- call-forward b2bua noan command [54](#)
- Cisco CallManager
  - versions supported by Cisco SRST [8](#)
- Cisco IP phones
  - supported by each SRST version [8](#)
- codec command [51](#)
- configuration
  - call blocking [55](#)
  - call forwarding [53](#)
  - Cisco SRST, order of tasks [11](#)
  - codecs [53](#)
  - ring sound [52](#)
  - SIP proxy [29](#)
  - SIP registrar [24](#)
  - SIP-to-SIP connection capabilities [20](#)
  - translation rules [31](#)
  - voice activity detection (VAD) [52](#)

- cor command [30](#)

---

## D

- description of SIP SRST [5](#)
- documentation references [11](#)
- dtmf-relay command [30](#)

---

## E

- external ring command [51](#)

---

## F

- feature roadmap [1](#)

---

## H

- hairpin call routing
  - enabling SIP-to-SIP connections [20](#)

---

## I

- incoming called-number command [30](#)

---

## M

- max-pool command [51](#)
- max registrations command [30](#)
- MIBs (Management Information Bases)
  - supported by Cisco SRST [12](#)

---

**N**

number command [30](#)

---

**P**

platforms

supported by each SRST version [8](#)

preference command [27](#)

prerequisites for configuring Cisco SIP SRST [8](#)

proxy command [27](#)

---

**R**

redirect contact order command [44](#)

redirect ip2ip command [18, 41](#)

registrar server command

SIP networks [25](#)

restrictions

for each Cisco SRST version [10](#)

RFCs

supported by Cisco SIP SRST [12](#)

---

**S**

show dial-peer voice command [35](#)

show voice register dial-peers command [35](#)

SIP SRST description [5](#)

standards

supported by Cisco SRST [12](#)

---

**T**

translate-outgoing command [30](#)

---

**V**

vad command [51](#)

voice-class codec command [27](#)

voice register global command [51](#)

voice register pool command [27, 30, 51, 54, 56](#)

VoIP-to-VoIP connections

configuring [20](#)