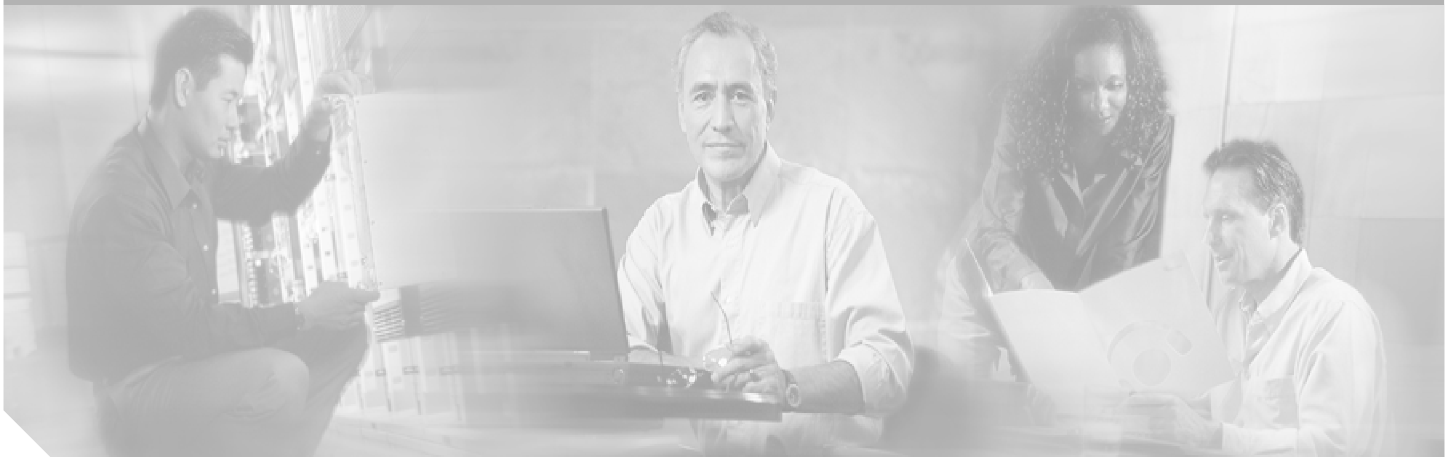

Cisco IOS Survivable Remote Site Telephony Version 3.4 System Administrator Guide

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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

Cisco IOS Survivable Remote Site Telephony Version 3.4 System Administrator Guide

CONTENTS

Cisco IOS Survivable Remote Site Telephony Fea ture Roadmap

This chapter contains a list of Cisco IOS SRST features and the location of feature documentation.

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

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

Contents

Documentation Organization

This document consists of the following chapters or appendixes as shown in Table 1.

Table 1 Cisco SRST Configuration Sequence

Chapter or Appendix	Description
Overview of Cisco IOS SRST	Provides a summary of SRST. This chapter includes the following sections: • Cisco SRST Description, page 19 • Support for Cisco IP Phones, Platforms, Cisco CallManager, Signals, Languages, and Switches, page 23 • Prerequisites for Configuring Cisco SRST, page 26 • Restrictions for Configuring Cisco SRST, page 28 • Additional References, page 30
Setting Up the Network	Describes how to set up a Cisco SRST system to communicate with your network. This chapter includes the following tasks: • Enabling IP Routing, page 34 • Configuring DHCP for Cisco SRST Phones, page 36 • Specifying Keepalive Intervals, page 39 • Configuring Cisco SRST to Support Phone Functions, page 40 • Verifying That Cisco SRST Is Enabled, page 42
Setting Up Cisco IP Phones	Describes how to set up the basic Cisco SRST phone configuration. This chapter includes the following tasks: • Configuring IP Phone Clock, Date, and Time Formats, page 46 • Configuring IP Phone Language Display, page 47 • Configuring Customized System Messages for Cisco IP Phones, page 48 • Configuring a Secondary Dial Tone, page 50 • Configuring Dual-Line Phones, page 51
Setting Up Call Handling	Describes how to configure incoming and outgoing calls. This chapter includes the following tasks: • Configuring Incoming Calls, page 56 • Configuring Outgoing Calls, page 73
Configuring Additional Call Features	Describes how to configure optional system and phone parameters. This chapter includes the following tasks: • Enabling Three-Party G.711 Ad Hoc Conferencing, page 92 • Configuring MOH for G.711 VoIP and PSTN Calls, page 93 • Configuring MOH from Flash Files, page 94

Table 1 Cisco SRST Configuration Sequence (continued)

Chapter or Appendix	Description
Setting Up Secure SRST	Describes the Media and Signaling Authentication and Encryption feature for Cisco IOS MGCP gateways in SRST mode. This chapter includes the following tasks: • Preparing the SRST Router for Secure Communication, page 105 • Importing Phone Certificate Files in PEM Format to the Secure SRST Router, page 114 • Configuring Cisco CallManager to the Secure SRST Router, page 118 • Enabling SRST Mode on the Secure SRST Router, page 121 • Verifying Phone Status and Registrations, page 123
Integrating Voice Mail with Cisco SRST	Describes how to set up voice mail. This chapter includes the following tasks: • Configuring Direct Access to Voice Mail, page 137 • Configuring Message Buttons, page 140 • Redirecting to Cisco CallManager Gateway, page 142 • Configuring Call Forwarding to Voice Mail, page 142
Monitoring and Maintaining Cisco SRST	Provides a list of useful **show** commands for monitoring and maintaining SRST.
Appendix A: Preparing Cisco SRST Support for SIP	Describes special configurations to support SIP calls.

Feature Roadmap

Table 2 provides a feature history summary of Cisco IOS SRST features.

Table 2 Cisco IOS SRST Features by Cisco IOS Release

Cisco SRST Version	Cisco IOS Release	Modifications
Version 3.4	12.4(4)T	• SIP SRST, Version 3.4, page 6
Version 3.3	12.3(14)T	• Secure SRST, page 7. • Cisco IP Phone 7970G and Cisco 7971G-GE Support, page 7 • Enhancement to the show ephone Command, page 8

Table 2 Cisco IOS SRST Features by Cisco IOS Release (continued)

Cisco SRST Version	Cisco IOS Release	Modifications
Version 3.2	12.3(11)T	• Enhancement to the alias Command, page 8 • Enhancement to the pickup Command, page 8 • Enhancement to the user-locale Command, page 9 • Enhancement to the user-locale Command, page 9 • Increased the Number of Cisco IP Phones Supported on the Cisco 3845, page 9 • MOH Live-Feed Support, page 9 • No Timeout for Call Preservation, page 9 • RFC 2833 DTMF Relay Support, page 9 • Translation Profile Support, page 9
Version 3.1	12.3(7)T	• Cisco IP Phone 7920 Support, page 10 • Cisco IP Phone 7936 Support, page 10
Version 3.0	12.3(4)T	—
	12.2(15)ZJ	• Additional Language Options for IP Phone Display, page 11 • Consultative Call Transfer and Forward Using H.450.2 and H.450.3, page 11 • Customized System Message for Cisco IP Phones, page 12 • Dual-Line Mode, page 12 • E1 R2 Signaling Support, page 12 • European Date Formats, page 13 • Huntstop for Dual-Line Mode, page 13 • Music on Hold for Multicast from Flash Files, page 13 • Ringing Timeout Default, page 14 • Secondary Dial Tone, page 14 • Enhancement to the show ephone Command, page 14 • System Log Messages for Phone Registrations, page 14 • Three-Party G.711 Ad Hoc Conferencing, page 14 • Support for Cisco VG248 Analog Phone Gateway Version 1.2(1) and Higher, page 14

Table 2 Cisco IOS SRST Features by Cisco IOS Release (continued)

Cisco SRST Version	Cisco IOS Release	Modifications
Version 2.1	12.2(15)T1	• Cisco IP Phone 7902G Support, page 16 • Cisco IP Phone 7912G Support, page 16
	12.2(15)T	—
	12.2(11)YT	• Additional Language Options for IP Phone Display, page 15 • Cisco SRST Aggregation, page 15 • Cisco ATA 186 and ATA 188 Support, page 16 • Cisco IP Phone 7905G Support, page 16 • Cisco IP Phone Expansion Module 7914 Support, page 17 • Enhancement to the dialplan-pattern Command, page 17
Version 2.02	12.2(13)T	• Cisco IP Phone Conference Station 7935 Support, page 17. • Increase in Directory Numbers, page 17. • Unity Voice Mail Integration Using In-Band DTMF Signaling Across the PSTN and BRI/PRI, page 18. • Cisco SRST was implemented on the Cisco Catalyst 4500 access gateway module and Cisco 7200 routers (NPE-225, NPE-300, and NPE400). • Support was removed for the Cisco MC3810-V3 concentrator.
Version 2.01	12.2(11)T	• Cisco SRST was implemented on the Cisco 1760 routers, and support for the Cisco 1750 was removed. • Support was added for additional connected Cisco IP phones. • Support was added for additional directory numbers or virtual voice ports on Cisco IP phones.
Version 2.0	12.2(8)T1	Cisco SRST was implemented on the Cisco 2600XM and Cisco 2691 routers.
	12.2(8)T	Cisco SRST was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745 routers and the Cisco MC3810-V3 concentrators.
	12.2(2)XT	• Cisco SRST was implemented on the Cisco 1750 and Cisco 1751 routers. • Huntstop support. • Class of restriction (COR). • Translation rule support. • Music on hold and tone on hold. • Distinctive ringing. • Forward to a central voice mail or auto-attendant (AA) through PSTN during Cisco CallManager fallback. • Phone number alias support during Cisco CallManager fallback: enhanced default destination support. • List-based call restrictions for Cisco CallManager fallback.

Table 2 Cisco IOS SRST Features by Cisco IOS Release (continued)

Cisco SRST Version	Cisco IOS Release	Modifications
Version 1.0	12.1(5)YD1	Support was added for 144 Cisco IP phones on the Cisco 3660 multiservice routers.
	12.1(5)YD	• Cisco SRST introduced on the Cisco 2600 series and Cisco 3600 series multiservice routers and the Cisco IAD2420 series integrated access devices. • Cisco IP phones able to establish a connection with an SRST router in the event of a WAN link to Cisco CallManager failure. • Dimming of all Cisco IP phone function keys that are not supported during Cisco SRST operation. • Extension-to-extension dialing. • Direct Inward Dialing (DID). • Direct Outward Dialing (DOD). • Calling party ID (Caller ID/ANI) display. • Last number redial. • Preservation of local extension-to-extension calls when WAN link fails. • Preservation of local extension to PSTN calls when WAN link fails. • Preservation of calls in progress when failed WAN link is reestablished. • Blind transfer of calls within IP network. • Multiple lines per Cisco IP phone. • Multiple-line appearance across telephones. • Call hold (shared lines). • Analog Foreign Exchange Station (FXS) and Foreign Exchange Office (FXO) ports. • BRI support for EuroISDN. • PRI support for NET5 switch type.

Information About New Features in Cisco SRST V3.4

Cisco SRST V3.4 introduced the new features described in the following section:

- SIP SRST, Version 3.4

SIP SRST, Version 3.4

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

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

Information About New Features in Cisco SRST V3.3

Cisco SRST V3.3 introduced the new features described in the following sections:

- Secure SRST
- Cisco IP Phone 7970G and Cisco 7971G-GE Support
- Enhancement to the show ephone Command

Secure SRST

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

Secure SRST provides new SRST security features such as authentication, integrity, and media encryption. Authentication provides assurance to one party that another party is whom it claims to be. Integrity provides assurance that the given data has not been altered between the entities. Encryption implies confidentiality; that is, that no one can read the data except the intended recipient. These security features allow privacy for SRST voice calls and protect against voice security violations and identity theft. For more information see the chapter "Setting Up Secure SRST" section on page 97.

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

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

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

For more information, see the following URL:

http://www.cisco.com/univercd/cc/td/doc/product/voice/c_ipphon/english/ipp7970/index.htm

Note The Cisco IP Phone 7914 Expansion Module can attach to your Cisco IP phones 7970G and Cisco 7971G-GE. See Cisco IP Phone Expansion Module 7914 Support, page 17 for more information.

Enhancement to the show ephone Command

The **show ephone** command has been enhanced to display the configuration and status of the Cisco 7970G and Cisco 7971G-GE phones. For more information, see the **show ephone** command in the *Cisco IOS Survivable Remote Site Telephony (SRST) Command Reference (All Versions)*.

Information About New Features in Cisco SRST V3.2

Cisco SRST V3.2 introduced the new features described in the following sections:

- Enhancement to the alias Command
- Enhancement to the cor Command
- Enhancement to the pickup Command
- Enhancement to the user-locale Command
- Increased the Number of Cisco IP Phones Supported on the Cisco 3845
- MOH Live-Feed Support
- No Timeout for Call Preservation
- RFC 2833 DTMF Relay Support
- Translation Profile Support

Enhancement to the alias Command

The **alias** command has been enhanced as follows:

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

For more information, see the **alias** command in the *Cisco IOS Survivable Remote Site Telephony (SRST) Command Reference (All Versions).*

Enhancement to the cor Command

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

For more information, see the **cor** command in the *Cisco IOS Survivable Remote Site Telephony (SRST) Command Reference (All Versions)*.

Enhancement to the pickup Command

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

For more information, see the **pickup** command in the *Cisco IOS Survivable Remote Site Telephony (SRST) Command Reference (All Versions)*.

Enhancement to the user-locale Command

The**user-locale** command has been enhanced to display the Japanese Katakana country code. Japanese Katakana is available under Cisco CallManager V4.0 or later.

For more information, see the **user-locale** command in the *Cisco IOS Survivable Remote Site Telephony (SRST) Command Reference (All Versions)*.

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

The Cisco 3845 now supports 720 phones and up to 960 ephone-dns or virtual voice ports. For more information, see *Cisco IOS Survivable Remote Site Telephony (SRST) 3.2 Specifications for Cisco IOS Software Release 12.3(11)T.*

MOH Live-Feed Support

Cisco SRST has been enhanced with the new **moh-live** command. The **moh-live** command provides live-feed MOH streams from an audio device connected to an E&M or FXO port to Cisco IP phones in SRST mode. If an FXO port is used for a live feed, the port must be supplied with an external third-party adapter to provide a battery feed. Music from a live feed is obtained from a fixed source and is continuously fed into the MOH playout buffer instead of being read from a flash file. Live-feed MOH can also be multicast to Cisco IP phones. See Configuring SRST MOH Live-Feed Support for configuration instructions.

No Timeout for Call Preservation

To preserve existing H.323 calls on the branch in the event of an outage, disable the H.225 keepalive timer by entering the **no h225 timeout keepalive** command. This feature is supported in Cisco IOS Releases 12.3(7)T1 and higher. See the "Cisco SRST Description" section on page 19 for more information.

RFC 2833 DTMF Relay Support

Cisco Skinny Client Control Protocol (SCCP) phones, such as those used with Cisco SRST systems, provide only out-of-band DTMF digit indications. To enable SCCP phones to send digit information to remote SIP-based IVR and voice-mail applications, Cisco SRST 3.2 and later versions provide conversion from the out-of-band SCCP digit indication to the SIP standard for DTMF relay, which is RFC 2833. You select this method in the SIP VoIP dial peer using the **dtmf-relay rtp-nte** command. See Appendix A: Preparing Cisco SRST Support for SIP, page 155 for configuration instructions.

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

Translation Profile Support

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

- Called numbers
- Calling numbers

- Redirected called numbers

See the “Enabling Translation Profiles” section on page 66 for more configuration information. For more information on the**translation-profile**, command see the *Cisco IOS Survivable Remote Site Telephony (SRST) Command Reference (All Versions)*.

Information About New Features in Cisco SRST V3.1

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

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

Cisco IP Phone 7920 Support

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

No configuration is necessary.

For more information, see the following URL:

http://www.cisco.com/univercd/cc/td/doc/product/voice/c_ipphon/english/wip7920/

Cisco IP Phone 7936 Support

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

No configuration is necessary.

For more information, see the following URL:

http://www.cisco.com/univercd/cc/td/doc/product/voice/c_ipphon/english/ipp7936/

Information About New Features in Cisco SRST V3.0

Cisco SRST V3.0 introduced the new features described in the following sections:

- Additional Language Options for IP Phone Display
- Consultative Call Transfer and Forward Using H.450.2 and H.450.3
- Customized System Message for Cisco IP Phones
- Dual-Line Mode
- E1 R2 Signaling Support
- European Date Formats
- Huntstop for Dual-Line Mode

- Music on Hold for Multicast from Flash Files
- Ringing Timeout Default
- Secondary Dial Tone
- Enhancement to the show ephone Command
- System Log Messages for Phone Registrations
- Three-Party G.711 Ad Hoc Conferencing
- Support for Cisco VG248 Analog Phone Gateway Version 1.2(1) and Higher

Additional Language Options for IP Phone Display

Displays for the Cisco IP Phone 7940G and Cisco IP Phone 7960G can be configured with additional ISO-3166 codes for Denmark, The Netherlands, Norway, and Sweden.

Note This feature is available only for Cisco SRST running under Cisco CallManager V3.2.

Consultative Call Transfer and Forward Using H.450.2 and H.450.3

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

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

For consultative transfer to be available, the Cisco SRST router must be configured with the dual-line mode. See the "Configuring Dual-Line Phones" section on page 51.

Note All voice gateway routers in the VoIP network must support H.450. For H.450 support, routers with Cisco SRST must run either Cisco SRST V3.0 and higher versions or Cisco IOS Release 12.2(15)ZJ and later releases. Routers without Cisco SRST must run either Cisco SRST V2.1 and higher versions or Cisco IOS Release 12.2(11)YT and later releases.

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

For configuration information, see the "Enabling Consultative Call Transfer and Forward Using H.450.2 and H.450.3 with Cisco SRST V3.0" section on page 74.

Customized System Message for Cisco IP Phones

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

For further information, see the "Configuring Customized System Messages for Cisco IP Phones" section on page 48.

Dual-Line Mode

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

For configuration information, see the "Configuring Dual-Line Phones" section on page 51.

E1 R2 Signaling Support

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

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

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

- Argentina
- Australia
- Bolivia
- Brazil
- Bulgaria
- China
- Colombia
- Costa Rica
- East Europe (includes Croatia, Russia, and Slovak Republic)
- Ecuador (ITU)
- Ecuador (LME)
- Greece
- Guatemala

- Hong Kong (uses the China variant)
- Indonesia
- Israel
- Korea
- Laos
- Malaysia
- Malta
- New Zealand
- Paraguay
- Peru
- Philippines
- Saudi Arabia
- Singapore
- South Africa (Panaftel variant)
- Telmex corporation (Mexico)
- Telnor corporation (Mexico)
- Thailand
- Uruguay
- Venezuela
- Vietnam

European Date Formats

The date format on Cisco IP phone displays can be configured with the following two additional formats:

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

For configuration information, see the "Configuring IP Phone Clock, Date, and Time Formats" section on page 46.

Huntstop for Dual-Line Mode

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

For configuration information, see the "Configuring Dial-Peer and Channel Hunting" section on page 70.

Music on Hold for Multicast from Flash Files

Cisco SRST can be configured to support continuous multicast output of music on hold (MOH) from a flash MOH file in flash memory.

For more information, see the "Configuring MOH from Flash Files" section on page 94.

Ringing Timeout Default

A ringing timeout default can be configured for extensions on which no-answer call forwarding has not been enabled. Expiration of the timeout causes incoming calls to return a disconnect code to the caller. This mechanism provides protection against hung calls for inbound calls received over interfaces such as Foreign Exchange Office (FXO) that do not have forward-disconnect supervision. For more information, see the "Configuring the Ringing Timeout Default" section on page 72.

Secondary Dial Tone

A secondary dial tone is available for Cisco IP phones running Cisco SRST. The secondary dial tone is generated when a user dials a predefined PSTN access prefix. An example would be the different dial tone heard when a designated number is pressed to reach an outside line.

The secondary dial tone is created through the secondary dialtone command. For more information, see the "Configuring a Secondary Dial Tone" section on page 50.

Enhancement to the show ephone Command

The **show ephone** command has been enhanced to display the following:

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

For more information, see the **show ephone** command in the *Cisco IOS Survivable Remote Site Telephony (SRST) Command Reference (All Versions)*.

System Log Messages for Phone Registrations

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

Three-Party G.711 Ad Hoc Conferencing

Cisco SRST supports three-party ad hoc conferencing using the G.711 coding technique. For conferencing to be available, an IP phone must have a minimum of two lines connected to one or more buttons.

For more information, see the "Enabling Three-Party G.711 Ad Hoc Conferencing" section on page 92.

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

The Cisco VG248 Analog Phone Gateway is a mixed-environment solution, enabled by Cisco AVVID (Architecture for Voice, Video and Integrated Data), that allows organizations to support their legacy analog devices while taking advantage of the new opportunities afforded through the use of IP telephony. The Cisco VG248 is a high-density gateway for using analog phones, fax machines, modems, voice-mail systems, and speakerphones within an enterprise voice system based on Cisco CallManager.

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

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

Information About Features That Were New in Cisco SRST V2.1

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

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

Additional Language Options for IP Phone Display

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

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

Note This feature is available only in Cisco SRST running under Cisco CallManager V3.2.

For configuration information, see the "Configuring IP Phone Language Display" section on page 47.

Cisco SRST Aggregation

For systems running Cisco CallManager 3.3(2) and later, the restriction of running Cisco SRST on a default gateway was removed. Multiple SRST routers can be used to support additional phones. Note that dial peers and dial plans need to be carefully planned and configured in order for call transfer and forwarding to work properly.

Cisco ATA 186 and ATA 188 Support

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

Cisco IP Phone 7902G Support

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

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

For further information, go to Cisco.com and click **Products & Solutions > Voice & IP Communications > 7900 Series IP Phones > Product Literature > Data Sheets** or go to http://www.cisco.com/univercd/cc/td/doc/product/voice/c_ipphon/english/ipp7902/index.htm.

Cisco IP Phone 7905G Support

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

No configuration is necessary.

For more information, see the following URL:

http://www.cisco.com/univercd/cc/td/doc/product/voice/c_ipphon/7905_g/index.htm

Cisco IP Phone 7912G Support

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

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

For further information, go to Cisco.com and click **Products & Solutions > Voice & IP Communications > 7900 Series IP Phones > Product Literature > Data Sheets**.

Cisco IP Phone Expansion Module 7914 Support

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

No configuration is necessary.

For more information, see the *Cisco IP Phone 7914 Expansion Module Quick Start Guide*.

Enhancement to the dialplan-pattern Command

A new keyword has been added to the **dialplan-pattern** command. The **extension-pattern** keyword sets an extension number's leading digit pattern when it is different from the E.164 telephone number's leading digits defined in the *pattern* variable. This enhancement allows manipulation of IP phone abbreviated extension number prefix digits. See the **dialplan-pattern** command in the *Cisco IOS Survivable Remote Site Telephony (SRST) Command Reference (All Versions)*.

Information About Features That Were New in Cisco SRST V2.02

Cisco SRST Version 2.02 introduced the new features described in the following sections:

- Cisco IP Phone Conference Station 7935 Support
- Increase in Directory Numbers
- Unity Voice Mail Integration Using In-Band DTMF Signaling Across the PSTN and BRI/PRI

Cisco IP Phone Conference Station 7935 Support

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

The Cisco 7935 provides three soft keys and menu navigation keys that guide a user through call features and functions. The Cisco 7935 also features a pixel-based LCD display. The display provides features such as date and time, calling party name, calling party number, digits dialed, and feature and line status.

No configuration is necessary.

Increase in Directory Numbers

Directory numbers were increased for the platforms shown in Table 3.

Table 3 Increases in Directory Numbers in Cisco IOS Release 12.2(11)T

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

Unity Voice Mail Integration Using In-Band DTMF Signaling Across the PSTN and BRI/PRI

Unity Voice Mail and other voice-mail systems can be integrated with Cisco SRST. Voice-mail integration introduces six new commands:

- pattern direct
- pattern ext-to-ext busy
- pattern ext-to-ext no-answer
- pattern trunk-to-ext busy
- pattern trunk-to-ext no-answer
- vm-integration

For further information, see the *Cisco IOS Survivable Remote Site Telephony (SRST) Command Reference (All Versions)* and the “Integrating Voice Mail with Cisco SRST” section on page 135.

Overview of Cisco IOS SRST

This chapter describes Cisco Survivable Remote Site Telephony (SRST) and what it does. It also includes information about Cisco IP phone, platform, and Cisco CallManager version support; specifications; features; 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, maximum DNs or virtual voice ports, and memory requirements for Cisco 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 SRST Description

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

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

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

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

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

Note The time it takes for an IP phone to fallback to the SRST router can vary depending on the phone type. Phones such as the Cisco 7902, Cisco 7905, and Cisco 7912 can take approximately 2.5 minutes to fallback to SRST mode.

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

Note During a WAN connection failure, when Cisco SRST is enabled, Cisco IP phones display a message informing you that they are operating in Cisco CallManager fallback mode. The Cisco IP Phone 7960G and Cisco IP Phone 7940G display a "CM Fallback Service Operating" message, and the Cisco IP Phone 7910 displays a "CM Fallback Service" message when operating in Cisco CallManager fallback mode. When the Cisco CallManager is restored, the message goes away and full Cisco IP phone functionality is restored.

While in Cisco CallManager fallback mode, Cisco IP phones periodically attempt to reestablish a connection with Cisco CallManager at the central office. Generally the default time that Cisco IP phones wait before attempting to reestablish a connection to a remote Cisco CallManager is 120 seconds. The time can be changed in Cisco CallManager; see the "Device Pool Configuration Settings" chapter in the *Cisco CallManager Administration Guide*. A manual reboot can immediately reconnect Cisco IP phones to Cisco CallManager.

Once a connection is reestablished with Cisco CallManager, Cisco IP phones automatically cancel their registration with the Cisco SRST router. However, if a WAN link is unstable, Cisco IP phones can bounce between Cisco CallManager and Cisco SRST. A Cisco IP phone cannot reestablish a connection with the primary Cisco CallManager at the central office if it is currently engaged in an active call.

Figure 1 shows a branch office with several Cisco IP phones connected to a Cisco SRST router. The router provides connections to both a WAN link and the PSTN. The Cisco IP phones connect to their primary Cisco CallManager at the central office via this WAN link.

Figure 1 Branch Office Cisco IP Phones Connected to a Remote Central Cisco CallManager

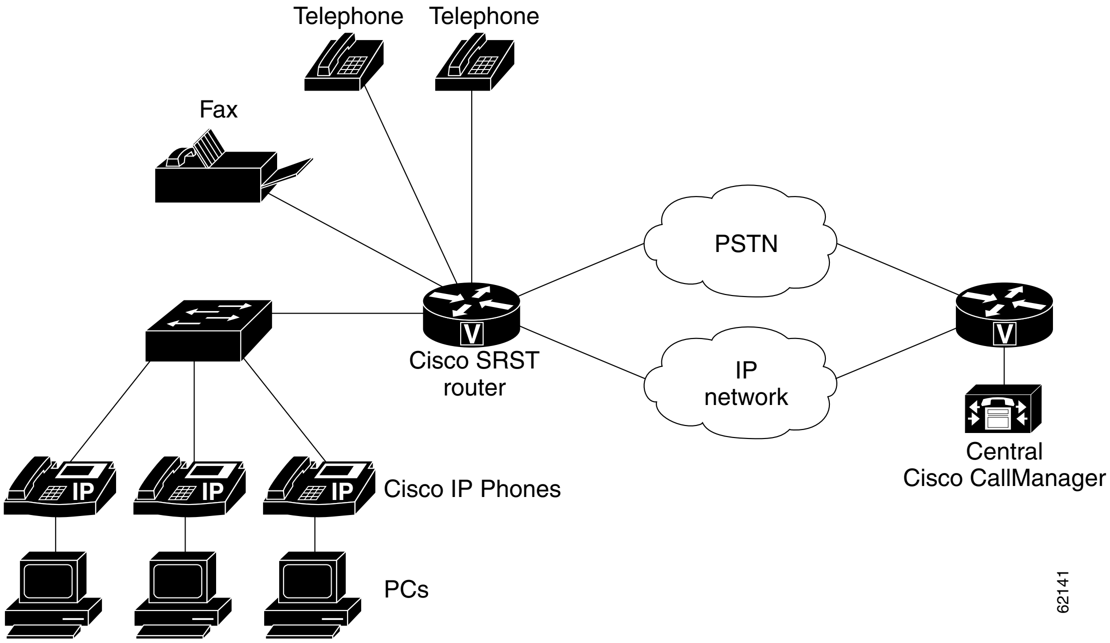
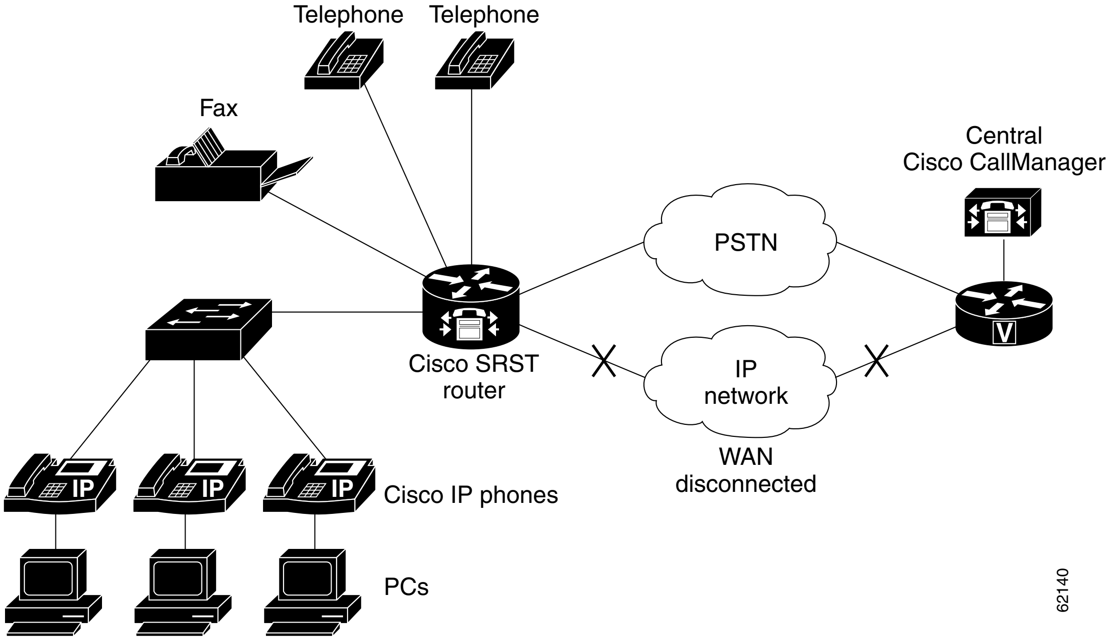


Figure 2 shows the same branch office telephone network with the WAN connection down. In this situation, the Cisco IP phones use the Cisco SRST router as a fallback for their primary Cisco CallManager. The branch office Cisco IP phones are connected to the PSTN through the Cisco SRST router and are able to make and receive off-net calls.

Figure 2 Branch Office Cisco IP Phones Operating in SRST Mode



H.323 Gateways and SRST

On H.323 gateways, when the WAN link fails, active calls from Cisco IP phones to the PSTN are not maintained by default. Call preservation may work with the **no h225 timeout keepalive** command, but call preservation using the **no h225 timeout keepalive** command is not officially supported by Cisco Technical Support.

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

MGCP Gateways and SRST

MGCP fallback is a different feature than SRST and, when configured as an individual feature, can be used by a PSTN gateway. To use SRST as your fallback mode on an MGCP gateway, SRST and MGCP fallback must both be configured on the same gateway. MGCP and SRST have had the capability to be configured on the same gateway since Cisco IOS Release 12.2(11)T.

To make outbound calls while in SRST mode on your MGCP gateway, two fallback commands must be configured on the MGCP gateway. These two commands allow SRST to assume control over the voice port and over call processing on the MGCP gateway. With Cisco IOS releases prior to 12.3(14)T, the two commands are the **ccm-manager fallback-mgcp** and **call application alternate** commands. With Cisco IOS releases after 12.3(14)T, the **ccm-manager fallback-mgcp** and **service** commands must be configured. A complete configuration for these commands is shown in the section “Enabling SRST on an MGCP Gateway” section on page 34.

Note The commands listed above are ineffective unless both commands are configured. For instance, your configuration will not work if you only configure the **ccm-manager fallback-mgcp** command.

For more information on the fallback methods for MGCP gateways, see the *Configuring MGCP Gateway Support for Cisco CallManager* document or the *MGCP Gateway Fallback Transition to Default H.323 Session Application* document.

Support for Cisco IP Phones, Platforms, Cisco CallManager, Signals, Languages, and Switches

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

- Finding Cisco IOS Software Releases That Support Cisco SRST, page 23
- Cisco IP Phone Support, page 24
- Platform and Memory Support, page 24
- Cisco CallManager Compatibility, page 25
- Signal Support, page 25
- Language Support, page 25
- Switch Support, page 26

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. 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 IOS 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

The following IP phones are supported by Cisco SRST 3.4:

- Cisco Analog Telephone Adaptor (ATA) 186 and Cisco ATA 188 Version 2.16 and higher with Cisco CallManager 3.3 and higher

  Cisco SRST supports Cisco ATA 186 and Cisco ATA 188 using Skinny Client Control Protocol (SCCP) for voice calls only
- Cisco IP Phone 7902G
- Cisco IP Phone 7905G
- Cisco IP Phone 7910
- Cisco IP Phone 7912G
- Cisco IP Phone Expansion Module 7914
- Cisco Wireless IP Phone 7920
- Cisco IP Conference Station 7935
- Cisco IP Conference Station 7936
- Cisco IP Phone 7940 and Cisco IP Phone 7940G
- Cisco IP Phone 7960 and Cisco IP Phone 7960G
- Cisco IP Phone 7970G
- Cisco IP Phone 7971G-GE
- Cisco VG224 Analog Phone Gateway, IOS Version 12.4(2)T with Cisco SRST 3.4 running Cisco IOS Software Releases 12.3(14)T, 12.4 mainline, and later. For configuration information see, the "Enabling Fallback to Cisco Unified SRST" section in *SCCP Controlled Analog (FXS) Ports with Supplementary Features in Cisco IOS Gateways* at http://www.cisco.com/en/US/products/ps6441/products_feature_guide09186a0080483a76.html.
- Cisco VG248 Analog Phone Gateway Version 1.2(1) and higher

Note During Cisco CallManager fallback, Cisco SRST considers the Cisco VG248 to be a group of Cisco IP phones. Cisco SRST counts each of the 48 ports on the Cisco VG248 as a separate Cisco IP phone. Support for Cisco VG248 Version 1.2(1) and higher is available as of Cisco SRST Version 2.1. For more information, see the *Cisco VG248 Analog Phone Gateway Data Sheet* and the *Cisco VG248 Analog Phone Gateway Version 1.2(1) Release Notes*.

Platform and Memory Support

For the most up-to-date information about the maximum number of Cisco IP phones, maximum DNs or virtual voice ports, and memory requirements for Cisco SRST, see the *Cisco IOS 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

Determining Platform Support Through Cisco Feature Navigator

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

Availability of Cisco IOS Software Images

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

Note For the most up-to-date information about Cisco IOS software images, see the *Cisco IOS 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 CallManager Compatibility

See the Cisco Call Manager Compatibility Matrix.

Signal Support

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

Language Support

Cisco SRST version 3.2 supports the following languages:

- Danish
- Dutch
- English
- French
- German
- Italian
- Japanese Katakana (available under Cisco CallManager V4.0 or later).
- Norwegian
- Portuguese
- Russian

- Spanish
- Sweden

Note The Cisco IP Phone 7970G and Cisco IP Phone 7971G-GE support English only.

Switch Support

Cisco SRST version 3.2 supports all PRI and BRI switches, including the following:

- basic-1tr6
- basic-5ess
- basic-dms100
- basic-net3
- basic-ni
- basic-ntt NTT switch type for Japan
- basic-ts013
- primary-4ess Lucent 4ESS switch type for the United States
- primary-5ess Lucent 5ESS switch type for the United States
- primary-dms100 Northern Telecom DMS-100 switch type for the United States
- primary-net5 NET5 switch type for the United Kingdom, Europe, Asia, and Australia
- primary-ni National ISDN switch type for the United States
- primary-ntt NTT switch type for Japan
- primary-qsig QSIG switch type
- primary-ts014 TS014 switch type for Australia (obsolete)

Prerequisites for Configuring Cisco SRST

Before configuring Cisco SRST you must do the following:

- You have an account on Cisco.com to download software.

  To obtain an account on Cisco.com, go to www.cisco.com and click **Register** at the top of the screen.
- You have purchased a Cisco SRST license.

  To purchase a license, go to http://www.cisco.com/cgi-bin/tablebuild.pl/ip-key.
- Choose an appropriate Cisco SRST version. Each SRST version supports a specific set of IP phones, memory requirements, features, and directory numbers (DNs). See the "Platform and Memory Support" section on page 24 and the "Restrictions for Configuring Cisco SRST" section on page 29.
- Choose an appropriate phoneload. SRST only supports certain phoneloads that have been tested with the various Cisco CallManager versions. For the most up-to-date phoneloads, see the *Cisco IOS 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

- If you have Cisco CallManager already installed, verify that your version of Cisco CallManager is compatible with your Cisco SRST release. See the "Cisco CallManager Compatibility" section on page 25.

Installing Cisco CallManager

When installing Cisco CallManager consider the following:

- Follow the installation instructions under the appropriate Cisco CallManager version listed at http://www.cisco.com/univercd/cc/td/doc/product/voice/c_callmg/index.htm.
- Integrate Cisco SRST with Cisco CallManager. Integration is performed from Cisco CallManager. See "Integrating Cisco SRST with Cisco CallManager" section on page 28

Installing Cisco SRST

Cisco SRST versions have different installation instructions:

- Installing Cisco SRST V3.0 or Higher, page 27
- Installing Cisco SRST V2.0 and V2.1, page 27
- Installing Cisco SRST V1.0, page 27

To update Cisco SRST, follow the installation instructions described in this section.

Installing Cisco SRST V3.0 or Higher

Install the Cisco IOS software release image containing the Cisco SRST version that is compatible with your Cisco CallManager version. See the "Cisco CallManager Compatibility" section on page 25. Cisco IOS software can be downloaded from the Cisco Software Center at http://www.cisco.com/public/sw-center/.

Cisco SRST can be configured to support continuous multicast output of music on hold (MOH) from a flash MOH file in flash memory. For more information, see the "Configuring MOH from Flash Files" section on page 94. If you plan use music on hold, go to the Technical Support Software Download site at http://www.cisco.com/cgi-bin/tablebuild.pl/ip-iostsp and copy the music-on-hold.au file to the flash memory on your Cisco SRST router.

Installing Cisco SRST V2.0 and V2.1

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

Installing Cisco SRST V1.0

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

Integrating Cisco SRST with Cisco CallManager

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

If You Have Cisco CallManager V3.3 or Later

If you have Cisco CallManager V3.3 or later, you must create an SRST reference and apply it to a device pool. An SRST reference is the IP address of the Cisco SRST router.

Step 1 Create an SRST reference.

a. From any page in Cisco CallManager, click **System** and **SRST**.

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

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

d. Click **Insert**.

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

a. From any page in Cisco CallManager, click **System** and **Device Pool**.

b. On the Device Pool Configuration page, click on the desired device pool icon.

c. On the Device Pool Configuration page, choose an SRST reference or “Use Default Gateway” from the SRST Reference field’s menu.

If You Have Cisco CallManager Prior to V3.3

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

Step 1 Go to the Cisco CallManager Phone Configuration page.

a. From any page in Cisco CallManager, click **Device** and **Phone**.

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

c. After a list of phones appears, click on the desired device name.

d. The Phone Configuration appears.

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

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

Restrictions for Configuring Cisco SRST

Table 4 provides a history of restrictions from Cisco SRST Version 1.0 to the present version.

Table 4 *History of Restrictions from Cisco SRST V1.0 to the Present Version*

Cisco SRST Version	Cisco IOS Release	Restrictions
Version 3.4	12.4(4)T	• All of the restrictions in Cisco SRST Version 1.0. • Call transfer is supported only on the following: – VoIP H.323, VoFR, and VoATM between Cisco gateways running Cisco IOS Release 12.2(11)T and using the H.323 nonstandard information element – FXO and FXS loop-start (analog) – FXO and FXS ground-start (analog) – Ear and mouth (E&M) (analog) and DID (analog) – T1 channel-associated signaling (CAS) with FXO and FXS ground-start signaling – T1 CAS with E&M signaling – All PRI and BRI switch types • The following Cisco IP phone function keys are dimmed because they are not supported during SRST operation: – MeetMe – GPickUp (group pickup) – Park – Confrn (conference) • Although the Cisco IAD2420 series integrated access devices (IADs) support the Cisco SRST feature, this feature is not recommended as a solution for enterprise branch offices.
Version 3.3	12.3(14)T	
Version 3.2	12.3(11)T	
Version 3.1	12.3(7)T	
Version 3.0	12.2(15)ZJ	
Version 2.1	12.2(15)T	
Version 2.02	12.2(13)T	
Version 2.01	12.2(11)T	
Version 2.0	12.2(8)T1	
Version 2.0	12.2(8)T	
Version 2.0	12.2(2)XT	
Version 1.0	12.2(2)XB	• Does not support first generation Cisco IP phones, such as Cisco IP Phone 30 VIP and Cisco IP Phone 12 SP+. • Does not support other Cisco CallManager applications or services: Cisco IP SoftPhone, Cisco uOne—Voice and Unified Messaging Application, or Cisco IP Contact Center. • Does not support Centralized Automatic Message Accounting (CAMA) trunks on the Cisco 3660 routers. **Note** If you are in one of the states in the United States of America where there is a regulatory requirement for CAMA trunks to interface to 911 emergency services, and you would like to connect more than 48 Cisco IP phones to the Cisco 3660 multiservice routers in your network, contact your local Cisco account team for help in understanding and meeting the CAMA regulatory requirements.
	12.2(2)XG	
	12.1(5)YD	

Where to Go Next

The next chapters of this guide describe how to configure Cisco SRST. As shown in Table 5, each chapter takes you through these 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 is configured correctly for Cisco SRST. For instructions, see the "Prerequisites for Configuring Cisco SRST" section on page 26.

Table 5 Cisco SRST Configuration Sequence

Task	Where Task Is Described
1. Setting up a Cisco SRST system to communicate with your network	"Setting Up the Network" chapter
2. Setting up the basic Cisco SRST phone configuration	"Setting Up Cisco IP Phones" chapter
3. Configuring incoming and outgoing calls	"Setting Up Call Handling" chapter
4. Configuring optional system and phone parameters	"Configuring Additional Call Features" chapter
5. Configuring optional security for SRST	"Setting Up Secure SRST" chapter
6. Setting up voice mail	"Integrating Voice Mail with Cisco SRST" chapter

Additional References

The following sections provide additional references related to Cisco SRST:

- Related Documents, page 31
- Standards, page 31
- MIBs, page 31
- RFCs, page 32
- Technical Assistance, page 32

Related Documents

Related Topic	Documents
SRST Commands	• *Cisco IOS Survivable Remote Site Telephony (SRST) Command Reference (All Versions)*
Cisco IP phones	• *Cisco IP Phone 7902 Quick Start Guide* • *Cisco IP Phone 7902G Quick Start Guide* • *Getting Started with the Cisco IP Phone 7910* • *At a Glance Cisco IP Phone 7912G* • *Cisco IP Phone 7914 Expansion Module Quick Start Guide* • *Cisco IP Conference Station 7935 Documents* • *Phone Guide Cisco IP Phone 7960 and 7940 Series* • *Cisco IP Phone 7960 and 7940 Series User Guide*
Command reference and configuration information for voice and telephony commands	• *Cisco IOS Voice Command Reference* • *Cisco IOS Debug Command Reference*
Configuring SRS Telephony and MGCP Fallback	• *Configuring MGCP Gateway Support for Cisco CallManager* • *MGCP Gateway Fallback Transition to Default H.323 Session Application* • *Configuring SRS Telephony and MGCP Fallback*
Cisco CallManager user documentation	• *Cisco CallManager*
DHCP	• *Cisco IOS DHCP Server*
Media Inactive Call Detection	• *Media Inactive Call Detection*
Standard Preface	• *Cisco IOS Voice Configuration Library Preface*
Standard Glossary	• *Cisco IOS Voice Configuration Library Glossary*

Standards

Standard	Title
No new or modified standards are supported by this feature, and support for existing standards has not been modified by this feature.	—

MIBs

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

RFCs

RFC	Title
No new or modified RFCs are supported by this feature, and support for existing RFCs has not been modified by this feature.	—

Technical Assistance

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

Setting Up the Network

This chapter describes how to configure your Cisco Survivable Remote Site Telephony (SRST) router to run DHCP and to communicate with the IP phones during Cisco CallManager fallback.

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

Information About Setting Up the Network

When the WAN link fails, the Cisco IP phones detect that they are no longer receiving keepalive packets from Cisco CallManager. The Cisco IP phones then register with the router. The Cisco SRST software is automatically activated and builds a local database of all Cisco IP phones attached to it (up to its configured maximum). The IP phones are configured to query the router as a backup call-processing source when the central Cisco CallManager does not acknowledge keepalive packets. The Cisco SRST router now performs call setup and processing, call maintenance, and call termination.

Cisco CallManager uses DHCP to provide Cisco IP phones with the IP address of Cisco CallManager. In a remote branch office, DHCP service is typically provided either by the SRST router itself or through the Cisco SRST router using DHCP relay. Configuring DHCP is one of two main tasks in setting up network communication. The other task is configuring the Cisco SRST router to receive messages from the Cisco IP phones through the specified IP addresses. Keepalive intervals are also set at this time.

How to Set Up the Network

This section contains the following tasks:

- Enabling IP Routing, page 34 (Required)
- Enabling SRST on an MGCP Gateway (Required)
- Configuring DHCP for Cisco SRST Phones, page 36 (Required)
- Specifying Keepalive Intervals, page 39 (Optional)
- Configuring Cisco SRST to Support Phone Functions, page 40 (Required)
- Verifying That Cisco SRST Is Enabled, page 42 (Optional)

Enabling IP Routing

For information about enabling IP routing, see the "Enabling IP Routing" section in the "IP Addressing and Services" chapter of the *Cisco IOS IP Configuration Guide*, Release 12.2.

Enabling SRST on an MGCP Gateway

To use SRST as your fallback mode with an MGCP gateway, SRST and MGCP fallback must both be configured on the same gateway. The configuration below allows SRST to assume control over the voice port and over call processing on the MGCP gateway.

Note The commands described in the configuration below are ineffective unless both commands are configured. For instance, your configuration will not work if you only configure the **ccm-manager fallback-mgcp** command.

Restrictions

Effective with Cisco IOS Release 12.3(14)T, the **call application alternate** command is replaced by the **service** command. The **service** command can be used in all releases after Cisco IOS Release 12.3(14)T. Both commands are reflected in Step 4.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **ccm-manager fallback-mgcp**
4. **call application alternate** [*application-name*]
   or
   service [**alternate** | **default**] *service-name location*
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password when prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**ccm-manager fallback-mgcp** **Example:** `Router(config)# ccm-manager fallback-mgcp`	Enables the gateway fallback feature and allows an MGCP voice gateway to provide call processing services through SRST or other configured applications when Cisco CallManager is unavailable.

Step 4	**call application alternate** [*application-name*]
	or
	service [**alternate** | **default**] *service-name location*
	Example:
	`Router(config)# call application alternate`
	`or`
	`Router(config)# service default`
		The **call application alternate** command specifies that the default voice application takes over if the MGCP application is not available. The *application-name* argument is optional and indicates the name of the specific voice application to use if the application in the dial peer fails. If a specific application name is not entered, the gateway uses the DEFAULT application.
		Or
		The **service** command loads and configures a specific, standalone application on a dial peer. The keywords and arguments are as follows:
		• **alternate**—Optional. Alternate service to use if the service that is configured on the dial peer fails.
		• **default**—Optional. Specifies that the default service (“DEFAULT”) on the dial peer is used if the alternate service fails.
		• *service-name*—Name that identifies the voice application.
		• *location*—Directory and filename of the Tcl script or VoiceXML document in URL format. For example, flash memory (flash:filename), a TFTP (tftp://../filename) or an HTTP server (http://../filename) are valid locations

Step 5	**exit** **Example:** `Router(config)# exit`	Exits global configuration mode and returns to privileged EXEC mode.

Configuring DHCP for Cisco SRST Phones

To perform this task, you must have your network configured with DHCP. For further details about DHCP configuration, see the *Cisco IOS DHCP Server* document and refer to your Cisco CallManager documentation.

When a Cisco IP phone is connected to the Cisco SRST system, it automatically queries for a DHCP server. The DHCP server responds by assigning an IP address to the Cisco IP phone and providing the IP address of the TFTP server through DHCP option 150. Then the phone registers with the Cisco CallManager system server and attempts to get configuration and phone firmware files from the Cisco CallManager TFTP server address provided by the DHCP server.

When setting up your network, configure your DHCP server local to your site. You may use your SRST router to provide DHCP service (recommended). If your DHCP server is across the WAN and there is an extended WAN outage, the DHCP lease times on your Cisco IP phones may expire. This may cause your phones to lose their IP addresses, resulting in a loss of service. Rebooting your phones when there is no DHCP server available after the DHCP lease has expired will not reactivate the phones, because they will be unable to obtain an IP address or other configuration information. Having your DHCP server local to your remote site ensures that the phones can continue to renew their IP address leases in the event of an extended WAN failure.

Choose one of the following tasks to set up DHCP service for your IP phones:

- Defining a Single DHCP IP Address Pool, page 36—Use this method if the Cisco SRST router is a DHCP server and if you can use a single shared address pool for all your DHCP clients.
- Defining a Separate DHCP IP Address Pool for Each Cisco IP Phone, page 37—Use this method if the Cisco SRST router is a DHCP server and you need separate pools for non-IP-phone DHCP clients.
- Defining the DHCP Relay Server, page 38—Use this method if the Cisco SRST router is not a DHCP server and you want to relay DHCP requests from IP phones to a DHCP server on a different router.

Defining a Single DHCP IP Address Pool

This task creates a large shared pool of IP addresses in which all DHCP clients receive the same information, including the option 150 TFTP server IP address. The benefit of selecting this method is that you set up only one DHCP pool. However, defining a single DHCP IP address pool can be a problem if some (non-IP phone) clients need to use a different TFTP server address.

SUMMARY STEPS

1. **ip dhcp pool** *pool-name*
2. **network** *ip-address* [*mask* | *prefix-length*]
3. **option 150 ip** *ip-address*
4. **default-router** *ip-address*
5. **exit**

DETAILED STEPS

	Command or Action	Purpose

Step 6	**`ip dhcp pool`** *`pool-name`*
	Example:
	`Router(config)# ip dhcp pool mypool`
		Creates a name for the DHCP server address pool and enters DHCP pool configuration mode.

Step 7	**`network`** *`ip-address`* `[`*`mask`* `|` *`prefix-length`*`]`
	Example:
	`Router(config-dhcp)# network 10.0.0.0 255.255.0.0`
		Specifies the IP address of the DHCP address pool and the optional mask or number of bits in the address prefix, preceded by a forward slash.

Step 8	**`option 150 ip`** *`ip-address`*
	Example:
	`Router(config-dhcp)# option 150 ip 10.0.22.1`
		Specifies the TFTP server address from which the Cisco IP phone downloads the image configuration file. This needs to be the IP address of CallManager.

Step 9	**`default-router`** *`ip-address`*
	Example:
	`Router(config-dhcp)# default-router 10.0.0.1`
		Specifies the router to which the Cisco IP phones are connected directly.
		• This router should be the Cisco SRST router because this is the default address that is used to obtain SRST service in the event of a WAN outage. As long as the Cisco IP phones have a connection to the Cisco SRST router, the phones are able to get the required network details.

Step 10	**`exit`**
	Example:
	`Router(config-dhcp)# exit`
		Exits DHCP pool configuration mode.

Defining a Separate DHCP IP Address Pool for Each Cisco IP Phone

This task creates a name for the DHCP server address pool and specifies IP addresses. This method requires you to make an entry for every IP phone.

SUMMARY STEPS

1. **ip dhcp pool** *pool-name*
2. **host** *ip-address subnet-mas*k
3. **option 150 ip** *ip-address*
4. **default-router** *ip-address*
5. **exit**

DETAILED STEPS

	Command or Action	Purpose

Step 1	**`ip dhcp pool`** *`pool-name`*
	Example:
	`Router(config)# ip dhcp pool pool2`
		Creates a name for the DHCP server address pool and enters DHCP pool configuration mode.

Step 2	**`host`** *`ip-address subnet-mask`*
	Example:
	`Router(config-dhcp)# host 10.0.0.0 255.255.0.0`
		Specifies the IP address that you want the phone to use.

Step 3	**`option 150 ip`** *`ip-address`*
	Example:
	`Router(config-dhcp)# option 150 ip 10.0.22.1`
		Specifies the TFTP server address from which the Cisco IP phone downloads the image configuration file. This needs to be the IP address of CallManager.

Step 4	**`default-router`** *`ip-address`*
	Example:
	`Router(config-dhcp)# default-router 10.0.0.1`
		Specifies the router to which the Cisco IP phones are connected directly.
		• This router should be the Cisco SRST router because this is the default address that is used to obtain SRST service in the event of a WAN outage. As long as the Cisco IP phones have a connection to the Cisco SRST router, the phones are able to get the required network details.

Step 5	**`exit`**
	Example:
	`Router(config-dhcp)# exit`
		Exits DHCP pool configuration mode.

Defining the DHCP Relay Server

This task sets up DHCP relay on the LAN interface where the Cisco IP phones are connected and enables the Cisco IOS DHCP server feature to relay requests from DHCP clients (phones) to a DHCP server. For further details about DHCP configuration, see the *Cisco IOS DHCP Server* document.

The Cisco IOS DHCP server feature is enabled on routers by default. If the DHCP server is not enabled on your Cisco SRST router, use the following steps to enable it.

SUMMARY STEPS

1. **service dhcp**
2. **interface** *type number*
3. **ip helper-address** *ip-address*
4. **exit**

DETAILED STEPS

	Command or Action	Purpose

Step 1	**`service dhcp`**
	Example:
	`Router(config)# service dhcp`
		Enables the Cisco IOS DHCP Server feature on the router.

Step 2	**`interface`** *`type number`*
	Example:
	`Router(config)# interface serial 0`
		Enters interface configuration mode for the specified interface. See the *Cisco IOS Interface and Hardware Component Command Reference, Release 12.3T* for more information.

Step 3	**`ip helper-address`** *`ip-address`*
	Example:
	`Router(config-if)# ip helper-address 10.0.22.1`
		Specifies the helper address for any unrecognized broadcast for TFTP server and Domain Name System (DNS) requests. For each server, a separate **ip helper-address** command is required if the servers are on different hosts. You can also configure multiple TFTP server targets by using the **ip helper-address** commands for multiple servers.

Step 4	**`exit`**
	Example:
	`Router(config-if)# exit`
		Exits interface configuration mode.

Specifying Keepalive Intervals

The keepalive interval is the period of time between keepalive messages sent by a network device. A keepalive message is a message sent by one network device to inform another network device that the virtual circuit between the two is still active.

Note If you plan to use the default time interval between messages, which is 30 seconds, you do not have to perform this task.

SUMMARY STEPS

1. **call-manager-fallback**
2. **keepalive** *seconds*
3. **exit**

DETAILED STEPS

	Command or Action	Purpose

Step 1	**`call-manager-fallback`**
	Example:
	`Router(config)# call-manager-fallback`
		Enters call-manager-fallback configuration mode.

Step 2	**`keepalive`** *`seconds`*
	Example:
	`Router(config-cm-fallback)# keepalive 60`
		Sets the time interval, in seconds, between keepalive messages that are sent to the router by Cisco IP phones.
		• *seconds*—Range is 10 to 65535. Default is 30.

Step 3	**`exit`**
	Example:
	`Router(config-cm-fallback)# exit`
		Exits call-manager-fallback configuration mode.

Example

The following example sets a keepalive interval of 45 seconds:

```
call-manager-fallback
 keepalive 45
```

Configuring Cisco SRST to Support Phone Functions

Tip When the Cisco SRST is enabled, Cisco IP phones do not have to be reconfigured while in Cisco CallManager fallback mode because phones retain the same configuration that was used with Cisco CallManager.

To configure Cisco SRST on the router to support the Cisco IP phone functions, use the following commands beginning in global configuration mode.

SUMMARY STEPS

1. **call-manager-fallback**
2. **ip source-address** *ip-address* [**port** *port*] [**any-match** | **strict-match**]
3. **max-dn** *max-directory-numbers* [**dual-line**] [**preference** *preference-order*]
4. **max-ephones** *max-phones*
5. **limit-dn** {**7910** | **7935** | **7940** | **7960**} *max-lines*
6. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**call-manager-fallback** **Example:** Router(config)# call-manager-fallback	Enters call-manager-fallback configuration mode.
Step 2	**ip source-address** *ip-address* [**port** *port*] [**any-match** | **strict-match**] **Example:** Router(config-cm-fallback)# ip source-address 10.6.21.4 port 2002 strict-match	Enables the router to receive messages from the Cisco IP phones through the specified IP addresses and provides for strict IP address verification. The default port number is 2000.
Step 3	**max-dn** *max-directory-numbers* [**dual-line**] [**preference** *preference-order*] **Example:** Router(config-cm-fallback)# max-dn 15 dual-line preference 1	Sets the maximum number of directory numbers (DNs) or virtual voice ports that can be supported by the router and activates the dual-line mode. • *max-directory-numbers*—Maximum number of directory numbers or virtual voice ports supported by the router. The maximum number is platform-dependent. The default is 0. See the “Platform and Memory Support” section on page 24 for further details. • **dual-line**—(Optional) Allows IP phones in Cisco CallManager fallback mode to have a virtual voice port with two channels. • **preference** *preference-order* (Optional)—Sets the global preference for creating the VoIP dial peers for all directory numbers that are associated with the primary number. Range is from 0 to 10. Default is 0, which is the highest preference. The **alias** command also has a **preference** keyword that sets **alias** command preference values. Setting the **alias** command **preference** keyword allows the default preference set with the **max-dn** command to be overriden. See Configuring Call Rerouting, page 58 for more information on using the **max-dn** command with the **alias** command. **Note** You must reboot the router in order to reduce the limit of the directory numbers or virtual voice ports after the maximum allowable number is configured.

	Command or Action	Purpose
Step 4	**max-ephones** *max-phones* **Example:** `Router(config-cm-fallback)# max-ephones 24`	Configures the maximum number of Cisco IP phones that can be supported by the router. The default is 0. The maximum number is platform dependent. See the "Platform and Memory Support" section on page 24 for further details. **Note** You must reboot the router in order to reduce the limit of Cisco IP phones after the maximum allowable number is configured.
Step 5	**limit-dn** {**7910** | **7935** | **7940** | **7960**} *max-lines* **Example:** `Router(config-cm-fallback)# limit-dn 7910 2`	Limits the directory number lines on Cisco IP phones during Cisco CallManager fallback. **Note** You must configure this command during initial Cisco SRST router configuration, before any phone actually registers with the Cisco SRST router. However, you can modify the number of lines at a later time. The setting for maximum lines is from 1 to 6. The default number of maximum directory lines is set to 6. If there is any active phone with the last line number greater than this limit, warning information is displayed for phone reset.
Step 6	**exit** **Example:** `Router(config-cm-fallback)# exit`	Exits call-manager-fallback configuration mode.

Verifying That Cisco SRST Is Enabled

To verify that the Cisco SRST feature is enabled, perform the following steps:

Step 1 Enter the **show running-config** command to verify the configuration.

Step 2 Enter the **show call-manager-fallback all** command to verify that the Cisco SRST feature is enabled.

Step 3 Use the Settings display on the Cisco IP phones in your network to verify that the default router IP address on the phones matches the IP address of the Cisco SRST router.

Step 4 To temporarily block the TCP port 2000 Skinny Client Control Protocol (SCCP) connection for one of the Cisco IP phones in order to force the Cisco IP phone to lose its connection to the Cisco CallManager and register with the Cisco SRST router, perform the following steps:

a. Use the appropriate IP **access-list** command to temporarily disconnect a Cisco IP phone from the Cisco CallManager.

During a WAN connection failure, when Cisco SRST is enabled, Cisco IP phones display a message informing you that they are operating in Cisco CallManager fallback mode. The Cisco IP Phone 7960 and Cisco IP Phone 7940 display a "CM Fallback Service Operating" message, and the Cisco IP Phone 7910 displays a "CM Fallback Service" message when operating in Cisco CallManager fallback mode. When the Cisco CallManager is restored, the message goes away and full Cisco IP phone functionality is restored.

b. Enter the **no** form of the appropriate **access-list** command to restore normal service for the phone.

c. Use the **debug ephone register** command to observe the registration process of the Cisco IP phone on the Cisco SRST router.

d. Use the **show ephone** command to display the Cisco IP phones that have registered to the Cisco SRST router.

Troubleshooting

To troubleshoot your Cisco SRST configuration, use the following commands:

- To set keepalive debugging for Cisco IP phones, use the **debug ephone keepalive** command.
- To set registration debugging for Cisco IP phones, use the **debug ephone register** command.
- To set state debugging for Cisco IP phones, use the **debug ephone state** command.
- To set detail debugging for Cisco IP phones, use the **debug ephone detail** command.
- To set error debugging for Cisco IP phones, use the **debug ephone error** command.
- To set call statistics debugging for Cisco IP phones, use the **debug ephone statistics** command.
- To provide voice-packet-level debugging and to display the contents of one voice packet in every 1024 voice packets, use the **debug ephone pak** command.
- To provide raw low-level protocol debugging display for all SCCP messages, use the **debug ephone raw** command.

For further debugging, you can use the debug commands in the *Cisco IOS Debug Command Reference*.

Where to Go Next

The next step is setting up the phone and getting a dial tone. For instructions, see the "Setting Up Cisco IP Phones" chapter.

Setting Up Cisco IP Phones

This chapter describes how to set up the displays and features that callers will see and use on Cisco IP phones during Cisco CallManager fallback.

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

Information About Setting Up Cisco IP Phones

Cisco IP phone configuration is limited for Cisco Survivable Remote Site Telephony (SRST) because IP phones retain nearly all Cisco CallManager settings during Cisco CallManager fallback. You can configure the date format, time format, language, and system messages that appear on Cisco IP phones during Cisco CallManager fallback. All four of these settings have defaults, and the available language options depend on the IP phones and Cisco CallManager version in use. Also available for configuration is a secondary dial tone, which can be generated when a phone user dials a predefined PSTN access prefix and can be terminated when additional digits are dialed. Dual-line phone configuration is required for dual-line phone operation during Cisco CallManager fallback.

How to Set Up Cisco IP Phones

This section contains the following tasks:

- Configuring IP Phone Clock, Date, and Time Formats, page 46 (Optional)
- Configuring IP Phone Language Display, page 47 (Optional)
- Configuring Customized System Messages for Cisco IP Phones, page 48 (Optional)

- Configuring a Secondary Dial Tone, page 50 (Optional)
- Configuring Dual-Line Phones, page 51 (Required Under Certain Conditions)

Configuring IP Phone Clock, Date, and Time Formats

The Cisco 7970G and Cisco 7971G-GE IP phones obtain the correct timezone from Cisco CallManager. They also receive the Coordinated Universal Time (UTC) time from the SRST router during SRST registration. When in SRST mode, the phones take the timezone and the UTC time, and apply a timezone offset to produce the correct time display.

Cisco 7960 IP phones and other similar SCCP phones such as the Cisco 7940, get their display clock information from the local time of the SRST router during SRST registration. If the SRST router is configured to use the Network Time Protocol (NTP) to automatically sync the SRST router time from an NTP time server, only UTC time is delivered to the router. This is because the NTP server could be physically located anywhere in the world, in any timezone. As it is important to display the correct local time, use the clock time-zone command to adjust or offset the SRST router time.

The date and time formats that appear on the displays of all Cisco IP phones in Cisco CallManager fallback mode are selected using the **date-format** and **time-format** commands as configured below:

SUMMARY STEPS

1. **clock timezone** *zone hours-offset* [*minutes-offset*]
2. **call-manager-fallback**
3. **date-format** {**mm-dd-yy** | **dd-mm-yy** | **yy-dd-mm** | **yy-mm-dd**}
4. **time-format** {**12** | **24**}
5. **exit**

DETAILED STEPS

	Command or Action	Purpose

Step 1	`clock timezone zone hours-offset [minutes-offset]`
	Example:
	`Router(config)# clock timezone PST -8`
		Sets the time zone for display purposes.
		• *zone*—Name of the time zone to be displayed when standard time is in effect. The length of the zone argument is limited to 7 characters.
		• *hours-offset*—The number of hour difference from Coordinated Universal Time (UTC).
		• *minutes-offset*—(Optional) Minutes difference from UTC.

Step 2	`call-manager-fallback`
	Example:
	`Router(config)# call-manager-fallback`
		Enters call-manager-fallback configuration mode.

	Command or Action	Purpose

Step 3	**date-format** {**mm-dd-yy** | **dd-mm-yy** | **yy-dd-mm** | **yy-mm-dd**}
	Example:
	`Router(config-cm-fallback)# date-format yy-dd-mm`
		Sets the date format for IP phone display. The choices are **mm-dd-yy**, **dd-mm-yy**, **yy-dd-mm**, and **yy-mm-dd**, where
		• **dd**—day
		• **mm**—month
		• **yy**—year
		The default is set to **mm-dd-yy**.

Step 4	**time-format** {**12** | **24**}
	Example:
	`Router(config-cm-fallback)# time-format 24`
		Sets the time display format on all Cisco IP phones registered with the router. The default is set to a 12-hour clock.

Step 5	**exit** **Example:** `Router(config-cm-fallback)# exit`	Exits call-manager-fallback configuration mode.

Example

The following example sets the time zone to Pacific Standard Time (PST), which is 8 hours behind UTC and sets the time display format to a 24 hour clock:

```
Router(config)# clock timezone PST -8
Rounter(config)# call-manager-fallback
Rounter(config-cm-fallback)# time-format 24
```

Configuring IP Phone Language Display

During Cisco CallManager fallback, the language displays shown on Cisco IP phones default to the ISO-3166 country code of US (United States). The Cisco IP Phone 7940 and Cisco IP Phone 7960 can be configured for different languages (character sets and spelling conventions) using the **user-locale** command.

Note This configuration option is available in Cisco SRST V2.1 and later running under Cisco CallManager V3.2 and later. Systems with software prior to Cisco SRST V2.1 and Cisco CallManager V3.2 can use the default country, United States (US), only.

SUMMARY STEPS

1. **call-manager-fallback**
2. **user-locale** *country-code*
3. **exit**

DETAILED STEPS

	Command or Action	Purpose

Step 1	**`call-manager-fallback`**
	Example:
	`Router(config)# call-manager-fallback`
		Enters call-manager-fallback configuration mode.

Step 2	**`user-locale`** *`country-code`*
	Example:
	`Router(config-cm-fallback)# user-locale ES`
		Selects a language by country for displays on the Cisco IP Phone 7940 and Cisco IP Phone 7960.
		The following ISO-3166 codes are available to Cisco SRST systems running under Cisco CallManager V3.2 or later:
		• **DE**—German.
		• **DK**—Danish.
		• **ES**—Spanish.
		• **FR**—French.
		• **IT**—Italian.
		• **JP**—Japanese Katakana (available under Cisco CallManager V4.0 or later).
		• **NL**—Dutch.
		• **NO**—Norwegian.
		• **PT**—Portuguese.
		• **RU**—Russian.
		• **SE**—Swedish.
		• **US**—United States English (default).

Step 3	**`exit`**
	Example:
	`Router(config-cm-fallback)# exit`
		Exits call-manager-fallback configuration mode.

Examples

The following example offers a configuration for the Portugal user locale.

```
call-manager-fallback
 user-locale PT
```

Configuring Customized System Messages for Cisco IP Phones

The **system message** command is used to customize the system message displayed on all Cisco IP Phone 7910, Cisco IP Phone 7940G, and Cisco IP Phone 7960G units during Cisco CallManager fallback.

One of two keywords, **primary** and **secondary**, must be included in the command. The **primary** keyword is for IP phones that can support static text messages during fallback, such as the Cisco IP Phone 7940 and Cisco IP Phone 7960 units. The default display message for primary IP phones in fallback mode is “CM Fallback Service Operating.”

The **secondary** keyword is for Cisco IP phones that do not support static text messages and have a limited display space, such as the Cisco IP Phone 7910. Secondary IP phones flash messages during fallback. The default display message for secondary IP phones in fallback mode is “CM Fallback Service.”

Changes to the display message will occur immediately after configuration or at the end of each call.

Note The normal in-service static text message is controlled by Cisco CallManager.

SUMMARY STEPS

1. **call-manager-fallback**
2. **system message** {**primary** *primary-string* | **secondary** *secondary-string*}
3. **exit**

DETAILED STEPS

	Command or Action	Purpose

Step 1	`call-manager-fallback`
	Example:
	`Router(config)# call-manager-fallback`
		Enters call-manager-fallback configuration mode.

Step 2	`system message {primary primary-string | secondary secondary-string}`
	Example:
	`Router(config-cm-fallback)# system message primary Custom Message`
		Declares the text for the system display message on IP phones in fallback mode.
		• **primary** *primary-string*—For Cisco IP phones that can support static text messages during fallback, such as the Cisco IP Phone 7940 and Cisco IP Phone 7960 units. A string of approximately 27 to 30 characters is allowed.
		• **secondary** *secondary-string*—For Cisco IP phones that do not support static text messages, such as the Cisco IP Phone 7910. A string of approximately 20 characters is allowed.

Step 3	`exit`
	Example:
	`Router(config-cm-fallback)# exit`
		Exits call-manager-fallback configuration mode.

Examples

The following example sets “SRST V3.0” as the system display message for all Cisco IP phones on a router:

```
call-manager-fallback
 system message primary SRST V3.0
 system message secondary SRST V3.0
 exit
```

Configuring a Secondary Dial Tone

A secondary dial tone can be generated when a phone user dials a predefined PSTN access prefix and can be terminated when additional digits are dialed. An example is when a secondary dial tone is heard after the number 9 is dialed to reach an outside line.

SUMMARY STEPS

1. **call-manager-fallback**
2. **secondary-dialtone** *digit-string*
3. **exit**

DETAILED STEPS

	Command or Action	Purpose

Step 1	**`call-manager-fallback`**
	Example:
	`Router(config)# call-manager-fallback`
		Enters call-manager-fallback configuration mode.

Step 2	**`secondary-dialtone`** *`digit-string`*
	Example:
	`Router(config-cm-fallback)# secondary-dialtone 9`
		Activates a secondary dial tone when a digit string is dialed.

Step 3	**`exit`**
	Example:
	`Router(config-cm-fallback)# exit`
		Exits call-manager-fallback configuration mode.

Examples

The following example sets the number 8 to trigger a secondary dial tone:

```
call-manager-fallback
 secondary-dialtone 8
```

Configuring Dual-Line Phones

Dual-line phone configuration is required for dual-line phone operation during Cisco CallManager fallback. Consultative transfer is also required (see the "Enabling Consultative Call Transfer and Forward Using H.450.2 and H.450.3 with Cisco SRST V3.0" section on page 74).

Dual-line IP phones are supported during Cisco CallManager fallback using the **max-dn** command. Dual-line IP phones have one voice port with two channels to handle two independent calls. This capability enables call waiting, call transfer, and conference functions on a phone-line button.

In dual-line mode, each IP phone and its associated line button can support one or two calls. Selection of one of two calls on the same line is made using the blue Navigation button located below the phone display. When one of the dual-line channels is used on a specific phone, other phones that share the ephone-dn will be unable to use the secondary channel. The secondary channel will be reserved for use with the primary dual-line channel.

It is recommended that hunting be disabled to the second channel. For more information, see the "Configuring Dial-Peer and Channel Hunting" section on page 70.

SUMMARY STEPS

1. **call-manager-fallback**
2. **max-dn** *max-directory-numbers* [**dual-line**] [**preference** *preference-order*]
3. **exit**

DETAILED STEPS

	Command or Action	Purpose

Step 1	**`call-manager-fallback`**
	Example:
	`Router(config)# call-manager-fallback`
		Enters call-manager-fallback configuration mode.

Step 2	**`max-dn`** *`max-directory-numbers`* `[`**`dual-line`**`] [`**`preference`** *`preference-order`*`]`
	Example:
	`Router(config-cm-fallback)# max-dn 15 dual-line preference 1`
		Sets the maximum number of directory numbers (DNs) or virtual voice ports that can be supported by the router and activates the dual-line mode.
		• *max-directory-numbers*—Maximum number of directory numbers or virtual voice ports supported by the router. The maximum number is platform-dependent. The default is 0. See the "Platform and Memory Support" section on page 24 for further details.
		• **dual-line**—(Optional) Allows IP phones in Cisco CallManager fallback mode to have a virtual voice port with two channels.
		• **preference** *preference-order* (Optional)—Sets the global preference for creating the VoIP dial peers for all directory numbers that are associated with the primary number. Range is from 0 to 10. Default is 0, which is the highest preference.
		The **alias** command also has a **preference** keyword that sets **alias** command preference values. Setting the **alias** command **preference** keyword allows the default preference set with the **max-dn** command to be overriden. See Configuring Call Rerouting, page 58 for more information on using the **max-dn** command with the **alias** command.

Step 3	**`exit`**
	Example:
	`Router(config-cm-fallback)# exit`
		Exits call-manager-fallback configuration mode.

Examples

The following example sets the maximum number of DNs or virtual voice ports that can be supported by a router to 10 and activates the dual-line mode for all IP phones in Cisco CallManager fallback mode.

```
call-manager-fallback
 max-dn 10 dual-line
 exit
```

Where to Go Next

The next step is setting up call handling. For instructions, see the "Setting Up Call Handling" chapter.

Setting Up Call Handling

This chapter describes how to configure Cisco Survivable Remote Site Telephony (SRST) for incoming calls and outgoing calls.

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

Information About Setting Up Call Handling

Cisco SRST offers a smaller set of call handling capabilities than Cisco CallManager, and much of the configuration for these feature involves enabling existing Cisco CallManager or IP phone settings.

How to Set Up Call Handling

Setting up call handling involves the following set of tasks:

- Configuring Incoming Calls, page 56
- Configuring Outgoing Calls, page 73

Configuring Incoming Calls

Incoming call configuration can include the following tasks:

- Call Forwarding and Rerouting
  - Configuring Call Forwarding During a Busy Signal or No Answer, page 56 (Optional)
  - Configuring Call Rerouting, page 58 (Optional)
  - Configuring Call Pickup, page 61 (Optional)
- Phone Number Conversion and Translation
  - Configuring Global Prefixes, page 63 (Optional)
  - Enabling Digit Translation Rules, page 65 (Optional)
  - Enabling Translation Profiles, page 66 (Optional)
  - Verifying Translation Profiles, page 69 (Optional)
- Hunting and Ringing Timeout Behavior
  - Configuring Dial-Peer and Channel Hunting, page 70 (Optional)
  - Configuring Busy Timeout, page 71 (Optional)
  - Configuring the Ringing Timeout Default, page 72 (Optional)

Configuring Call Forwarding During a Busy Signal or No Answer

Incoming calls that reach a busy signal or go unanswered during Cisco CallManager fallback can be configured to be forwarded to one or more E.164 numbers.

SUMMARY STEPS

1. **call-manager-fallback**
2. **call-forward busy** *directory-number*
3. **call-forward noan** *directory-number* **timeout** *seconds*
4. **exit**

DETAILED STEPS

	Command or Action	Purpose

Step 1	**`call-manager-fallback`**
	Example:
	`Router(config)# call-manager-fallback`
		Enters call-manager-fallback configuration mode.

Step 2	**`call-forward busy`** *`directory-number`*
	Example:
	`Router(config-cm-fallback)# call-forward busy 50..`
		Configures call forwarding to another number when the Cisco IP phone is busy.
		• *directory-number*—Selected directory number representing a fully qualified E.164 number. This number can contain "." wildcard characters that correspond to the right-justified digits in the directory number extension.

Step 3	**`call-forward noan`** *`directory-number`* **`timeout`** *`seconds`*
	Example:
	`Router(config-cm-fallback)# call-forward noan 5005 timeout 10`
		Configures call forwarding to another number when no answer is received from the Cisco IP phone.
		• *directory-number*—Selected directory number representing a fully qualified E.164 number or a local extension number. This number can contain "." wildcard characters that correspond to the right-justified digits in the directory number extension.
		• **timeout** *seconds*—Sets the waiting time, in seconds, before the call is forwarded to another phone. The *seconds* range is from 3 to 60000.

Step 4	**`exit`**
	Example:
	`Router(config-cm-fallback)# exit`
		Exits call-manager-fallback configuration mode.

Examples

The following example forwards calls to extension number 5005 when an incoming call reaches a busy or unattended IP phone extension number. Incoming calls will ring for 15 seconds before being forwarded to extension 5005.

```
call-manager-fallback
 call-forward busy 5005
 call-forward noan 5005 timeout seconds 15
```

The following example transforms an extension number for call forwarding when the extension number is busy or unattended. The **call-forward busy** command has an argument of 50.., which prepends the digits 50 to the last two digits of the called extension. The resulting extension is the number to which incoming calls are forwarded when the original extension number is busy or unattended. For instance, an incoming call to the busy extension 6002 will be forwarded to extension 5002, and an incoming call to the busy extension 3442 will be forwarded to extension 5042. Incoming calls will ring for 15 seconds before being forwarded.

```
call-manager-fallback
 call-forward busy 50..
 call-forward noan 50.. timeout seconds 15
```

Configuring Call Rerouting

Note The **alias** command obsoletes the **default-destination** command and is recommended over the **default-destination** command.

The **alias** command provides a mechanism for rerouting calls to telephone numbers that are unavailable during fallback. Up to 50 sets of rerouting alias rules can be created for calls to telephone numbers that are unavailable during Cisco CallManager fallback. Sets of alias rules are created using the **alias** command. An alias is activated when a telephone registers that has a phone number matching a configured *alternate-number* alias. Under that condition, an incoming call is rerouted to the alternate number. The *alternate-number* argument can be used in multiple **alias** commands, allowing you to reroute multiple different numbers to the same target number.

The configured *alternate-number* must be a specific E.164 phone number or extension that belongs to an IP phone registered on the Cisco SRST router. When an IP phone registers with a number that matches an *alternate-number*, an additional POTS dial peer is created. The destination pattern is set to the initial configured *number-pattern*, and the POTS dial peer voice port is set to match the voice port associated with the *alternate-number.*

If other IP phones register with specific phone numbers within the range of the initial *number-pattern*, the call is routed back to the IP phone rather than to the *alternate-number* (according to normal dial-peer longest-match, preference, and huntstop rules).

Call Forward Destination

The **cfw** keyword allows you to configure a call forward destination for calls that are busy or not answered. Call forward no answer is defined as when the phone rings for a user configurable amount of time, the call is not answered, and is forwarded to the configured destination. Call forward busy and call forward no answer can be configured to a set string and override globally configured call forward settings.

Note Globally configured settings are selected under call-manager-fallback and apply to all phones that register for SRST service.

You can also create a specific call forwarding path for a particular number. The benefit of using the **cfw** keyword is that during SRST, you can reroute calls from otherwise unreachable numbers onto phones that are available. Basic hunt groups can be established with call-forwarding rules so that if the first SRST phone is busy, you can forward the call to a second SRST phone.

The **cfw** keyword also allows you to alias a phone number to itself, permitting setting of per-phone number forwarding. An example of aliasing a number to itself follows. If a phone registers with extension 1001, a dial peer that routes calls to the phone is automatically created for 1001. If the call-manager-fallback dial-peer preference (set with the **max-dn** command) for this initial dial peer is set to 2, the dial peer uses 2 as its preference setting.

Then, use the **alias** command to alias the phone number to itself:

```
alias 1 1001 to 1001 preference 1 cfw 2001 timeout 20
```

In this example, you have created a second dial peer for 1001 to route calls to 1001, but that has preference 1 and call forwarding to 2001. Because the preference on the dial peer created by the **alias** command is now a lower numeric value than the preference that the dial peer first created, all calls come initially to the dial peer created by the **alias** command. In that way they are subject to the forward as set by the **alias** command, instead of any call forwarding that may have been set globally.

Huntstop on an Individual Alias

The alias **huntstop** keyword is relevant only if you have also set the global **no huntstop** command under call-manager-fallback. Also, you may need to set the global **no huntstop** if you have multiple **alias** commands with the same *number-pattern*, and you want to enable hunting on busy between the aliases. That is, one alias for *number-pattern* is tried, and then if that phone is busy, the second alias for *number-pattern* is tried.

The alias **huntstop** keyword allows you to turn huntstop behavior back on for an individual alias, if huntstop is turned off globally by the **no huntstop** command. Setting the **huntstop** keyword on an individual alias stops hunting at the alias, making the alias the final member of the hunt sequence.

SUMMARY STEPS

1. **call-manager-fallback**
2. **alias** *tag number-pattern* **to** *alternate-number* [**preference** *preference-value*] [**cfw** *number* **timeout** *timeout-value*] [**huntstop**]
3. **max-dn** *max-directory-numbers* [**dual-line**] [**preference** *preference-order*]
4. **end**
5. **show dial-peer voice summary**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**call-manager-fallback** **Example:** `Router(config)# call-manager-fallback`	Enters call-manager-fallback configuration mode.
Step 2	**alias** *tag number-pattern* **to** *alternate-number* [**preference** *preference-value*] [**cfw** *number* **timeout** *timeout-value*] [**huntstop**] **Example:** `Router(config-cm-fallback)# alias 1 60.. to 5001 preference 1 cfw 2000 timeout 10`	Creates a set rules for rerouting calls to sets of phones that are unavailable during Cisco CallManager fallback. • *tag*—Identifier for alias rule range. The range is from 1 to 50. • *number-pattern*—Pattern to match the incoming telephone number. This pattern may include wildcards. • **to**—Connects the tag number pattern to the alternate number. • *alternate-number*—Alternate telephone number to route incoming calls to match the number pattern. The alternate number has to be a specific extension that belongs to an IP phone that is actively registered on the Cisco SRST router. The alternate telephone number can be used in multiple **alias** commands. • **preference** *preference-value*—(Optional) Assigns a dial-peer preference value to the alias. The preference value of the associated dial peer is from 0 to 10. Use with the **max-dn** command. • **cfw** *number*—(Optional) The **cfw** keyword allows users to set call forward busy and call forward no answer to a set string and override globally configured call forward settings. • **timeout** *timeout-value*—(Optional) Sets the ring no-answer timeout duration for call forwarding, in seconds. Range is from 3 to 60000. • **huntstop**—(Optional) Stops call hunting after trying the alternate number.
Step 3	**max-dn** *max-directory-numbers* [**dual-line**] [**preference** *preference-order*] **Example:** `Router(config-cm-fallback)# max-dn 10 preference 2`	Sets the maximum possible number of directory numbers or virtual voice ports that can be supported by a router and sets the global preference for creating the VoIP dial peers for all directory numbers that are associated with the primary number. • Using the **max-dn** command sets the preference for the default dial peers created with the **alias** command. • When configuring call rerouting, set the **max-dn preference** to a higher numeric preference than the preference that was set with the **alias** command.

	Command or Action	Purpose

Step 4	**end**
	Example:
	`Router(config-cm-fallback)# end`
		Returns to privileged EXEC mode.

Step 5	**show dial-peer voice summary** **Example:** `Router# show dial-peer voice summary`	Displays information for voice dial peers. • If you suspect a problem with the dial peers, use this command to display the dial peers created by the **alias** command.

Example

The following example sets the **preference** keyword in the **alias** command to a lower preference value that the preference value created by the **max-dn** command. Setting the value lower allows the **cfw** keyword to take effect. The incoming call to extension 1000 hunts to alias because it has a lower preference, and no-answer/busy calls to 1000 are forwarded to 2000. All incoming calls to other extensions in SRST mode are forwarded to 3000 after 10 seconds.

```
call-manager-fallback
 alias 1 1000 to 1000 preference 1 cfw 2000 timeout 10
 max-dn 10 preference 2
 call-forward busy 3000
 call-forward noan 3000 timeout 10
```

Configuring Call Pickup

Configuring the **pickup** command enables the PickUp soft key on all SRST phones. You can then press the PickUp key and answer any currently ringing IP phone that has a DID called number that matches the configured *telephone-number*. This command does not enable the Group PickUp (GPickUp) soft key.

When a user presses the PickUp soft key, SRST searches through all the SRST phones to find a ringing call that has a called number that matches the configured *telephone-number*. When a match is found, the call is automatically forwarded to the extension number of the phone that requested the call pickup.

The SRST **pickup** command is designed to operate in a manner compatible with Cisco CallManager.

Note The default phone load on Cisco CallManager, Release 4.0(1), for the Cisco 7905 and Cisco 7912 IP phones does not enable the PickUp soft key during fallback. To enable the PickUp soft key on Cisco 7905 and Cisco 7912 IP phones, upgrade your default phone load to Cisco CallManager, Release 4.0(1) Sr2. Alternatively, you can upgrade the phone load to cmterm-7905g-sccp.3-3-8.exe or cmterm-7912g-sccp.3-3-8.exe, respectively.

SUMMARY STEPS

1. **call-manager-fallback**
2. **no huntstop**
3. **alias** *tag number-pattern* **to** *alternate-number*
4. **pickup** *telephone-number*
5. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**call-manager-fallback** **Example:** `Router(config)# call-manager-fallback`	Enters call-manager-fallback configuration mode.
Step 2	no huntstop **Example:** `Router(config-cm-fallback)# no huntstop`	Disables huntstop.
Step 3	**alias** *tag number-pattern* **to** *alternate-number* **Example:** `Router(config-cm-fallback)# alias 1 8005550100 to 5001`	Creates a set rules for rerouting calls to sets of phones that are unavailable during Cisco CallManager fallback. • *tag*—Identifier for alias rule range. The range is from 1 to 50. • *number-pattern*—Pattern to match the incoming telephone number. This pattern may include wildcards. • **to**—Connects the tag number pattern to the alternate number. • *alternate-number*—Alternate telephone number to route incoming calls to match the number pattern. The alternate number has to be a specific extension that belongs to an IP phone that is actively registered on the Cisco SRST router. The alternate telephone number can be used in multiple **alias** commands.
Step 4	**pickup** *telephone-number* **Example:** `Router(config-cm-fallback)# pickup 8005550100`	Enables the PickUp soft key on all Cisco IP phones, allowing an external Direct Inward Dialing (DID) call coming into one extension to be picked up from another extension during SRST. The *telephone-number* argument is the telephone number to match an incoming called number.

Step 5	**end**
	Example:
	`Router(config-cm-fallback)# end`
		Returns to privileged EXEC mode.

Example

The **pickup** command is best used with the **alias** command. The following partial output from the **show running-config** command shows the **pickup** command and the **alias** command configured to provide call routing for a pilot number of a hunt group.

```
call-manager-fallback
 no huntstop
 alias 1 8005550100 to 5001
 alias 2 8005550100 to 5002
 alias 3 8005550100 to 5003
 alias 4 8005550100 to 5004
 pickup 8005550100
```

When a DID incoming call to 800 555-0100 is received, the **alias** command routes the call at random to one of the four extensions (5001 to 5004). Because the **pickup** command is configured, if the DID call rings on extension 5002, the call can be answered from any of the other extensions (5001, 5003, 5004) by pressing the PickUp soft key.

The **pickup** command works by finding a match based on the incoming DID called number. In this example, a call from extension 5004 to extension 5001 (an internal call) does not activate the **pickup** command because the called number (5001) does not match the configured pickup number (800 555-0100). Thus, the **pickup** command distinguishes between internal and external calls if multiple calls are ringing simultaneously.

Configuring Global Prefixes

The **dialplan-pattern** command creates a dial-plan pattern that specifies a global prefix for the expansion of abbreviated extension numbers into fully qualified E.164 numbers.

The **extension-pattern** keyword allows additional manipulation of abbreviated extension-number prefix digits. When this keyword and its argument are used, the leading digits of an extension pattern are stripped and replaced by the corresponding leading digits of the dial-plan pattern. This command can be used to avoid Direct Inward Dialing (DID) numbers like 408 555-0101 resulting in 4-digit extensions such as 0101.

Global prefixes are set with the **dialplan-pattern** command. Up to five dial-plan patterns can be created. The **no-reg** keyword provides dialing flexibility and prevents the E.164 numbers in the dial peer from registering to the gatekeeper. You have the option not to register numbers to the gatekeeper so that those numbers can be used for other telephony services.

SUMMARY STEPS

1. **call-manager-fallback**
2. **dialplan-pattern** *tag pattern* **extension-length** *length* [**extension-pattern** *extension-pattern*] [**no-reg**]
3. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**call-manager-fallback** **Example:** `Router(config)# call-manager-fallback`	Enters call-manager-fallback configuration mode.
Step 2	**dialplan-pattern** *tag pattern* **extension-length** *length* [**extension-pattern** *extension-pattern*] [**no-reg**] **Example:** `Router(config-cm-fallback)# dialplan-pattern 1 4085550100 extension-length 3 extension-pattern 4..` **Note** This example maps all extension numbers 4xx to the PSTN number 40855501xx, so that extension 412 corresponds to 4085550112.	Creates a global prefix that can be used to expand the abbreviated extension numbers into fully qualified E.164 numbers • *tag*—Dial-plan string tag used before a 10-digit telephone number. The tag number is from 1 to 5. • *pattern*—Dial-plan pattern, such as the area code, the prefix, and the first one or two digits of the extension number, plus wildcard markers or dots (.) for the remainder of the extension number digits. • **extension-length**—Sets the number of extension digits. • *length*—The number of extension digits. The range is from 1 to 32. • **extension-pattern**—(Optional) Sets an extension number's leading digit pattern when it is different from the E.164 telephone number's leading digits defined in the *pattern* argument. • *extension-pattern*—(Optional) The extension number's leading digit pattern. Consists of one or more digits and wildcard markers or dots (.). For example, 5.. would include extension 500 to 599; 5... would include 5000 to 5999. • **no-reg**—(Optional) Prevents the E.164 numbers in the dial peer from registering with the gatekeeper.
Step 3	**exit** **Example:** `Router(config-cm-fallback)# exit`	Exits call-manager-fallback configuration mode.

Examples

The following example shows how to create dial-plan pattern 1 for extension numbers 101 to 199 with the telephone prefix starting with 4085550. If the following example is set, the router will recognize that 4085550144 matches dial-plan pattern 1. It will use the **extension-length** keyword to extract the last three digits of the number 144 and present this as the caller ID for the incoming call.

```
call-manager-fallback
 dialplan-pattern 1 40855501.. extension-length 3 no-reg
```

In the following example, the leading prefix digit for the 3-digit extension numbers is transformed from 0 to 4, so that the extension-number range becomes 400 to 499.

```
call-manager-fallback
 dialplan-pattern 1 40855500.. extension-length 3 extension-pattern 4..
```

In the following example, the **dialplan-pattern** command creates dial-plan pattern 2 for extensions 801 to 899 with the telephone prefix starting with 4085559. As each number in the extension pattern is declared with the number command, two POTS dial peers are created. In the example, they are 801 (an internal office number) and 4085559001 (an external number).

```
call-manager-fallback
 dialplan-pattern 2 40855590.. extension-length 3 extension-pattern 8..
```

Enabling Digit Translation Rules

Digit translation rules can be enabled during Cisco CallManger fallback. Translation rules are a number-manipulation mechanism that performs operations such as automatically adding telephone area codes and prefix codes to dialed numbers. Translation rules can be used as follows:

- To manipulate the answer number indication (ANI) (calling number) or dialed number identification service (DNIS) (called number) digits for a voice call.
- To convert a telephone number into a different number before the call is matched to an inbound dial peer or before the call is forwarded by the outbound dial peer.

To view the translation rules configured for your system, use the **show translation-rule** command.

Note Digit translation rules have many applications and variations. For further information about them, see the "Configuration Dial Plans, Dial Peers, and Digit Manipulation" chapter of the *Cisco IOS Voice, Video, and Fax Configuration Guide*, Release 12.2.

If you are running Cisco SRST 3.2 or a later version, use the configuration described in the "Enabling Translation Profiles" section on page 66 instead of using the **translate** command as described below. Translation Profiles are new to Cisco SRST 3.2 and provide added capabilities.

SUMMARY STEPS

1. **call-manager-fallback**
2. **translate** {**called** | **calling**} *translation-rule-tag*
3. **exit**

DETAILED STEPS

	Command or Action	Purpose

Step 1	**`call-manager-fallback`**
	Example:
	`Router(config)# call-manager-fallback`
		Enters call-manager-fallback configuration mode.

Step 2	**`translate`** `{`**`called`** `|` **`calling`**`}` *`translation-rule-tag`*
	Example:
	`Router(config-cm-fallback)# translate called 20`
		Applies a translation rule to modify the phone number dialed or received by any Cisco IP phone user while CallManager fallback is active.
		• **called**—Applies the translation rule to an outbound call number.
		• **calling**—Applies the translation rule to an inbound call number.
		• *translation-rule-tag*—The reference number of the translation rule from 1 to 2147483647.

Step 3	**`exit`**
	Example:
	`Router(config-cm-fallback)# exit`
		Exits call-manager-fallback configuration mode.

Examples

The following example applies translation rule 10 to the calls coming into extension 1111. All inbound calls to 1111 will go to 2222 during Cisco CallManager fallback.

```
translation-rule 10
 rule 1 1111 2222 abbreviated
 exit
call-manager-fallback
 translate calling 10
```

The following is a sample configuration of digit translation rule 20, where the priority of the translation rule is 1 (the range is from 1 to 15) and the abbreviated representation of a complete number (1234) is replaced with the number 2345:

```
translation-rule 20
 rule 1 1234 2345 abbreviated
 exit
```

Enabling Translation Profiles

Cisco SRST version 3.2 and later versions support translation profiles. Translation profiles are the suggested way to allow you to group translation rules and provide instructions on how to apply the translation rules to the following:

- Called numbers
- Calling numbers
- Redirected called numbers

In the configuration below, the **voice translation-rule** and the **rule** command allow you to set and define how a number is to be manipulated. The **translate** command in voice translation-profile mode defines the type of number you are going to manipulate; such as a called, calling, or a redirecting number. Once you have defined your translation profiles, you can then apply the translation profiles in various places, such as dial peers and voice ports. For SRST, you apply your profiles in call-manager fallback mode.

Cisco IP phones support one incoming and one outgoing translation profile when in SRST mode.

Note For Cisco SRST Version 3.2 and later versions use the **voice translation-rule** and **translation-profile** commands shown below instead of the translation rule configuration described in "Enabling Digit Translation Rules" section on page 65. Voice translation rules are a separate feature from translation rules. See the **voice translation-rule** command in the *Cisco IOS Voice Command Reference*, Release 12.3 T for more information, and the *VoIP Gateway Trunk and Carrier Based Routing Enhancements* documentation for more general information on translation rules and profiles.

SUMMARY STEPS

1. **voice translation-rule** *number*
2. **rule** *precedence***/***match-pattern***/ /***replace-pattern***/**
3. **exit**
4. **voice translation-profile** *name*
5. **translate** {**called** | **calling** | **redirect-called**} *voice-translation-rule-tag*
6. **exit**
7. **call-manager-fallback**
8. **translation-profile** {**incoming** | **outgoing**} *name*
9. **exit**

DETAILED STEPS

	Command or Action	Purpose

Step 1	**`voice translation-rule`** *`number`*
	Example:
	`Router(config)# voice translation-rule 1`
		Defines a translation rule for voice calls and enters voice translation-rule configuration mode.
		• *number*—Number that identifies the translation rule. Range is from 1 to 2147483647.

Step 2	**`rule`** *`precedence`***`/`***`match-pattern`***`/`** **`/`***`replace-pattern`***`/`**
	Example:
	`Router(cfg-translation-rule)# rule 1/^9/ //`
		Defines a translation rule.
		• *precedence*—Priority of the translation rule. Range is from 1 to 15.
		• *match-pattern*—Stream editor (SED) expression used to match incoming call information. The slash (/) is a delimiter in the pattern.
		• *replace-pattern*—SED expression used to replace the match pattern in the call information. The slash (/) is a delimiter in the pattern.

Step 3	**`exit`**
	Example:
	`Router(cfg-translation-rule)# exit`
		Exits voice translation-rule configuration mode.

Step 4	**`voice translation-profile`** *`name`*
	Example:
	`Router(config)# voice translation-profile name1`
		Defines a translation profile for voice calls.
		• *name*—Name of the translation profile. Maximum length of the voice translation profile name is 31 alphanumeric characters.

Step 5	**`translate`** `{`**`called`** `|` **`calling`** `|` **`redirect-called`**`}` *`translation-rule-number`*
	Example:
	`Router(cfg-translation-profile)# translate called 1`
		Associates a voice translation rule with a voice translation profile.
		• **called**—Associates the translation rule with called numbers.
		• **calling**—Associates the translation rule with calling numbers.
		• **redirect-called**—Associates the translation rule with redirected called numbers.
		• *translation-rule-number*—The reference number of the translation rule from 1 to 2147483647.

Step 6	**`exit`**
	Example:
	`Router(cfg-translation-profile)# exit`
		Exits translation-profile configuration mode.

Step 7	**`call-manager-fallback`**
	Example:
	`Router(config)# call-manager-fallback`
		Enters call-manager-fallback configuration mode.

	Command or Action	Purpose
Step 8	**translation-profile** {**incoming** | **outgoing**} *name* **Example:** `Router(config-cm-fallback)# translation-profile outgoing name1`	Assigns a translation profile for incoming or outgoing call legs on a Cisco IP phone. • **incoming**—Applies the translation profile to incoming calls. • **outgoing**—Applies the translation profile to outgoing calls. • *name*—The name of the translation profile.
Step 9	**exit** **Example:** `Router(config-cm-fallback)# exit`	Exits call-manager-fallback configuration mode.

Example

The following example shows the configuration where a translation profile called name1 is created with two voice translation rules. Rule1 consists of associated calling numbers, and rule2 consists of redirected called numbers. The Cisco IP phones in SRST mode are configured with name1.

```
voice translation-profile name1
 translate calling 1
 translate called redirect-called 2

call-manager-fallback
 translation-profile incoming name1
```

Verifying Translation Profiles

To verify translation profiles, perform the following steps.

SUMMARY STEPS

1. **show voice translation-rule** *number*
2. **test voice translation-rule** *number input-test-string* [**type** *match-type* [**plan** *match-type*]]

DETAILED STEPS

Step 1 **show voice translation-rule** *number*

Use this command to verify the translation rules that you have defined for your translation profiles.

```
Router# show voice translation-rule 6

Translation-rule tag: 6
   Rule 1:
   Match pattern: 65088801..
   Replace pattern: 6508880101
   Match type: none   Replace type: none
   Match plan: none   Replace plan: none
```

Step 2 **test voice translation-rule** *number input-test-string* [**type** *match-type* [**plan** *match-type*]]

Use this command to test your translation profiles. See the **test voice translation-rule** command in the *Cisco IOS Voice Command Reference*, Release 12.3 T for more information.

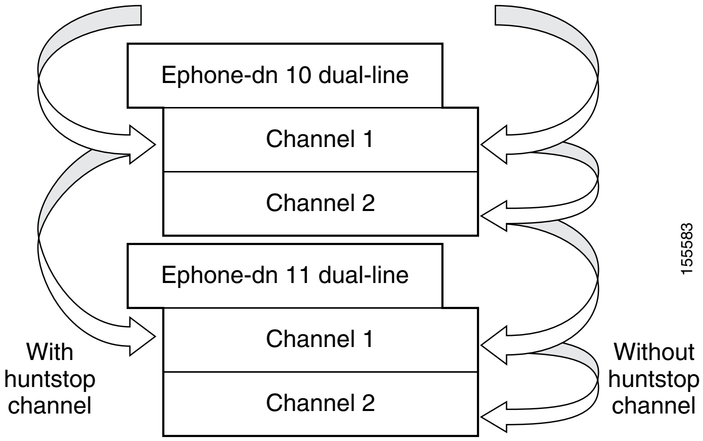
```
Router(config)# voice translation-rule 5
Router(cfg-translation-rule)# rule 1 /201/ /102/
Router(cfg-translation-rule)# end
Router# test voice translation-rule 5 2015550101
Matched with rule 5
Original number:2015550101  Translated number:1025550101
Original number type: none    Translated number type: none
Original number plan: none    Translated number plan: none
```

Configuring Dial-Peer and Channel Hunting

Dial-peer hunting, the search through a group of dial peers for an available phone line, is disabled during Cisco CallManager fallback by default. To enable dial-peer hunting, use the **no huntstop** command. For more information about dial-peer hunting, see the "Configuring Dial Peer Hunting" section in the *Cisco IOS Voice, Video, and Fax Configuration Guide*, Release 12.2.

If you have a dual-line phone configuration (see the "Configuring Dual-Line Phones" section on page 56), you may want to keep incoming calls from hunting to the second channel if the first channel is busy or does not answer by using the **channel** keyword in the **huntstop** command. As show in Figure 3, this keeps the second channel free for call transfer, call waiting, or three-way conferencing.

Figure 3 Hunt Pattern for Dual-Line Configurations With and Without Huntstop



Channel huntstop also prevents situations in which a call can ring for 30 seconds on the first channel of a line with no person available to answer and then ring for another 30 seconds on the second channel before rolling over to another line.

SUMMARY STEPS

1. **call-manager-fallback**
2. **huntstop** [**channel**]
3. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**call-manager-fallback** **Example:** `Router(config)# call-manager-fallback`	Enters call-manager-fallback configuration mode.
Step 2	**huntstop** [**channel**] **Example:** `Router(config-cm-fallback)# huntstop channel`	Sets the huntstop attribute for the dial peers associated with the Cisco IP phone dial peers created during CallManager fallback. • For dual-line configurations, the **channel** keyword keeps incoming calls from hunting to the second channel if the first channel is busy or does not answer.
Step 3	**exit** **Example:** `Router(config-cm-fallback)# exit`	Exits call-manager-fallback configuration mode.

Example

The following example disables dial-peer hunting during Cisco CallManager fallback and hunting to the secondary channels in dual-line phone configurations:

```
call-manager-fallback
 no huntstop channel
```

Configuring Busy Timeout

This task sets the timeout value for call transfers to busy destinations. The busy timeout value is the amount of time that can elapse after a transferred call reaches a busy signal before the call is disconnected.

SUMMARY STEPS

1. **call-manager-fallback**
2. **timeouts busy** *seconds*
3. **exit**

DETAILED STEPS

	Command or Action	Purpose

Step 1	**`call-manager-fallback`**
	Example:
	`Router(config)# call-manager-fallback`
		Enters call-manager-fallback configuration mode.

Step 2	**`timeouts busy`** *`seconds`*
	Example:
	`Router(config-cm-fallback)# timeouts busy 20`
		Sets the amount of time after which calls are disconnected when they are transferred to busy destinations.
		• *seconds*—Number of seconds. Range is from 0 to 30. Default is 10.
		Note This command sets the busy timeout only for calls that are transferred to busy destinations and does not affect the timeout for calls that directly dial busy destinations.

Step 3	**`exit`**
	Example:
	`Router(config-cm-fallback)# exit`
		Exits call-manager-fallback configuration mode.

Example

The following example sets a timeout of 20 seconds for calls that are transferred to busy destinations:

```
call-manager-fallback
 timeouts busy 20
```

Configuring the Ringing Timeout Default

The ringing timeout default is the length of time for which a phone can ring with no answer before returning a disconnect code to the caller. This timeout prevents hung calls received over interfaces such as Foreign Exchange Office (FXO) that do not have forward-disconnect supervision. It is used only for extensions that do not have no-answer call forwarding enabled.

SUMMARY STEPS

1. **call-manager-fallback**
2. **timeouts ringing** *seconds*
3. **exit**

DETAILED STEPS

	Command or Action	Purpose

Step 1	**`call-manager-fallback`**
	Example:
	`Router(config)# call-manager-fallback`
		Enters call-manager-fallback configuration mode.

Step 2	**`timeouts ringing`** *`seconds`*
	Example:
	`Router(config-cm-fallback)# timeouts ringing 30`
		Sets the ringing timeout default, in seconds. The range is from 5 to 60000. There is no default value.

Step 3	**`exit`**
	Example:
	`Router(config-cm-fallback)# exit`
		Exits call-manager-fallback configuration mode.

Example

The following example sets the ringing timeout default to 30 seconds:

```
call-manager-fallback
 timeouts ringing 30
```

Configuring Outgoing Calls

Outgoing call configuration can include the following tasks:

- Configuring Call Transfer
  - Configuring Local and Remote Call Transfer, page 73 (Optional)
  - Enabling Consultative Call Transfer and Forward Using H.450.2 and H.450.3 with Cisco SRST V3.0, page 74 (Optional)
  - Enabling Analog Transfer Using Hookflash and the H.450.2 Standard with Cisco SRST V3.0 or Lower, page 78 (Optional)
- Configuring Trunk Access Codes, page 81 (Required Under Certain Conditions)
- Configuring Interdigit Timeout Values, page 82 (Optional)
- Configuring Class of Restriction, page 83 (Optional)
- Call Blocking (Toll Bar) Based on Time of Day and Day of Week or Date, page 87 (Optional)

Configuring Local and Remote Call Transfer

You must configure Cisco SRST to allow Cisco IP phones to transfer telephone calls from outside the local IP network to another Cisco IP phone. By default, all Cisco IP phone directory numbers or virtual voice ports are allowed as transfer targets. A maximum of 32 transfer patterns can be entered.

Call transfer configuration is performed using the **transfer-pattern** command.

SUMMARY STEPS

1. **call-manager-fallback**
2. **transfer-pattern** *transfer-pattern*
3. **exit**

DETAILED STEPS

	Command or Action	Purpose

Step 1	**`call-manager-fallback`**
	Example:
	`Router(config)# call-manager-fallback`
		Enters call-manager-fallback configuration mode.

Step 2	**`transfer-pattern`** *`transfer-pattern`*
	Example:
	`Router(config-cm-fallback)# transfer-pattern 52540..`
		Enables the transfer of a call from a non-IP phone number to another Cisco IP phone on the same IP network using the specified transfer pattern.
		• *transfer-pattern*—String of digits for permitted call transfers. Wildcards are permitted.

Step 3	**`exit`**
	Example:
	`Router(config-cm-fallback)# exit`
		Exits call-manager-fallback configuration mode.

Example

In the following example, the **transfer-pattern** command permits transfers from a non-IP phone number to any Cisco IP phone on the same IP network with a number in the range from 5550100 to 5550199:

```
call-manager-fallback
 transfer-pattern 55501..
```

Enabling Consultative Call Transfer and Forward Using H.450.2 and H.450.3 with Cisco SRST V3.0

Consultative call transfer using H.450.2 adds support for initiating call transfers and call forwarding on a call leg using the ITU-T H.450.2 and ITU-T H.450.3 standards. Call transfers and call forwarding using H.450.2 and H.450.3 can be blind or consultative. A blind call transfer or blind call forward is one in which the transferring or forwarding phone connects the caller to a destination line before a ringing tone begins. A consultative transfer is one in which the transferring or forwarding party either connects the caller to a ringing phone (ringback heard) or speaks with the third party before connecting the caller to the third party.

Note For Cisco SRST Versions 3.1 and higher, call transfer and call forward using H.450.2 is supported automatically with the default session application.

Prerequisites

- Call transfer with consultation is available only when a second line or call instance is supported by the IP phone. Please see the **dual-line** keyword in the **max-dn** command.
- All voice gateway routers in the VoIP network must support the H.450 standard.
- All voice gateway routers in the VoIP network must be running the following software:
  - Cisco IOS Release 12.3(2)T or a later release
  - Cisco SRST V3.0

Restrictions

H.450.12 Supplementary Services Capabilities exchange among routers is not implemented.

SUMMARY STEPS

1. **call-manager-fallback**
2. **call-forward pattern** *pattern* (call forward only)
3. **transfer-system** {**blind** | **full-blind** | **full-consult** | **local-consult**} (call transfer only)
4. **transfer-pattern** *transfer-pattern* (call transfer only)
5. **exit**
6. **voice service voip**
7. **h323**
8. **h450 h450-2 timeout** {**T1** | **T2** | **T3** | **T4**} *milliseconds*
9. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**call-manager-fallback** **Example:** `Router(config)# call-manager-fallback`	Enters call-manager-fallback configuration mode.
Step 2	**call-forward pattern** *pattern* **Example:** `Router(config-cm-fallback)# call-forward pattern 4...`	Specifies the H.450.3 standard for call forwarding. • *pattern*—Digits to match for call forwarding using the H.450.3 standard. If an incoming calling-party number matches the pattern, it can be forwarded using the H.450.3 standard. A pattern of .T forwards all calling parties using the H.450.3 standard.

	Command or Action	Purpose

Step 3	**transfer-system** {**blind** | **full-blind** | **full-consult** | **local-consult**}
	Example:
	`Router(config-cm-fallback)# transfer-system full-consult`
		Defines the call-transfer method for all lines served by the Cisco SRST router.
		• **blind**—Calls are transferred without consultation with a single phone line using the Cisco proprietary method.
		Note: The keyword **blind** is not recommended. Use either the **full-blind** or **full-consult** keyword instead.
		• **full-blind**—Calls are transferred without consultation using H.450.2 standard methods.
		• **full-consult**—Calls are transferred with consultation using a second phone line if available. The calls fall back to **full-blind** if the second line is unavailable.
		• **local-consult**—Calls are transferred with local consultation using a second phone line if available. The calls fall back to **blind** for nonlocal consultation or nonlocal transfer target.

Step 4	**transfer-pattern** *transfer-pattern* **Example:** `Router(config-cm-fallback)# transfer-pattern 52540..`	Allows transfer of telephone calls by Cisco IP phones to specified phone number patterns. • *transfer-pattern*—String of digits for permitted call transfers. Wildcards are allowed.
Step 5	**exit** **Example:** `Router(config-cm-fallback)# exit`	Exits call-manager-fallback configuration mode. **Timesaver** Before exiting call-manager-fallback configuration mode, configure any other parameters that you need to set for the entire Cisco SRST phone network.
Step 6	**voice service voip** **Example:** `Router(config)# voice service voip`	(Optional) Enters voice service configuration mode.
Step 7	**h323** **Example:** `Router(conf-voi-serv)# h323`	(Optional) Enters H.323 voice service configuration mode.

	Command or Action	Purpose
Step 8	**h450 h450-2 timeout** {**T1** | **T2** | **T3** | **T4**} *milliseconds* **Example:** `Router(conf-serv-h323)# h450 h450-2 timeout T1 750`	(Optional) Sets timeouts for supplementary service timers, in milliseconds. This command is used primarily when the default settings for these timers do not match your network delay parameters. See the ITU-T H.450.2 specification for more information on these timers. • **T1**—Timeout value to wait to identify a response. Default is 2000. • **T2**—Timeout value to wait for call setup. Default is 5000. • **T3**—Timeout value to wait to initiate a response. Default is 5000. • **T4**—Timeout value to wait for setup of a response. Default is 5000. • *milliseconds*—Number of milliseconds. Range is from 500 to 60000.

Step 9	**end**
	Example:
	`Router(conf-serv-h323)# end`
		(Optional) Returns to privileged EXEC mode.

Examples

The following example specifies transfer with consultation using the H.450.2 standard for all IP phones serviced by the Cisco SRST router:

```
dial-peer voice 100 pots
 destination-pattern 9.T
 port 1/0/0

dial-peer voice 4000 voip
 destination-pattern 4…
 session-target ipv4:10.1.1.1

call-manager-fallback
 transfer-pattern 4…
 transfer-system full-consult
```

The following example enables call forwarding using the H.450.3 standard:

```
dial-peer voice 100 pots
 destination-pattern 9.T
 port 1/0/0
!
dial-peer voice 4000 voip
 destination-pattern 4
 session-target ipv4:10.1.1.1
!
call-manager-fallback
 call-forward pattern 4
```

Enabling Analog Transfer Using Hookflash and the H.450.2 Standard with Cisco SRST V3.0 or Lower

Analog call transfer using hookflash and the H.450.2 standard allows analog phones to transfer calls with consultation by using the hookflash to initiate the transfer. Hookflash refers to the short on-hook period usually generated by a telephone-like device during a call to indicate that the telephone is attempting to perform a dial-tone recall from a PBX. Hookflash is often used to perform call transfer. For example, a hookflash occurs when a caller quickly taps once on the button in the cradle of an analog phone's handset.

This feature requires installation of a Tool Command Language (Tcl) script. The script app-h450-transfer.tcl must be downloaded from the Cisco Software Center at http://www.cisco.com/cgi-bin/tablebuild.pl/ip-iostsp and copied to a TFTP server that is available to the Cisco SRST router or copied to the flash memory on the Cisco SRST router. To apply this script globally to all dial peers, use the **call application global** command in global configuration mode. The Tcl script has parameters to which you can pass values using attribute-value (AV) pairs in the **call application voice** command. The parameter that applies to this feature is as follows:

- **delay-time**—Speeds up or delays the setting up of the consultation call during a call transfer from an analog phone using a delay timer. When all digits have been collected, the delay timer is started. The call setup to the receiving party does not begin until the delay timer expires. If the transferring party goes on-hook before the delay timer expires, the transfer is considered a blind transfer rather than a consultative transfer. If the transferring party goes on-hook after the delay timer expires, either while the destination phone is ringing or after the destination party answers, the transfer is considered a consultative transfer.

In addition to the Tcl script, a ReadMe file describes the script and the configurable AV pairs. Read this file whenever you download a new version of the script because it may contain additional script-specific information, such as configuration parameters and user interface descriptions.

Note For Cisco SRST Versions 3.1 and higher, call transfer using H.450.2 is supported automatically with the default session application.

Prerequisites

- The H.450 Tcl script named app-h450-transfer.tcl must be downloaded from the Cisco Software Center. The following versions of the script are available:
  - app-h450-transfer.2.0.0.2.tcl for Cisco IOS Release 12.2(11)YT1 and later releases
  - app-h450-transfer.2.0.0.1.tcl for Cisco IOS Release 12.2(11)YT
- All voice gateway routers in the VoIP network must support H.450 and be running the following software:
  - Cisco IOS 12.2(11)YT or a later release
  - Cisco SRST V3.0 or a lower version
  - Tcl IVR 2.0
  - H.450 Tcl script (app-h450-transfer.tcl)

Note You can continue to use the app-h450-transfer.2.0.0.1.tcl script if you install Cisco IOS Release 12.2(11)YT1 or later, but you cannot use the app-h450-transfer.2.0.0.2.tcl script with a release of Cisco IOS software that is earlier than Cisco IOS Release 12.2(11)YT1.

Restrictions

- When a consultative transfer is made by an analog FXS phone using hookflash, the consultation call itself cannot be further transferred (that is, it cannot become a recursive or chained transfer) until after the initial transfer operation has been completed and the transferee and transfer-to parties are connected. Once the initial call transfer operation has been completed and the transferee and transfer-to parties are now the only parties in the call, the transfer-to party may further transfer the call.
- Call transfer with consultation is not supported for Cisco ATA-186, Cisco ATA-188, and Cisco IP Conference Station 7935. Transfer attempts from these devices are executed as blind transfers.

SUMMARY STEPS

1. **call application voice** *application-name location*
2. **call application voice** *application-name* **language** *number language*
3. **call application voice** *application-name* **set-location** *language category location*
4. **call application voice** *application-name* **delay-time** *seconds*
5. **dial-peer voice** *number* **pots**
6. **application** *application-name*
7. **exit**
8. **dial-peer voice** *number* **voip**
9. **application** *application-name*
10. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**call application voice** *application-name location* **Example:** `Router(config)# call application voice transfer_app flash:app-h450-transfer.tcl`	Loads the Tcl script and specifies its application name. • *application-name*—User-defined name for the IVR application. This name does not have to match the script filename. • *location*—Script directory and filename in URL format. For example, flash memory (flash:*filename*), a TFTP (tftp://../*filename*) or an HTTP server (http://../*filename*) are valid locations.
Step 2	**call application voice** *application-name* **language** *number language* **Example:** `Router(config)# call application voice transfer_app language 1 en`	(Optional) Sets the language for dynamic prompts used by the application. • *application-name*—IVR application name that was assigned in Step 1. • *number*—Number that identifies the language used by the audio files for the IVR application. • *language*—Two-character code that specifies the language of the prompts. Valid entries are **en** (English—default), **sp** (Spanish), **ch** (Chinese), or **aa** (all).

	Command or Action	Purpose
Step 3	**call application voice** *application-name* **set-location** *language category location* **Example:** `Router(config)# call application voice transfer_app set-location en 0 flash:/prompts`	Defines the location and category of the audio files that are used by the application for dynamic prompts. • *application-name*—Name of the Tcl IVR application. • *language*—Two-character code to specify the language of the prompts. Valid entries are **en** (English—default), **sp** (Spanish), **ch** (Chinese), or **aa** (all). • *category*—Category group (0 to 4) for the audio files from this location. The value 0 means all categories. • *location*—URL of the directory that contains the language audio files used by the application, without filenames. Flash memory (flash) or a directory on a server (TFTP, HTTP, or RTSP) are all valid. Prompts are required for call transfer from analog FXS phones. No prompts are needed for call transfer from IP phones.
Step 4	**call application voice** *application-name* **delay-time** *seconds* **Example:** `Router(config)# call application voice transfer_app delay-time 1`	(Optional) Sets the delay time for consultation call setup for an analog phone that is making a call transfer using the H.450 application. This command passes a value to the Tcl script by using an attribute-value (AV) pair. • *seconds*—Number of seconds to delay call setup. Range is from 1 to 10. Default is 2. A delay of more than 2 seconds is generally noticeable to users. For more information about AV pairs and the Tcl script for H.450 call transfer and forwarding, see the ReadMe file that accompanies the script.
Step 5	**dial-peer voice** *number* **pots** **Example:** `Router(config)# dial-peer voice 25 pots`	Enters dial-peer configuration mode to configure a POTS dial peer.
Step 6	**application** *application-name* **Example:** `Router(config-dial-peer)# application transfer_app`	Loads the application named in Step 1 onto the dial peer.
Step 7	**exit** **Example:** `Router(config-dial-peer)# exit`	Exits dial-peer configuration mode. **Timesaver** Before exiting dial-peer configuration mode, configure any other dial-peer parameters that you need to set for this dial peer.
Step 8	**dial-peer voice** *number* **voip** **Example:** `Router(config)# dial-peer voice 29 voip`	Enters dial-peer configuration mode to configure a VoIP dial peer.

	Command or Action	Purpose
Step 9	**application** *application-name* **Example:** `Router(config-dial-peer)# application transfer_app`	Loads the application named in Step 1 onto the dial peer.
Step 10	**exit** **Example:** `Router(config-dial-peer)# exit`	Exits dial-peer configuration mode. **Timesaver** Before exiting dial-peer configuration mode, configure any other dial-peer parameters that you need to set for this dial peer.

Example

The following example enables the H.450 Tcl script for analog transfer using hookflash and sets a delay time of 1 second:

```
call application voice transfer_app flash:app-h450-transfer.tcl
call application voice transfer_app language 1 en
call application voice transfer_app set-location en 0 flash:/prompts
call application voice transfer_app delay-time 1
!
dial-peer voice 25 pots
 destination-pattern 9.T
 port 1/0/0
 application transfer_app
!
dial-peer voice 29 voip
 destination-pattern 4…
 session-target ipv4:10.1.10.1
 application transfer_app
```

Configuring Trunk Access Codes

Note Configure trunk access codes only if your normal network dial-plan configuration prevents you from configuring permanent POTS voice dial peers to provide trunk access for use during fallback. If you already have local PSTN ports configured with the appropriate access codes provided by dial peers (for example, dial 9 to select an FXO PSTN line), this configuration is not needed.

Trunk access codes provide IP phones with access to the PSTN during Cisco CallManger fallback by creating POTS voice dial peers that are active during Cisco CallManager fallback only. These temporary dial peers, which can be matched to voice ports (BRI, E&M, FXO, and PRI), allow Cisco IP phones access to trunk lines during Cisco CallManager mode. When Cisco SRST is active, all PSTN interfaces of the same type are treated as equivalent, and any port may be selected to place the outgoing PSTN call.

Trunk access codes are created using the **access-code** command.

SUMMARY STEPS

1. **call-manager-fallback**
2. **access-code** {{**fxo** | **e&m**} *dial-string* | {**bri** | **pri**} *dial-string* [**direct-inward-dial**]}
3. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**call-manager-fallback** **Example:** `Router(config)# call-manager-fallback`	Enters call-manager-fallback configuration mode.

Step 2	**access-code** {{**fxo** | **e&m**} *dial-string* | {**bri** | **pri**} *dial-string* [**direct-inward-dial**]}
	Example:
	`Router(config-cm-fallback)# access-code e&m 8`
		Configures trunk access codes for each type of line so that the Cisco IP phones can access the trunk lines only in Cisco CallManager fallback mode when the Cisco SRST is enabled.
		• **fxo**—Enables a Foreign Exchange Office (FXO) interface.
		• **e&m**—Enables an analog Ear and Mouth (E&M) interface.
		• *dial-string*—String of characters that sets up dial access codes for each specified line type by creating dial peers. The *dial-string* argument is used to set up temporary dial peers for each specified line type.
		• **bri**—Enables a BRI interface.
		• **pri**—Enables a PRI interface.
		• **direct-inward-dial**—(Optional) Enables Direct Inward Dialing (DID) on the POTS dial peer.

Step 3	**exit** **Example:** `Router(config-cm-fallback)# exit`	Exits call-manager-fallback configuration mode.

Example

The following example creates access code number 8 for BRI and enables DID on the POTS dial peer:

```
call-manager-fallback
 access-code bri 8 direct-inward-dial
```

Configuring Interdigit Timeout Values

Configuring interdigit timeout values involves specifying how long, in seconds, all Cisco IP phones attached to a Cisco SRST router are to wait after an initial digit or a subsequent digit is dialed. The **timeouts interdigit** timer is enabled when a caller enters a digit and is restarted each time the caller enters subsequent digits until the destination address is identified. If the configured timeout value is exceeded before the destination address is identified, a tone sounds and the call is terminated.

Note This value setting is important when using variable-length dial-peer destination patterns (dial plans). For more information on setting dial plans, see the "Configuration Dial Plans, Dial Peers, and Digit Manipulation" chapter of the *Cisco IOS Voice, Video, and Fax Configuration Guide*, Release 12.2.

SUMMARY STEPS

1. **call-manager-fallback**
2. **timeouts interdigit** *seconds*
3. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**call-manager-fallback** **Example:** `Router(config)# call-manager-fallback`	Enters call-manager-fallback configuration mode.
Step 2	**timeouts interdigit** *seconds* **Example:** `Router(config-cm-fallback)# timeouts interdigit 5`	(Optional) Configures the interdigit timeout value for all Cisco IP phones that are attached to the router. • *seconds*—Interdigit timeout duration, in seconds, for all Cisco IP phones. Valid entries are integers from 2 to 120.
Step 3	**exit** **Example:** `Router(config-cm-fallback)# exit`	Exits call-manager-fallback configuration mode.

Example

The following example sets the interdigit timeout value to 5 seconds for all Cisco IP phones. In this example, 5 seconds are the elapsed time after which an incompletely dialed number times out. For example, a caller who dials nine digits (408555010) instead of the required ten digits (4085550100) will hear a busy tone after the 5 timeout seconds have elapsed.

```
call-manager-fallback
 timeouts interdigit 5
```

Configuring Class of Restriction

The class of restriction (COR) functionality provides the ability to deny certain call attempts on the basis of the incoming and outgoing class of restrictions provisioned on the dial peers. This functionality provides flexibility in network design, allows users to block calls (for example, calls to 900 numbers), and applies different restrictions to call attempts from different originators. The **cor** command sets the dial-peer COR parameter for dial peers associated with the directory numbers created during CallManager fallback.

You can have up to 20 COR lists for each incoming and outgoing call. A default COR is assigned to directory numbers that do not match any COR list numbers or number ranges. An assigned COR is invoked for the dial peers and created for each directory number automatically during CallManager fallback registration.

If a COR is applied on an incoming dial peer (for incoming calls) and it is a superset of or is equal to the COR applied to the outgoing dial peer (for outgoing calls), the call will go through. Voice ports determine whether a call is considered incoming or outgoing. If you hook up a phone to an FXS port on a Cisco SRST router and try to make a call from that phone, the call will be considered an incoming call to the router and voice port. If you make a call to the FXS phone, the call will be considered outgoing.

By default, an incoming call leg has the highest COR priority; the outgoing call leg has the lowest priority. If there is no COR configuration for incoming calls on a dial peer, you can make a call from a phone attached to the dial peer, so that the call will go out of any dial peer regardless of the COR configuration on that dial peer. Table 6 describes call functionality based on how your COR lists are configured.

Table 6 Combinations of COR List and Results

COR List on Incoming Dial Peer	COR List on Outgoing Dial Peer	Result
No COR	No COR	Call will succeed.
No COR	COR list applied for outgoing calls	Call will succeed. By default, the incoming dial peer has the highest COR priority when no COR is applied. If you apply no COR for an incoming call leg to a dial peer, the dial peer can make a call out of any other dial peer regardless of the COR configuration on the outgoing dial peer.
COR list applied for incoming calls	No COR	Call will succeed. By default, the outgoing dial peer has the lowest priority. Because there are some COR configurations for incoming calls on the incoming or originating dial peer, it is a superset of the outgoing call's COR configuration for the outgoing or terminating dial peer.
COR list applied for incoming calls (superset of COR list applied for outgoing calls on the outgoing dial peer)	COR list applied for outgoing calls (subsets of COR list applied for incoming calls on the incoming dial peer)	Call will succeed. The COR list for incoming calls on the incoming dial peer is a superset of the COR list for outgoing calls on the outgoing dial peer.
COR list applied for incoming calls (subset of COR list applied for outgoing calls on the outgoing dial peer)	COR list applied for outgoing calls (supersets of COR list applied for incoming calls on the incoming dial peer)	Call will not succeed. The COR list for incoming calls on the incoming dial peer is not a superset of the COR list for outgoing calls on the outgoing dial peer.

SUMMARY STEPS

1. **call-manager-fallback**
2. **cor** {**incoming** | **outgoing**} *cor-list-name* {*cor-list-number starting-number* **-** *ending-number* | **default**}
3. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**call-manager-fallback** **Example:** `Router(config)# call-manager-fallback`	Enters call-manager-fallback configuration mode.
Step 2	**cor** {**incoming** | **outgoing**} *cor-list-name* [*cor-list-number starting-number* **-** *ending-number* | **default**] **Example:** `Router(config-cm-fallback)# cor outgoing LockforPhoneC 1 5010 - 5020`	Configures a COR on dial peers associated with directory numbers. • **incoming**—COR list to be used by incoming dial peers. • **outgoing**—COR list to be used by outgoing dial peers. • *cor-list-name*—COR list name. • *cor-list-number*—COR list identifier. The maximum number of COR lists that can be created is 20, comprised of incoming or outgoing dial peers. The first six COR lists are applied to a range of directory numbers. The directory numbers that do not have a COR configuration are assigned to the default COR list, providing a default COR list has been defined. • *starting-number* **-** *ending-number*—Directory number range; for example, 2000 - 2025. • **default**—Instructs the router to use an existing default COR list.
Step 3	**exit** **Example:** `Router(config-cm-fallback)# exit`	Exits call-manager-fallback configuration mode.

Examples

The following example shows how to set a dial-peer COR parameter for outgoing calls to the Cisco IP phone dial peers and directory numbers created during fallback:

```
call-manager-fallback
 cor outgoing LockforPhoneC 1 5010 - 5020
```

The following example shows how to set the dial-peer COR parameter for incoming calls to the Cisco IP phone dial peers and directory numbers in the default COR list:

```
call-manager-fallback
 cor incoming LockforPhoneC default
```

The following example shows how sub- and super-COR sets are created. First, a custom dial-peer COR is created with names declared under it:

```
dial-peer cor custom
 name 911
 name 1800
 name 1900
 name local_call
```

In the following configuration example, COR lists are created and applied to the dial peer.

```
dial-peer cor list call911
 member 911

dial-peer cor list call1800
 member 1800

dial-peer cor list call1900
 member 1900

dial-peer cor list calllocal
 member local_call

dial-peer cor list engineering
 member 911
 member local_call

dial-peer cor list manager
 member 911
 member 1800
 member 1900
 member local_call

dial-peer cor list hr
 member 911
 member 1800
 member local_call
```

In the example below, five dial peers are configured for destination numbers 734…., 1800……., 1900……., 316…., and 911. A COR list is applied to each of the dial peers.

```
dial-peer voice 1 voip
 destination pattern 734....
 session target ipv4:10.1.1.1
 cor outgoing calllocal

dial-peer voice 2 voip
 destination pattern 1800.......
 session target ipv4:10.1.1.1
 cor outgoing call1800

dial-peer voice 3 pots
 destination pattern 1900.......
 port 1/0/0
 cor outgoing call1900

dial-peer voice 5 pots
 destination pattern 316....
 port 1/1/0
! No COR is applied.

dial-peer voice 4 pots
 destination pattern 911
 port 1/0/1
 cor outgoing call911
```

Finally, the COR list is applied to the individual phone numbers.

```
call-manager-fallback
 max-conferences 8
 cor incoming engineering 1 1001 - 1001
 cor incoming hr 2 1002 - 1002
 cor incoming manager 3 1003 - 1008
```

The sample configuration allows for the following:

- Extension 1001 to call 734... numbers, 911, and 316....
- Extension 1002 to call 734..., 1800 numbers, 911, and 316....
- Extension 1003 through 1008 to call all of the possible Cisco SRST router numbers
- All extensions to call 316....

Call Blocking (Toll Bar) Based on Time of Day and Day of Week or Date

Call blocking to prevent unauthorized use of phones is implemented by matching a pattern of specified digits during a specified time of day and day of week or date. Up to 32 patterns of digits can be specified. Call blocking is supported on IP phones only and not on analog foreign exchange station (FXS) phones.

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, a fast busy signal is played for approximately 10 seconds. The call is then terminated, and the line is placed back in on-hook status.

In SRST (call-manager-fallback configuration) mode, there is no phone- or pin-based exemption to after-hours call blocking.

SUMMARY STEPS

1. **call-manager-fallback**
2. **after-hours block pattern** *tag pattern* [**7-24**]
3. **after-hours day** *day start-time stop-time*
4. **after-hours date** *month date start-time stop-time*
5. **exit**

DETAILED STEPS

	Command or Action	Purpose

Step 1	**`call-manager-fallback`**
	Example:
	`Router(config)# call-manager-fallback`
		Enters call-manager-fallback configuration mode.

Step 2	**`after-hours block pattern`** *`tag pattern`* `[`**`7-24`**`]`
	Example:
	`Router(config-cm-fallback)# after-hours block pattern 1 91900`
		Defines a pattern of outgoing digits to be blocked. Up to 32 patterns can be defined, using individual commands.
		• If the **7-24** keyword is specified, the pattern is always blocked, 7 days a week, 24 hours a day.
		• If the **7-24** keyword is not specified, the pattern is blocked during the days and dates that are defined using the **after-hours day** and **after-hours date** commands.

Step 3	**`after-hours day`** *`day start-time stop-time`*
	Example:
	`Router(config-cm-fallback)# after-hours day mon 19:00 7:00`
		Defines a recurring time period based on the day of the week during which calls are blocked to outgoing dial patterns that are defined using the **after-hours block pattern** command.
		• *day*—Day of the week abbreviation. The following are valid day abbreviations: **sun**, **mon**, **tue**, **wed**, **thu**, **fri**, **sat**.
		• *start-time stop-time*—Beginning and ending times for call blocking, in an HH:MM format using a 24-hour clock. If the stop time is a smaller value than the start time, the stop time occurs on the day following the start time. For example, “mon 19:00 07:00” means “from Monday at 7 p.m. until Tuesday at 7 a.m.”

Step 4	**`after-hours date`** *`month date start-time stop-time`*
	Example:
	`Router(config-cm-fallback)# after-hours date jan 1 0:00 0:00`
		Defines a recurring time period based on month and date during which calls are blocked to outgoing dial patterns that are defined using the **after-hours block pattern** command.
		• *month*—Month abbreviation. The following are valid month abbreviations: **jan**, **feb**, **mar**, **apr**, **may**, **jun**, **jul**, **aug**, **sep**, **oct**, **nov**, **dec**.
		• *date*—Date of the month. Range is from 1 to 31.
		• *start-time stop-time*—Beginning and ending times for call blocking, in an HH:MM format using a 24-hour clock. The stop time must be larger than the start time. The value 24:00 is not valid. If 00:00 is entered as an stop time, it is changed to 23:59. If 00:00 is entered for both start time and stop time, calls are blocked for the entire 24-hour period on the specified date.

Step 5	**`exit`**
	Example:
	`Router(config-cm-fallback)# exit`
		Exits call-manager-fallback configuration mode.

Example

The following example defines several patterns of digits for which 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., on Saturday before 7 a.m. and after 1 p.m., and all day Sunday. 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 block day mon 19:00 07:00
 after-hours block day tue 19:00 07:00
 after-hours block day wed 19:00 07:00
 after-hours block day thu 19:00 07:00
 after-hours block day fri 19:00 07:00
 after-hours block day sat 13:00 12:00
 after-hours block day sun 12:00 07:00
!
```

Where to Go Next

The next step is verifying whether you need to configure additional features available on Cisco SRST. For a description and configuration instructions, see the "Configuring Additional Call Features" chapter. If you need to configure security, see the "Setting Up Secure SRST" chapter, or if you need to configure voicemail, see the "Integrating Voice Mail with Cisco Unified SRST" chapter. If you do not need any of those features, go to the "Monitoring and Maintaining Cisco Unified SRST" chapter.

Configuring Additional Call Features

This chapter describe how to configure three-party G.711 ad hoc conferencing and music on hold (MOH) for Cisco Survivable Remote Site Telephony (SRST).

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

Information About Configuring Additional Call Features

Optional features available for configuration include three-party G.711 ad hoc conferencing and MOH. MOH is available from flash files on the Cisco SRST router and for G.711, on-net VoIP, and PSTN calls.

For information on configuring MOH from a live feed, see the *Configuring SRST MOH Live-Feed Support* section at *http://www.cisco.com/univercd/cc/td/doc/product/access/ip_ph/srs/srsinter/moh.htm.*

Also available is an eXtensible Markup Language (XML) application program interface (API). This interface supplies data from Cisco SRST to management software.

How to Configure Additional Call Features

This section contains the following tasks:

- Enabling Three-Party G.711 Ad Hoc Conferencing, page 92 (Optional)
- Configuring MOH for G.711 VoIP and PSTN Calls, page 93 (Optional)
- Configuring MOH from Flash Files, page 94 (Optional)
- Defining XML API Schema (Optional)

Enabling Three-Party G.711 Ad Hoc Conferencing

Enabling three-party G.711 ad hoc conferencing involves configuring the maximum number of simultaneous three-party conferences supported by the Cisco SRST router. For conferencing to be available, an IP phone must have a minimum of two lines connected to one or more buttons. See the "Configuring a Secondary Dial Tone" section on page 50.

SUMMARY STEPS

1. **call-manager-fallback**
2. **max-conferences** *max-conference-numbers*
3. **exit**

DETAILED STEPS

	Command or Action	Purpose

Step 1	`call-manager-fallback`
	Example:
	`Router(config)# call-manager-fallback`
		Enters call-manager-fallback configuration mode.

Step 2	`max-conferences` *`max-conference-numbers`*
	Example:
	`Router(config-cm-fallback)# max-conferences 16`
		Sets the maximum number of simultaneous three-party conferences supported by the router. The maximum number possible is platform dependent:
		• Cisco 1751 router—8
		• Cisco 1760 router—8
		• Cisco 2600 series routers—8
		• Cisco 2600-XM series routers—8
		• Cisco 2801 router—8
		• Cisco 2811, Cisco 2821, and Cisco 2851 routers—16
		• Cisco 3640 and Cisco 3640A routers—8
		• Cisco 3660 router—16
		• Cisco 3725 router—16
		• Cisco 3745 router—16
		• Cisco 3800 series router—24

Step 3	`exit`
	Example:
	`Router(config-cm-fallback)# exit`
		Exits call-manager-fallback configuration mode.

Examples

The following example configures up to eight simultaneous three-way conferences on a router.

```
call-manager-fallback
 max-conferences 8
```

Configuring MOH for G.711 VoIP and PSTN Calls

MOH configuration works with G.711 VoIP and PSTN calls only. For all other calls, such as internal calls between Cisco IP phones, a tone is heard. The MOH file can be in .wav or .au file format. However, the file format must contain 8-bit 8-kHz data, such as a-law or u-law data format.

The **moh** command allows you to specify the .au and .wav format music files that are played to callers who have been put on hold.

Prerequisites

You can obtain .au files from the Technical Support Software Download site at http://www.cisco.com/cgi-bin/tablebuild.pl/ip-iostsp. Copy the music-on-hold.au file to the flash memory on your Cisco SRST router.

SUMMARY STEPS

1. **call-manager-fallback**
2. **moh** *filename*
3. **exit**

DETAILED STEPS

	Command or Action	Purpose

Step 1	**`call-manager-fallback`**
	Example:
	`Router(config)# call-manager-fallback`
		Enters call-manager-fallback configuration mode.

Step 2	**`moh`** *`filename`*
	Example:
	`Router(config-cm-fallback)# moh jazz.wav`
		Enables MOH during G.711, on-net VoIP, and PSTN calls.
		• *filename*—Filename of the music file.

Step 3	**`exit`**
	Example:
	`Router(config-cm-fallback)# exit`
		Exits call-manager-fallback configuration mode.

Example

The following example enables the playing of an audio file called classical.au on G.711, on-net VoIP, and PSTN calls:

```
call-manager-fallback
 moh classical.au
```

Configuring MOH from Flash Files

The MOH Multicast from Flash Files feature facilitates the continuous multicast of MOH audio feed from files in the flash memories of Cisco SRST branch office routers during Cisco CallManager fallback and normal Cisco CallManager service. Multicasting MOH from individual branch routers saves WAN bandwidth by eliminating the need to stream MOH audio from central offices to remote branches.

Configuration for this feature involves configuring Cisco SRST and Cisco CallManager to work together, which is described in *Integrating Cisco CallManager and Cisco SRST to Use Cisco SRST As a Multicast MOH Resource* at http://www.cisco.com/univercd/cc/td/doc/product/access/ip_ph/srs/srsinter/moh.htm.

The MOH Multicast from Flash Files feature can act as a backup mechanism to the MOH live feed feature. MOH live feed provides live feed MOH streams from an audio device connected to an E&M or FXO port to Cisco IP phones in SRST mode. Music from a live feed is from a fixed source and is continuously fed into the MOH playout buffer instead of being read from a flash file. See the *Configuring SRST MOH Live-Feed Support* section at http://www.cisco.com/univercd/cc/td/doc/product/access/ip_ph/srs/srsinter/moh.htm.

Defining XML API Schema

The Cisco IOS commands in this section allow you to specify parameters associated with the XML API. For more information, refer to the *XML Developer Guide for Cisco CME/SRST*.

SUMMARY STEPS

1. **call-manager-fallback**
2. **xmlschema** *schema-url*
3. **exit**

DETAILED STEPS

	Command or Action	Purpose

Step 1	**`call-manager-fallback`**
	Example:
	`Router(config)# call-manager-fallback`
		Enters call-manager-fallback configuration mode.

Step 2	**`xmlschema`** *`schema-url`*
	Example:
	`Router(config-cm-fallback)# xmlschema http://server2.example.com/ schema/schema1.xsd`
		Specifies the URL for an XML API schema to be used with this Cisco SRST system.
		• *schema-url*—Local or remote URL as defined in RFC 2396.

Step 3	**`exit`**
	Example:
	`Router(config-cm-fallback)# exit`
		Exits call-manager-fallback configuration mode.

Where to Go Next

If you need to configure security, see the "Setting Up Secure SRST" chapter, or if you need to configure voicemail, see the "Integrating Voice Mail with Cisco SRST" chapter. If you do not need any of those features, go to the "Monitoring and Maintaining Cisco SRST" chapter.

Setting Up Secure SRST

This chapter describes new SRST security features such as authentication, integrity, and media encryption.

Contents

Prerequisites for Setting Up Secure SRST

General

- Secure Cisco IP phones supported in secure SRST must have certificates installed and encryption enabled.
- The SRST router must have a certificate; a certificate can be generated by a third party or by the Cisco IOS certificate authority (CA). The Cisco IOS CA can run on the same gateway as SRST.
- Cisco CallManager 4.1(2) or later must be installed and must support security mode (authenticate and encryption mode).
- Certificate trust lists (CTLs) on Cisco CallManager must be enabled. For complete instructions, see the “Configuring Secure IP Telephony Calls” procedure in the *Media and Signaling Authentication and Encryption Feature for Cisco IOS MGCP Gateways* feature.
- Gateway routers that run secure SRST must support voice- and security-enabled Cisco IOS images (a “k9” cryptographic software image). The following two images are supported:
  - Advanced IP Services. This image includes a number of advanced security features.
  - Advanced Enterprise Services. This image includes full Cisco IOS software.

Public Key Infrastructure

- Set the clock, either manually or by using Network Time Protocol (NTP). Setting the clock ensures synchronicity with Cisco CallManager.
- Enable the IP HTTP server (Cisco IOS processor) with the **ip http server** command, if not already enabled. For more information on public key infrastructure (PKI) deployment, see the Cisco IOS Certificate Server feature.
- If the certificate server is part of your startup configuration, you may see the following messages during the boot procedure:

  ```
  % Failed to find Certificate Server's trustpoint at startup
  % Failed to find Certificate Server's cert.
  ```

  These messages are informational messages and indicate a temporary inability to configure the certificate server, because the startup configuration has not been fully parsed yet. The messages are useful for debugging, in case the startup configuration has been corrupted.

  You can verify the status of the certificate server after the boot procedure using the **show crypto pki server** command.

SRST

- Secure SRST services cannot be enrolled while SRST is active. Therefore disable SRST with the **no call-manager-fallback** command.

Supported Cisco IP Phones, Platforms, and Memory Requirements

- For a list of supported Cisco IP phones, routers, network modules, and codecs for secure SRST, see the *Media and Signaling Authentication and Encryption Feature for Cisco IOS MGCP Gateways* feature.
- For the most up-to-date information about the maximum number of Cisco IP phones, the maximum number of directory numbers (DNs) or virtual voice ports, and the memory requirements for Cisco 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

Restrictions for Setting Up Secure SRST

General

- Cryptographic software features (“k9”) are under export controls. This product contains cryptographic features and is subject to United States and local country laws governing import, export, transfer, and use. Delivery of Cisco cryptographic products does not imply third-party authority to import, export, distribute or use encryption. Importers, exporters, distributors and, users are responsible for compliance with U.S. and local country laws. By using this product you agree to comply with applicable laws and regulations. If you are unable to comply with U.S. and local laws, return this product immediately.

  A summary of U.S. laws governing Cisco cryptographic products may be found at:

  http://www.cisco.com/wwl/export/crypto/tool/

  If you require further assistance, please contact us by sending e-mail to export@cisco.com.

- When a Secure Real-Time Transport Protocol (SRTP) encrypted call is made between Cisco IP phone endpoints or from a Cisco IP phone to a gateway endpoint, a lock icon is displayed on the IP phones. The lock indicates security only for the IP leg of the call. Security of the PSTN leg is not implied.
- Secure SRST is supported only within the scope of a single router.

Not Supported in Secure SRST Mode

- Cisco CallManager versions prior to 4.1(2)
- Secure music on hold (MoH); MoH stays active, but reverts to non-secure.
- Secure transcoding or conferencing
- Secure H.323 or SIP
- Hot Standby Routing Protocol (HSRP)

Supported Calls in Secure SRST Mode

Only voice calls are supported in secure SRST mode. Specifically, the following voice calls are supported:

- Basic call
- Call transfer (consult and blind)
- Call forward (busy, no-answer, all)
- Shared line (IP phones)
- Hold and resume

Information About Setting Up Secure SRST

To configure secure SRST, you should understand the following concepts:

- Benefits of Secure SRST, page 99
- Cisco IP Phones Clear-Text Fallback During SRST, page 100
- SRST Routers and the TLS Protocol, page 100
- SRST Routers and PKI, page 101
- Secure SRST Authentication and Encryption, page 102
- Cisco IOS Credentials Server on Secure SRST Routers, page 103
- Establishment of Secure SRST to the Cisco IP Phone, page 103

Benefits of Secure SRST

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

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

SRST security is achieved when:

- End devices are authenticated using certificates.
- Signaling is authenticated and encrypted using Transport Layer Security (TLS) for TCP.
- A secure media path is encrypted using Secure Real-Time Transport Protocol (SRTP).
- Certificates are generated and distributed by a CA.

Cisco IP Phones Clear-Text Fallback During SRST

Cisco SRST versions prior to 12.3(14)T are not capable of supporting secure connections or have security enabled. If an SRST router is not capable of secure SRST as a fallback mode—that is, it is not capable of completing a TLS handshake with Cisco CallManager—its certificate is not added to the configuration file of the Cisco IP phone. The absence of an SRST router certificate causes the Cisco IP phone to use nonsecure (clear-text) communication when in SRST fallback mode. The capability to detect and fallback in clear-text mode is built into Cisco IP phone firmware. See the *Media and Signaling Authentication and Encryption Feature for Cisco IOS MGCP Gateways* for more information on clear-text mode.

SRST Routers and the TLS Protocol

Transport Layer Security (TLS) Version 1.0 provides secure TCP channels between Cisco IP phones, secure SRST routers, and Cisco CallManager. The TLS process begins with the Cisco IP phone establishing a TLS connection when registering with Cisco CallManager. Assuming that Cisco CallManager is configured to fallback to SRST, the TLS connection between the Cisco IP phones and the secure SRST router is also established. If the WAN link or Cisco CallManager fails, call control reverts to the SRST router.

SRST Routers and PKI

The transfer of certificates between an SRST router and Cisco CallManager is mandatory for secure SRST functionality. Public key infrastructure (PKI) commands are used to generate, import, and export the certificates for secure SRST. Table 7 shows the secure SRST supported Cisco IP phones and the appropriate certificate for each phone. The "Importing Phone Certificate Files in PEM Format to the Secure SRST Router" section on page 114 contains information and configurations about generating, importing, and exporting certificates that use PKI commands.

Table 7 Supported Cisco IP Phones and Certificates

Cisco IP Phone 7940	Cisco IP Phone 7960	Cisco IP Phone 7970
The phone receives locally significant certificate (LSC) from Certificate Authority Proxy Function (CAPF) in Distinguished Encoding Rules (DER) format. • 59fe77ccd.0 The filename may change based on the CAPF certificate subject name and the CAPF certificate issuer. If Cisco CallManager is using a third-party certificate provider, there can be multiple .0 files (from two to ten). Each .0 certificate file must be imported individually during the configuration. Manual enrollment supported only.	The phone receives locally significant certificate (LSC) from Certificate Authority Proxy Function (CAPF) in Distinguished Encoding Rules (DER) format. • 59fe77ccd.0 The filename may change based on the CAPF certificate subject name and the CAPF certificate issuer. If Cisco CallManager is using a third-party certificate provider, there can be multiple .0 files (from two to ten). Each .0 certificate file must be imported individually during the configuration. Manual enrollment supported only.	The phone contains a manufacturing installed certificate (MIC) used for device authentication. If the Cisco 7970 implements MIC, two public certificate files are needed: • CiscoCA.pem (Cisco Root CA, used to authenticate the certificate) • a69d2e04.0, in Privacy Enhanced Mail (PEM) format If Cisco CallManager is using a third-party certificate provider, there can be multiple .0 files (from two to ten). Each .0 certificate file must be imported individually during the configuration. Manual enrollment supported only.

Secure SRST Authentication and Encryption

Figure 4 illustrates the process of secure SRST authentication and encryption, and Table 8 describes the process.

Figure 4 Secure SRST Authentication and Encryption

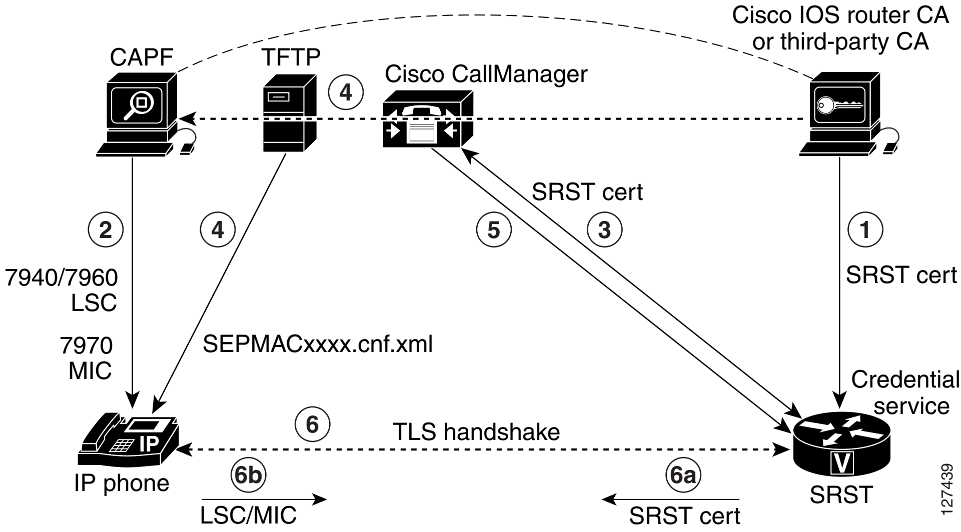


Table 8 Overview of the Process of Secure SRST Authentication and Encryption

Process Steps	Description or Detail
1.	The CA server, whether it is a Cisco IOS router CA or a third-party CA, issues a device certificate to the SRST gateway, enabling credentials service. Optionally, the certificate can be self-generated by the SRST router using a Cisco IOS CA server. The CA router is the ultimate trustpoint for the Certificate Authority Proxy Function (CAPF). For more information on CAPF, see the *Cisco CallManager Security Guide*.
2.	The CAPF is a process where supported devices can request a locally significant certificate (LSC). The CAPF utility generates a key pair and certificate that is specific for CAPF, copies this certificate to all Cisco CallManager servers in the cluster, and provides the LSC to the Cisco IP phone. An LSC is required for Cisco IP phones that do not have a manufacturing installed certificate (MIC). The Cisco 7970 is equipped with a MIC and therefore does not need to go through the CAPF process.
3.	Cisco CallManager requests the SRST certificate from credentials server, and the credentials server responds with the certificate.
4.	For each device, Cisco CallManager uses the TFTP process and inserts the certificate into the SEPMACxxxx.cnf.xml configuration file of the Cisco IP phone.
5.	Cisco CallManager provides the PEM format files that contain phone certificate information to the SRST router. Providing the PEM files to the SRST router is done manually; see SRST Routers and PKI, page 101 for more information. When the SRST router has the PEM files, the SRST router can authenticate the IP phone and validate the issuer of the IP phones certificate during the TLS handshake.

Table 8 Overview of the Process of Secure SRST Authentication and Encryption (continued)

Process Steps	Description or Detail
6.	The TLS handshake occurs, certificates are exchanged, and mutual authentication and registration occurs between the Cisco IP phone and the SRST router.
a.	The SRST router sends its certificate, and the phone validates the certificate to the certificate that it received from Cisco CallManager in Step 4.
b.	The Cisco IP phone provides the SRST router the LSC or MIC, and the router validates the LSC or MIC using the PEM format files that it was provided in Step 5.

Note The media is encrypted automatically once the phone and router certificates are exchanged and the TLS connection is established with the SRST router.

Cisco IOS Credentials Server on Secure SRST Routers

Secure SRST introduces a credentials server that runs on a secure SRST router. When the client, Cisco CallManager, requests a certificate through the TLS channel, the credentials server provides the SRST router certificate to Cisco CallManager. Cisco CallManager inserts the SRST router certificate in the Cisco IP phone configuration file and downloads the configuration files to the phones. The secure Cisco IP phone uses the certificate to authenticate the SRST router during fallback operations. The credentials service runs on default TCP port 2445.

Three Cisco IOS commands configure the credentials server in call-manager-fallback mode:

- **credentials**
- **ip source-address (credentials)**
- **trustpoint (credentials)**

Two Cisco IOS commands provide credential server debugging and verification capabilities:

- **debug credentials**
- **show credentials**

Establishment of Secure SRST to the Cisco IP Phone

Figure 5 and Table 9 show the interworking of the credentials server on the SRST router, Cisco CallManager, and the Cisco IP phone, and describe the establishment of secure SRST to the Cisco IP phone.

Figure 5 Interworking of Credentials Server on SRST Router, Cisco CallManager, and Cisco IP Phone

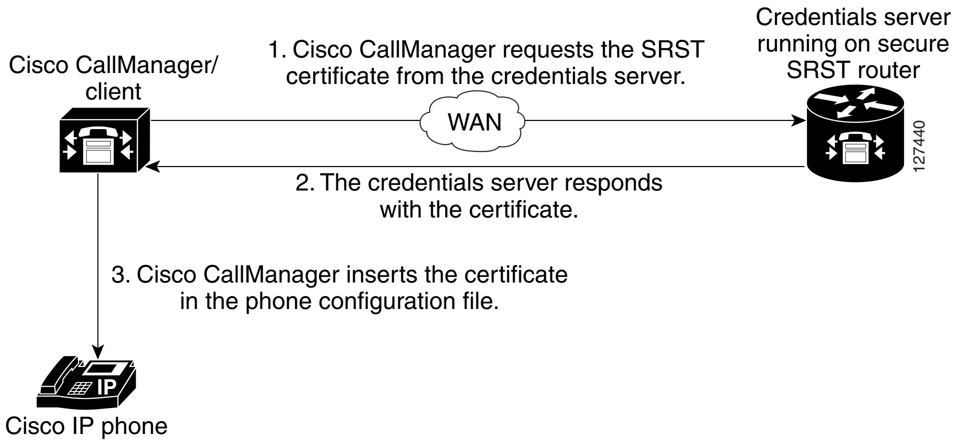

Cisco CallManager/ client
1. Cisco CallManager requests the SRST certificate from the credentials server.
WAN
Credentials server running on secure SRST router
2. The credentials server responds with the certificate.
127440
3. Cisco CallManager inserts the certificate in the phone configuration file.
IP
Cisco IP phone

Table 9 Establishing Secure SRST

Mode	Process	Description or Detail
Regular Mode	The Cisco IP phone configures DHCP and gets the TFTP server address.	—
	The Cisco IP phone retrieves a CTL file from the TFTP server.	The CTL file contains the certificates that the phone should trust.
	The Cisco IP phone opens a Transport Layer Security (TLS) protocol channel and registers to Cisco CallManager.	Cisco CallManager exports secure SRST router information and the SRST router certificate to the Cisco IP phone. The phone places the certificate into its configuration. Once the phone has the SRST certificate, the SRST router is considered secure. See Figure 5.
	If the Cisco IP phone is configured as “authenticated” or “encrypted” and Cisco CallManager is configured in mixed mode, the phone looks for an SRST certificate in its configuration file. If it finds an SRST certificate, it opens a standby TLS connection to the default port. The default port is the Cisco IP phone TCP port plus 443; that is, port 2443 on an SRST router.	The connection to the SRST router happens automatically, assuming there is not a secondary Cisco CallManager and SRST is configured as the backup device. See Figure 5. Cisco CallManager should be configured in mixed mode, which is its secure mode.
In case of WAN failure, the Cisco IP phone starts SRST registration.		
SRST Mode	The Cisco IP phone registers with the SRST router at the default port for secure communications.	—

How to Configure Secure SRST

The following configuration sections ensure that the secure SRST router and the Cisco IP phones can request mutual authentication during the TLS handshake. The TLS handshake occurs when the phone registers with the SRST router, either before or after the WAN link fails.

This section contains the following procedures:

- Preparing the SRST Router for Secure Communication, page 105 (required)
- Importing Phone Certificate Files in PEM Format to the Secure SRST Router, page 114 (required)
- Configuring Cisco CallManager to the Secure SRST Router, page 118 (required)
- Enabling SRST Mode on the Secure SRST Router, page 121 (required)
- Verifying Phone Status and Registrations, page 123 (required)

Preparing the SRST Router for Secure Communication

The following tasks prepare the SRST router to process secure communications.

- Configuring a Certificate Authority Server on a Cisco IOS Certificate Server, page 105 (optional)
- Autoenrolling and Authenticating the Secure SRST Router to the CA Server, page 107 (required)
- Disabling Automatic Certificate Enrollment, page 110 (required)
- Verifying Certificate Enrollment, page 111 (optional)
- Enabling Credentials Service on the Secure SRST Router, page 112 (required)
- Troubleshooting Credential Settings, page 113 (optional)

Configuring a Certificate Authority Server on a Cisco IOS Certificate Server

For SRST routers to provide secure communications, there must be a CA server that issues the device certificate in the network. The CA server can be a third-party CA or one generated from a Cisco IOS certificate server.

The Cisco IOS certificate server provides a certificate generation option to users who do not have a third-party CA in their network. The Cisco IOS certificate server can run on the SRST router or on a different Cisco IOS router.

If you do not have a third-party CA, full instructions on enabling and configuring a CA server can be found in the *Cisco IOS Certificate Server* documentation. A sample configuration is provided below.

SUMMARY STEPS

1. **crypto pki server** *cs-label*
2. **database level** {**minimal** | **names** | **complete**}
3. **database url** *root-url*
4. **issuer-name** *DN-string*
5. **grant auto**
6. **no shutdown**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**crypto pki server** *cs-label* **Example:** `Router (config)# crypto pki server srstcaserver`	Enables the certificate server and enters certificate server configuration mode. **Note** If you manually generated an RSA key pair, the *cs-label* argument must match the name of the key pair. For more information on the certificate server, see the *Cisco IOS Certificate Server* documentation.

Step 2	**database level** {**minimal** | **names** | **complete**}
	Example:
	`Router (cs-server)# database level complete`
		Controls what type of data is stored in the certificate enrollment database.
		• **minimal**—Enough information is stored only to continue issuing new certificates without conflict; this is the default.
		• **names**—In addition to the information given in the minimal level, the serial number and subject name of each certificate are stored.
		• **complete**—In addition to the information given in the minimal and names levels, each issued certificate is written to the database.
		Note The **complete** keyword produces a large amount of information; if it is issued, you should also specify an external TFTP server on which to store the data via the **database url** command.

Step 3	**database url** *root-url* **Example:** `Router (cs-server)# database url nvram`	Specifies the location where all database entries for the certificate server will be written. After you create a certificate server via the **crypto pki server** command, use this command to specify a combined list of all the certificates that have been issued. The *root-url* argument specifies the location where database entries are written. • The default location for the database entries to be written is flash; however, NVRAM is recommended for this task.
Step 4	**issuer-name** *DN-string* **Example:** `Router (cs-server)# issuer-name CN=srstcaserver`	Sets the CA issuer name to the specified distinguished name (DN-string). The default value is as follows: **issuer-name CN**=*cs-label*.

	Command or Action	Purpose
Step 5	**grant auto** **Example:** `Router (cs-server)# grant auto`	Allows an automatic certificate to be issued to any requestor. • This command is used only during enrollment and will be removed in the "Disabling Automatic Certificate Enrollment" section on page 110.
Step 6	**no shutdown** **Example:** `Router (cs-server)# no shutdown`	Enables the Cisco IOS certificate server. • You should issue this command only after you have completely configured your certificate server.

Examples

The following example reflects one way of generating a CA.

```
Router(config)# crypto pki server srstcaserver
Router(cs-server)# database level complete
Router(cs-server)# database url nvram
Router(cs-server)# issuer-name CN=srstcaserver
Router(cs-server)# grant auto

% This will cause all certificate requests to be automatically granted.
Are you sure you want to do this? [yes/no]: y
Router(cs-server)# no shutdown
% Once you start the server, you can no longer change some of
% the configuration.
Are you sure you want to do this? [yes/no]: y
% Generating 1024 bit RSA keys ...[OK]
% Certificate Server enabled.
```

Autoenrolling and Authenticating the Secure SRST Router to the CA Server

The secure SRST router needs to define a trustpoint; that is, it must obtain a device certificate from the CA server. The procedure is called certificate enrollment. Once enrolled, the secure SRST router can be recognized by Cisco CallManager as a secure SRST router.

There are three options to enroll the secure SRST router to a CA server: autoenrollment, cut and paste, and TFTP. When the CA server is a Cisco IOS certificate server, autoenrollment can be used. Otherwise, manual enrollment is required. Manual enrollment refers to cut and paste or TFTP.

Use the **enrollment url** command for autoenrollment and the **crypto pki authenticate** command to authenticate the SRST router. Full instructions for the commands can be found in the *Certification Authority Interoperability Commands* documentation. An example of autoenrollment is available in the *Certificate Enrollment Enhancements* feature. A sample configuration is provided below.

SUMMARY STEPS

1. **crypto pki trustpoint** *name*
2. **enrollment url** *url*
3. **revocation-check** *method1*
4. **exit**
5. **crypto pki authenticate** *name*
6. **crypto pki enroll** *name*

DETAILED STEPS

	Command or Action	Purpose
Step 1	**crypto pki trustpoint** *name* **Example:** `Router(config)# crypto pki trustpoint srstca`	Declares the CA that your router should use and enters ca-trustpoint configuration mode. • The name provided will be the same as the trustpoint name that will be declared in the "Enabling Credentials Service on the Secure SRST Router" section on page 112.
Step 2	**enrollment url** *url* **Example:** `Router(ca-trustpoint)# enrollment url http://10.1.1.22`	Specifies the enrollment parameters of your CA. • **url** *url*—Specifies the URL of the CA to which your router should send certificate requests. • If you are using Cisco proprietary SCEP for enrollment, *url* must be in the form http://*CA_name*, where *CA_name* is the host Domain Name System (DNS) name or IP address of the Cisco IOS CA. • If you used the procedure documented in the "Configuring a Certificate Authority Server on a Cisco IOS Certificate Server" section on page 105, the URL is the IP address of the certificate server router configured in Step 1. If a third-party CA was used, the IP address is to an external CA.
Step 3	**revocation-check** *method1* **Example:** `Router(ca-trustpoint)# revocation-check none`	Checks the revocation status of a certificate. The argument *method1* is the method used by the router to check the revocation status of the certificate. For this task, the only available method is **none.** The keyword **none** means that a revocation check will not be performed and the certificate will always be accepted. • Using the **none** keyword is mandatory for this task.
Step 4	**exit** **Example:** `Router(ca-trustpoint)# exit`	Exits ca-trustpoint configuration mode and returns to global configuration mode.

	Command or Action	Purpose
Step 5	**crypto pki authenticate** *name* **Example:** `Router(config)# crypto pki authenticate srstca`	Authenticates the CA (by getting the certificate from the CA). - Takes the name of the CA as the argument.
Step 6	**crypto pki enroll** *name* **Example:** `Router(config)# crypto pki enroll srstca`	Obtains the SRST router certificate from the CA. - Takes the name of the CA as the argument.

Examples

The following example autoenrolls and authenticates the SRST router.

```
Router(config)# crypto pki trustpoint srstca
Router(ca-trustpoint)# enrollment url http://10.1.1.22
Router(ca-trustpoint)# revocation-check none
Router(ca-trustpoint)# exit
Router(config)# crypto pki authenticate srstca

Certificate has the following attributes:
Fingerprint MD5: 4C894B7D 71DBA53F 50C65FD7 75DDBFCA
Fingerprint SHA1: 5C3B6B9E EFA40927 9DF6A826 58DA618A BF39F291
% Do you accept this certificate? [yes/no]: y
Trustpoint CA certificate accepted.

Router(config)# crypto pki enroll srstca
%
% Start certificate enrollment ..
% Create a challenge password. You will need to verbally provide this
password to the CA Administrator in order to revoke your certificate.
For security reasons your password will not be saved in the configuration.
Please make a note of it.
Password:
Re-enter password:

% The fully-qualified domain name in the certificate will be: router.cisco.com
% The subject name in the certificate will be: router.cisco.com
% Include the router serial number in the subject name? [yes/no]: y
% The serial number in the certificate will be: D0B9E79C
% Include an IP address in the subject name? [no]: n
Request certificate from CA? [yes/no]: y
% Certificate request sent to Certificate Authority
% The certificate request fingerprint will be displayed.
% The 'show crypto pki certificate' command will also show the fingerprint.


Sep 29 00:41:55.427: CRYPTO_PKI: Certificate Request Fingerprint MD5: D154FB75
2524A24D 3D1F5C2B 46A7B9E4
Sep 29 00:41:55.427: CRYPTO_PKI: Certificate Request Fingerprint SHA1: 0573FBB2
98CD1AD0 F37D591A C595252D A17523C1
Sep 29 00:41:57.339: %PKI-6-CERTRET: Certificate received from Certificate Authority
```

Disabling Automatic Certificate Enrollment

The command **grant auto** allows certificates to be issued and was activated in the optional task documented in the "Configuring a Certificate Authority Server on a Cisco IOS Certificate Server" section on page 105.

Note A security best practice is to disable the **grant auto** command so that certificates cannot be continually granted.

SUMMARY STEPS

1. **crypto pki server** *cs-label*
2. **shutdown**
3. **no grant auto**
4. **no shutdown**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**crypto pki server** *cs-label* **Example:** `Router (config)# crypto pki server srstcaserver`	Enables the certificate server and enters certificate server configuration mode. **Note** If you manually generated an RSA key pair, the *cs-label* argument must match the name of the key pair.
Step 2	**shutdown** **Example:** `Router (cs-server)# shutdown`	Disables the Cisco IOS certificate server.
Step 3	**no grant auto** **Example:** `Router (cs-server)# no grant auto`	Disables automatic certificates to be issued to any requestor. • This command was for use during enrollment only and thus needs to be removed in this task.
Step 4	**no shutdown** **Example:** `Router (cs-server)# no shutdown`	Enables the Cisco IOS certificate server. • You should issue this command only after you have completely configured your certificate server.

What to Do Next

For manual enrollment instructions, see the *Manual Certificate Enrollment (TFTP and Cut-and-Paste)* feature.

Verifying Certificate Enrollment

If you used the Cisco IOS certificate server as your CA, use the **show running-config** command to verify certificate enrollment or the **show crypto pki server** command to verify the status of the CA server.

SUMMARY STEPS

1. **show running-config**
2. **show crypto pki server**

DETAILED STEPS

Step 1 **show running-config**

Use the **show running-config** command to verify the creation of the CA server (01) and device (02) certificates. This example shows the enrolled certificates.

```
Router# show running-config
.
.
.
! SRST router device certificate.
crypto pki certificate chain srstca
 certificate 02
  308201AD 30820116 A0030201 02020102 300D0609 2A864886 F70D0101 04050030
  17311530 13060355 0403130C 73727374 63617365 72766572 301E170D 30343034
  31323139 35323233 5A170D30 35303431 32313935 3232335A 30343132 300F0603
  55040513 08443042 39453739 43301F06 092A8648 86F70D01 09021612 6A61736F
  32363931 2E636973 636F2E63 6F6D305C 300D0609 2A864886 F70D0101 01050003
  4B003048 024100D7 0CC354FB 5F7C1AE7 7A25C3F2 056E0485 22896D36 6CA70C19
  C98F9BAE AE9D1F9B D4BB7A67 F3251174 193BB1A3 12946123 E5C1CCD7 A23E6155
  FA2ED743 3FB8B902 03010001 A330302E 300B0603 551D0F04 04030205 A0301F06
  03551D23 04183016 8014F829 CE97AD60 18D05467 FC293963 C2470691 F9BD300D
  06092A86 4886F70D 01010405 00038181 007EB48E CAE9E1B3 D1E7A185 D7F0D565
  CB84B17B 1151BD78 B3E39763 59EC650E 49371F6D 99CBD267 EB8ADF9D 9E43A5F2
  FB2B18A0 34AF6564 11239473 41478AFC A86E6DA1 AC518E0B 8657CEBB ED2BDE8E
  B586FE67 00C358D4 EFDD8D44 3F423141 C2D331D3 1EE43B6E 6CB29EE7 0B8C2752
  C3AF4A66 BD007348 D013000A EA3C206D CF
  quit
 certificate ca 01
  30820207 30820170 A0030201 02020101 300D0609 2A864886 F70D0101 04050030
  17311530 13060355 0403130C 73727374 63617365 72766572 301E170D 30343034
  31323139 34353136 5A170D30 37303431 32313934 3531365A 30173115 30130603
  55040313 0C737273 74636173 65727665 7230819F 300D0609 2A864886 F70D0101
  01050003 818D0030 81890281 8100C3AF EE1E4BB1 9922A8DA 2BB9DC8E 5B1BD332
  1051C9FE 32A971B3 3C336635 74691954 98E765B1 059E24B6 32154E99 105CA989
  9619993F CC72C525 7357EBAC E6335A32 2AAF9391 99325BFD 9B8355EB C10F8963
  9D8FC222 EE8AC831 71ACD3A7 4E918A8F D5775159 76FBF499 5AD0849D CAA41417
  DD866902 21E5DD03 C37D4B28 0FAB0203 010001A3 63306130 0F060355 1D130101
  FF040530 030101FF 300E0603 551D0F01 01FF0404 03020186 301D0603 551D0E04
  160414F8 29CE97AD 6018D054 67FC2939 63C24706 91F9BD30 1F060355 1D230418
  30168014 F829CE97 AD6018D0 5467FC29 3963C247 0691F9BD 300D0609 2A864886
  F70D0101 04050003 8181007A F71B25F9 73D74552 25DFD03A D8D1338F 6792C805
  47A81019 795B5AAE 035400BB F859DABF 21892B5B E71A8283 08950414 8633A8B2
  C98565A6 C09CA641 88661402 ACC424FD 36F23360 ABFF4C55 BB23C66A C80A3A57
  5EE85FF8 C1B1A540 E818CE6D 58131726 BB060974 4E1A2F4B E6195522 122457F3
  DEDBAAD7 3780136E B112A6
  quit
```

Step 2 **show crypto pki server**

Use the **show crypto pki server** command to verify the status of the CA server after a boot procedure.

```
Router# show crypto pki server

Certificate Server srstcaserver:
Status: enabled
Server's configuration is locked (enter "shut" to unlock it)
Issuer name: CN=srstcaserver
CA cert fingerprint: AC9919F5 CAFE0560 92B3478A CFF5EC00
Granting mode is: auto
Last certificate issued serial number: 0x2
CA certificate expiration timer: 13:46:57 PST Dec 1 2007
CRL NextUpdate timer: 14:54:57 PST Jan 19 2005
Current storage dir: nvram
Database Level: Complete - all issued certs written as <serialnum>.cer
```

Enabling Credentials Service on the Secure SRST Router

Once the SRST router has its own certificate, you need to provide Cisco CallManager the certificate. Enabling credentials service allows Cisco CallManager to retrieve the secure SRST device certificate and place it in the configuration file of the Cisco IP phone.

Activate credentials service on all SRST routers.

Note A security best practice is to protect the credentials service port using Control Plane Policing. Control Plane Policing protects the gateway and maintains packet forwarding and protocol states despite a heavy traffic load. For more information on control planes, see the *Control Plane Policing* documentation. In addition, a sample configuration is given in the "Control Plane Policing: Example" section on page 132.

SUMMARY STEPS

1. **credentials**
2. **ip source-address** *ip-address* [**port** *port*]
3. **trustpoint** *trustpoint-name*
4. **exit**

DETAILED STEPS

	Command or Action	Purpose

Step 1	**`credentials`**
	Example:
	`Router(config)# credentials`
		Provides the SRST router certificate to Cisco CallManager and enters credentials configuration mode.

Step 2	**`ip source-address`** *`ip-address`* `[`**`port`** *`port`*`]`
	Example:
	`Router(config-credentials)# ip source-address 10.1.1.22 port 2445`
		Enables the SRST router to receive messages from Cisco CallManager through the specified IP address and port.
		• *ip-address*—The IP address is the preexisting router IP address, typically one of the addresses of the Ethernet port of the router.
		• **port** *port*—(Optional) The port to which the gateway router connects to receive messages from Cisco CallManager. The port number is from 2000 to 9999. The default port number is 2445.

Step 3	**`trustpoint`** *`trustpoint-name`*
	Example:
	`Router(config-credentials)# trustpoint srstca`
		Specifies the name of the trustpoint that is to be associated with the SRST router certificate. The *trustpoint-name* argument is the name of the trustpoint and corresponds to the SRST device certificate.
		• The trustpoint name should be the same as the one declared in the "Autoenrolling and Authenticating the Secure SRST Router to the CA Server" section on page 107.

Step 4	**`exit`**
	Example:
	`Router(config-credentials)# exit`
		Exits credentials configuration mode.

Examples

```
Router(config)# credentials
Router(config-credentials)# ip source-address 10.1.1.22 port 2445
Router(config-credentials)# trustpoint srstca
Router(config-credentials)# exit
```

Troubleshooting Credential Settings

The following steps display credential settings or set debugging on the credential settings of the SRST router.

SUMMARY STEPS

1. **show credentials**
2. **debug credentials**

DETAILED STEPS

Step 1 **show credentials**

Use the **show credentials** command to display the credential settings on the SRST router that are supplied to Cisco CallManager for use during secure SRST fallback.

```
Router# show credentials

Credentials IP: 10.1.1.22
Credentials PORT: 2445
Trustpoint: srstca
```

Step 2 **debug credentials**

Use the **debug credentials** command to set debugging on the credential settings of the SRST router.

```
Router# debug credentials

Credentials server debugging is enabled
Router#
Sep 29 01:01:50.903: Credentials service: Start TLS Handshake 1 10.1.1.13 2187
Sep 29 01:01:50.903: Credentials service: TLS Handshake returns OPSSLReadWouldBlockErr
Sep 29 01:01:51.903: Credentials service: TLS Handshake returns OPSSLReadWouldBlockErr
Sep 29 01:01:52.907: Credentials service: TLS Handshake returns OPSSLReadWouldBlockErr
Sep 29 01:01:53.927: Credentials service: TLS Handshake completes.
```

Importing Phone Certificate Files in PEM Format to the Secure SRST Router

This task completes the provisioning tasks required of Cisco IP phones to authenticate secure SRST. The secure SRST router must retrieve phone certificates so that it can authenticate Cisco IP phones during the TLS handshake. Different certificates are used for different IP phones. Table 7 on page 101 lists the certificates needed for each type of phone.

You must manually import certificates from Cisco CallManager to the SRST router. The number of certificates depends on the Cisco CallManager configuration. Manual enrollment refers to cut and paste or TFTP. For manual enrollment instructions, see the *Manual Certificate Enrollment (TFTP and Cut-and-Paste)* feature. Repeat the enrollment procedure for each phone or PEM file.

Note To complete this task, copy and paste the Cisco CallManager certificates to the SRST router as directed. That is, after using the **crypto pki authenticate** command, you will receive a prompt. Open the .0 files with Windows Wordpad or Notepad, and copy and paste the contents to the SRST router console. Then, repeat the procedure with the .pem file. Copy all of the contents that appear between “-----BEGIN CERTIFICATE-----" and "-----END CERTIFICATE-----".

Certificates are located in Cisco CallManager in the following location: In the menu bar in Cisco CallManager, choose **Program Files > Cisco > Certificates**.

Note HTTP automatic enrollment from Cisco CallManager through a virtual web server is not yet supported.

SUMMARY STEPS

1. **crypto pki trustpoint** *name*
2. **revocation-check** *method1*
3. **enrollment terminal**
4. **exit**
5. **crypto pki authenticate** *name*

DETAILED STEPS

	Command or Action	Purpose

Step 1	`crypto pki trustpoint name`
	Example:
	`Router (config)# crypto pki trustpoint 7970`
		Declares the CA that your router should use and enters ca-trustpoint configuration mode.

Step 2	`revocation-check method1`
	Example:
	`Router(ca-trustpoint)# revocation-check none`
		Checks the revocation status of a certificate. The argument *method1* is the method used by the router to check the revocation status of the certificate. For this task, the only available method is **none.** The keyword **none** means that a revocation check will not be performed and the certificate will always be accepted.
		• Using the **none** keyword is mandatory for this task.

Step 3	`enrollment terminal`
	Example:
	`Router(ca-trustpoint)# enrollment terminal`
		Specifies manual cut-and-paste certificate enrollment.

Step 4	`exit`
	Example:
	`Router(ca-trustpoint)# exit`
		Exits ca-trustpoint configuration mode and returns to global configuration.

Step 5	`crypto pki authenticate name`
	Example:
	`Router(config)# crypto pki authenticate 7970`
		Authenticates the CA (by getting the certificate from the CA).
		• Takes the name of the CA as the argument.

Examples

The following example shows three certificates imported to the SRST router (7970, 7960, PEM).

```
Router(config)# crypto pki trustpoint 7970
Router(ca-trustpoint)# revocation-check none
Router(ca-trustpoint)# enrollment terminal
Router(ca-trustpoint)# exit
Router(config)# crypto pki authenticate 7970

Enter the base 64 encoded CA certificate.
End with a blank line or the word "quit" on a line by itself
MIIDqDCCApCgAwIBAgIQNT+yS9cPFKNGwfOprHJWdTANBgkqhkiG9w0BAQUFADAu
```

```
MRYwFAYDVQQKEw1DaXNjbyBTeXN0ZW1zMRQwEgYDVQQDEwtDQVAtUlRQLTAwMjAe
Fw0wMzEwMTAyMDE4NDlaFw0yMzEwMTAyMDI3MzdaMC4xFjAUBgNVBAoTDUNpc2Nv
IFN5c3RlbXMxFDASBgNVBAMTC0NBUC1SVFAtMDAyMIIBIDANBgkqhkiG9w0BAQEF
AAOCAQ0AMIIBCAKCAQEAxCZlBK19w/2NZVVvpjCPrpW1cCY7V1q9lhzI85RZZdnQ
2M4CufgIzNa3zYxGJIAYeFfcRECnMB3f5A+x7xNiEuzE87UPvK+7S80uWCY0Uhtl
AVVf5NQgZ3YDNoNXg5MmONb8lT86F55EZyVac0XGne77TSIbIdejrTgYQXGP2MJx
Qhg+ZQlGFDRzbHfM84Duv2Msez+l+SqmqO80kIckqE9Nr3/XCSj1hXZNNVg8D+mv
Hth2P6KZqAKXAAStGRLSZX3jNbS8tveJ3Gi5+sj9+F6KKK2PD0iDwHcRKkcUHb7g
lI++U/5nswjUDIAph715Ds2rn9ehkMGipGLF8kpuCwIBA6OBwzCBwDALBgNVHQ8E
BAMCAYYwDwYDVR0TAQH/BAUwAwEB/zAdBgNVHQ4EFgQUUpIr4ojuLgmKTn5wLFal
mrTUm5YwbwYDVR0fBGgwZjBkoGKgYIYtaHR0cDovL2NhcC1ydHAtMDAyL0NlcnRF
bnJvbGwvQ0FQLVJUUC0wMDIuY3Jshi9maWxlOi8vXFxjYXAtcnRwLTAwMlxDZXJ0
RW5yb2xsXENBUC1SVFAtMDAyLmNybDAQBgkrBgEEAYI3FQEEAwIBADANBgkqhkiG
9w0BAQUFAAOCAQEAVoOM78TaOtHqj7sVL/5u5VChlyvU168f0piJLNWip2vDRihm
E+DlXdwMS5JaqUtuaSd/m/xzxpcRJm4ZRRwPq6VeaiiQGkjFuZEe5jSKiSAK7eHg
tup4HP/ZfKSwPA40DlsGSYsKNMm3OmVOCQUMH02lPkS/eEQ9sIw6QS7uuHN4y4CJ
NPnRbpFRLw06hnStCZHtGpKEHnY213QOy3h/EWhbnp0MZ+hdr20FujSI6G1+L39l
aRjeD708f2fYoz9wnEpZbtn2Kzse3uhU1Ygq1D1x9yuPq388C18HWdmCj4OVTXux
V6Y47H1yv/GJM8FvdgvKlExbGTFnlHpPiaG9tQ==
quit
Certificate has the following attributes:
Fingerprint MD5: F7E150EA 5E6E3AC5 615FC696 66415C9F
Fingerprint SHA1: 1BE2B503 DC72EE28 0C0F6B18 798236D8 D3B18BE6
% Do you accept this certificate? [yes/no]: y
Trustpoint CA certificate accepted.
% Certificate successfully imported

Router(config)# crypto pki trustpoint 7960
Router(ca-trustpoint)# revocation-check none
Router(ca-trustpoint)# enrollment terminal
Router(ca-trustpoint)# exit
Router(config)# crypto pki authenticate 7960

Enter the base 64 encoded CA certificate.
End with a blank line or the word "quit" on a line by itself
MIICKDCCAZGgAwIBAgIC8wEwDQYJKoZIhvcNAQEFBQAwQDELMAkGA1UEBhMCVVMx
GjAYBgNVBAoTEUNpc2NvIFN5c3RlbXMgSW5jMRUwEwYDVQQDEwxDQVBGLTdEN0Qw
QzAwHhcNMDQwNzE1MjIzODMyWhcNMTkwNzEyMjIzODMxWjBAMQswCQYDVQQGEwJV
UzEaMBgGA1UEChMRQ2lzY28gU3lzdGVtcyBJbmMxFTATBgNVBAMTDENBUEYtN0Q3
RDBDMDCBnzANBgkqhkiG9w0BAQEFAAOBjQAwgYkCgYEA0hvMOZZ9ENYWme11YGY1
it2rvE3Nk/eqhnv8P9eqB1iqt+fFBeAG0WZ5bO5FetdU+BCmPnddvAeSpsfr3Z+h
x+r58fOEIBRHQLgnDZ+nwYH39uwXcRWWqWwlW147YHjV7M5c/R8T6daCx4B5NBo6
kdQdQNOrV3IP7kQaCShdM/kCAwEAAaMxMC8wDgYDVR0PAQH/BAQDAgKEMB0GA1Ud
JQQWMBQGCCsGAQUFBwMBBggrBgEFBQcDBTANBgkqhkiG9w0BAQUFAAOBgQCaNi6x
sL6M5NlDezpSBO3QmUVyXMfrONV2ysrSwcXzHu0gJ9MSJ8TwiQmVaJ47hSTlF5a8
YVYJ0IdifXbXRo+/EEO7kkmFE8MZta5rM7UWj8bAeR42iqA3RzQaDwuJgNWT9Fhh
GgfuNAlo5h1AikxsvxivmDlLdZyCMoqJJd7B2Q==
quit
Certificate has the following attributes:
Fingerprint MD5: 4B9636DF 0F3BA6B7 5F54BE72 24762DBC
Fingerprint SHA1: A9917775 F86BB37A 5C130ED2 3E528BB8 286E8C2D
% Do you accept this certificate? [yes/no]: y
Trustpoint CA certificate accepted.
% Certificate successfully imported

Router(config)# crypto pki trustpoint PEM
Router(ca-trustpoint)# revocation-check none
Router(ca-trustpoint)# enrollment terminal
Router(ca-trustpoint)# exit
Router(config)# crypto pki authenticate PEM

Enter the base 64 encoded CA certificate.
End with a blank line or the word "quit" on a line by itself
MIIDqDCCApCgAwIBAgIQdhL5YBU9b59OQiAgMrcjVjANBgkqhkiG9w0BAQUFADAu
```

```
MRYwFAYDVQQKEw1DaXNjbyBTeXN0ZW1zMRQwEgYDVQQDEwtDQVAtUlRQLTAwMTAe
Fw0wMzAyMDYyMzI3MTNaFw0yMzAyMDYyMzM2MzRaMC4xFjAUBgNVBAoTDUNpc2Nv
IFN5c3RlbXMxFDASBgNVBAMTC0NBUC1SVFAtMDAxMIIBIDANBgkqhkiG9w0BAQEF
AAOCAQ0AMIIBCAKCAQEArFW77Rjem4cJ/7yPLVCauDohwZZ/3qf0sJaWlLeAzBlq
Rj2lFlSij0ddkDtfEEo9VKmBOJsvx6xJlWJiuBwUMDhTRbsuJz+npkaGBXPOXJmN
Vd54qlpc/hQDfWlbrIFkCcYhHws7vwnPsLuy1Kw2L2cP0UXxYghSsx8H4vGqdPFQ
NnYy7aKJ43SvDFt4zn37n8jrvlRuz0x3mdbcBEdHbA825Yo7a8sk12tshMJ/YdMm
vny0pmDNZXmeHjqEgVO3UFUn6GVCO+K1y1dUU1qpYJNYtqLkqj7wgccGjsHdHr3a
U+bw1uLgSGsQnxMWeMaWo8+6hMxwlANPweufgZMaywIBA6OBwzCBwDALBgNVHQ8E
BAMCAYYwDwYDVR0TAQH/BAUwAwEB/zAdBgNVHQ4EFgQU6Rexgscfz6ypG270qSac
cK4FoJowbwYDVR0fBGgwZjBkoGKgYIYtaHR0cDovL2NhcC1ydHAtMDAxL0NlcnRF
bnJvbGwvQ0FQLVJUUC0wMDEuY3Jshi9maWxlOi8vXFxjYXAtcnRwLTAwMVxDZXJ0
RW5yb2xsXENBUC1SVFAtMDAxLmNybDAQBgkrBgEEAYI3FQEEAwIBADANBgkqhkiG
9w0BAQUFAAOCAQEAq2T96/YMMtw2Dw4QX+F1+g1XSrUCrNyjx7vtFaRDHyB+kobw
dwkpohfkzfTyYpJELzV1r+kMRoyuZ7oIqqccEroMDnnmeApc+BRGbDJqS1Zzk4OA
c6Ea7fm53nQRlcSPmUVLjDBzKYDNbnEjizptaIC5fgB/S9S6C1q0YpTZFn5tjUjy
WXzeYSXPrcxb0UH7IQJ1ogpONAAUKLoPaZU7tVDSH3hD4+VjmLyysaLUhksGFrrN
phzZrsVVilK17qpqCPllKLGAS4fSbkruq3r/6S/SpXS6/gAoljBKixP7ZW2PxgCU
1aU9cURLPO95NDOFN3jBk3Sips7cVidcogowPQ==
quit
Certificate has the following attributes:
Fingerprint MD5: 233C8E33 8632EA4E 76D79FEB FFB061C6
Fingerprint SHA1: F7B40B94 5831D2AB 447AB8F2 25990732 227631BE
% Do you accept this certificate? [yes/no]: y
Trustpoint CA certificate accepted.
% Certificate successfully imported
```

Use the **show crypto pki trustpoint status** command to show that enrollment has succeeded and that five CA certificates were granted. The five certificates include the three certificates just entered and the CA server certificate and the SRST router certificate.

```
Router# show crypto pki trustpoint status

Trustpoint 7970:
Issuing CA certificate configured:
Subject Name:
cn=CAP-RTP-002,o=Cisco Systems
Fingerprint MD5: F7E150EA 5E6E3AC5 615FC696 66415C9F
Fingerprint SHA1: 1BE2B503 DC72EE28 0C0F6B18 798236D8 D3B18BE6
State:
Keys generated ............. Yes (General Purpose)
Issuing CA authenticated ....... Yes
Certificate request(s) ..... None

Trustpoint 7960:
Issuing CA certificate configured:
Subject Name:
cn=CAPF-508A3754,o=Cisco Systems Inc,c=US
Fingerprint MD5: 6BAE18C2 0BCE391E DAE2FE4C 5810F576
Fingerprint SHA1: B7735A2E 3A5C274F C311D7F1 3BE89942 355102DE
State:
Keys generated ............. Yes (General Purpose)
Issuing CA authenticated ....... Yes
Certificate request(s) ..... None

Trustpoint PEM:
Issuing CA certificate configured:
Subject Name:
cn=CAP-RTP-001,o=Cisco Systems
Fingerprint MD5: 233C8E33 8632EA4E 76D79FEB FFB061C6
Fingerprint SHA1: F7B40B94 5831D2AB 447AB8F2 25990732 227631BE
State:
Keys generated ............. Yes (General Purpose)
```

```
Issuing CA authenticated ....... Yes
Certificate request(s) ..... None

Trustpoint srstcaserver:
Issuing CA certificate configured:
Subject Name:
cn=srstcaserver
Fingerprint MD5: 6AF5B084 79C93F2B 76CC8FE6 8781AF5E
Fingerprint SHA1: 47D30503 38FF1524 711448B4 9763FAF6 3A8E7DCF
State:
Keys generated ............. Yes (General Purpose)
Issuing CA authenticated ....... Yes
Certificate request(s) ..... None

Trustpoint srstca:
Issuing CA certificate configured:
Subject Name:
cn=srstcaserver
Fingerprint MD5: 6AF5B084 79C93F2B 76CC8FE6 8781AF5E
Fingerprint SHA1: 47D30503 38FF1524 711448B4 9763FAF6 3A8E7DCF
Router General Purpose certificate configured:
Subject Name:
serialNumber=F3246544+hostname=c2611XM-sSRST.cisco.com
Fingerprint: 35471295 1C907EC1 45B347BC 7A9C4B86
State:
Keys generated ............. Yes (General Purpose)
Issuing CA authenticated ....... Yes
Certificate request(s) ..... Yes
```

Configuring Cisco CallManager to the Secure SRST Router

The following tasks are performed in Cisco CallManager.

- Adding an SRST Reference to Cisco CallManager, page 118 (required)
- Configuring SRST Fallback on Cisco CallManager, page 119 (required)
- Configuring CAPF on Cisco CallManager, page 121 (required)

Adding an SRST Reference to Cisco CallManager

The following procedure describes how to add an SRST reference to Cisco CallManager.

Before following this procedure, verify that credentials service is running in the SRST router. Cisco CallManager connects to the SRST router for its device certificate. To enable credentials service, see the "Enabling Credentials Service on the Secure SRST Router" section on page 112.

For complete information on adding SRST to Cisco CallManager, see the "Survivable Remote Site Telephony Configuration" section of the *Cisco CallManager Administration Guide, Release 4.1(2)*.

SUMMARY STEPS

1. Choose **SRST** in the Cisco CallManager menu bar.
2. Add a new SRST reference.
3. Enter the appropriate settings in the SRST fields.
4. Click **Insert**.
5. Repeat Steps 2 through 4 for additional SRST references.

DETAILED STEPS

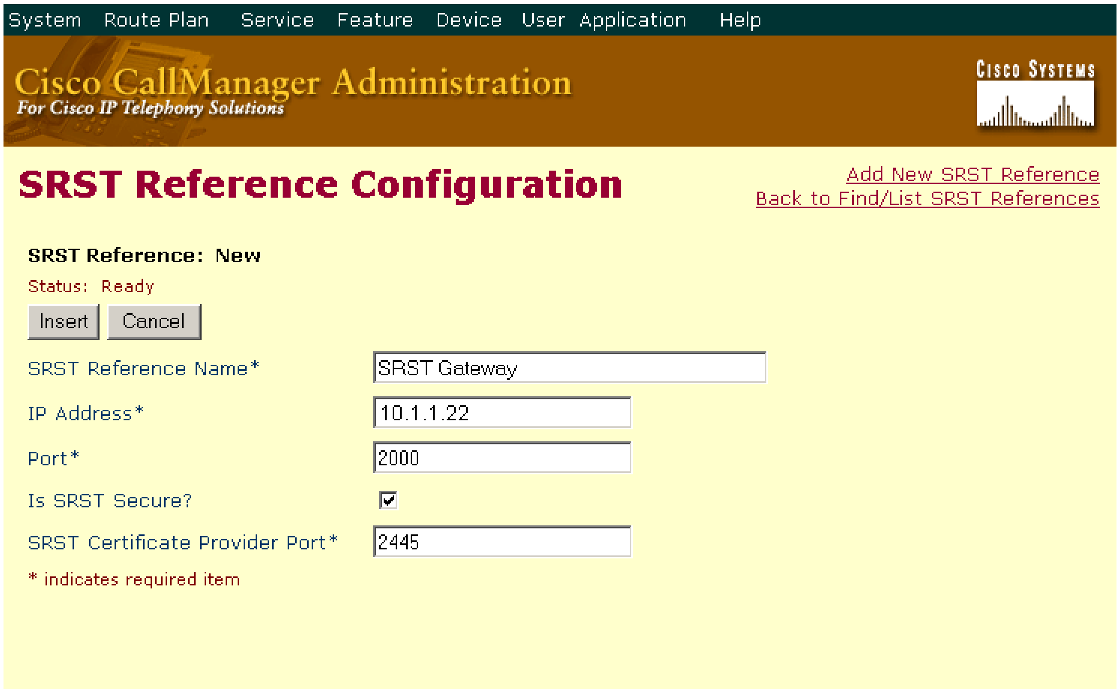
Step 1 In the menu bar in Cisco CallManager, choose **CCMAdmin > System > SRST**.

Step 2 Click **Add New SRST Reference**.

Step 3 Enter the appropriate settings. Figure 6 shows the available fields in the SRST Reference Configuration window.

- **a.** Enter the name of the SRST gateway, the IP address, and the port.
- **b.** Check the box asking if the SRST gateway is secure.
- **c.** Enter the certificate provider (credentials service) port number. Credentials service runs on default port 2445.

Figure 6 SRST Reference Configuration Window



SRST Reference Configuration

SRST Reference: New

Status: Ready

SRST Reference Name*	SRST Gateway
IP Address*	10.1.1.22
Port*	2000
Is SRST Secure?	☑
SRST Certificate Provider Port*	2445

* indicates required item

Step 4 To add the new SRST reference, click **Insert**. The message “Status: Insert completed” displays.

Step 5 To add more SRST references, repeat Steps 2 through 4.

Configuring SRST Fallback on Cisco CallManager

The following procedure describes how to configure SRST fallback on Cisco CallManager by assigning the device pool to SRST.

For complete information on adding a device pool to Cisco CallManager, see the “Device Pool Configuration” section of the *Cisco CallManager Administration Guide, Release 4.1(2)*.

SUMMARY STEPS

1. Choose **Device Pool** in the Cisco CallManager menu bar.
2. Add a device pool.
3. Click **Add New Device Pool.**
4. Enter the SRST reference.
5. Click **Update**.

DETAILED STEPS

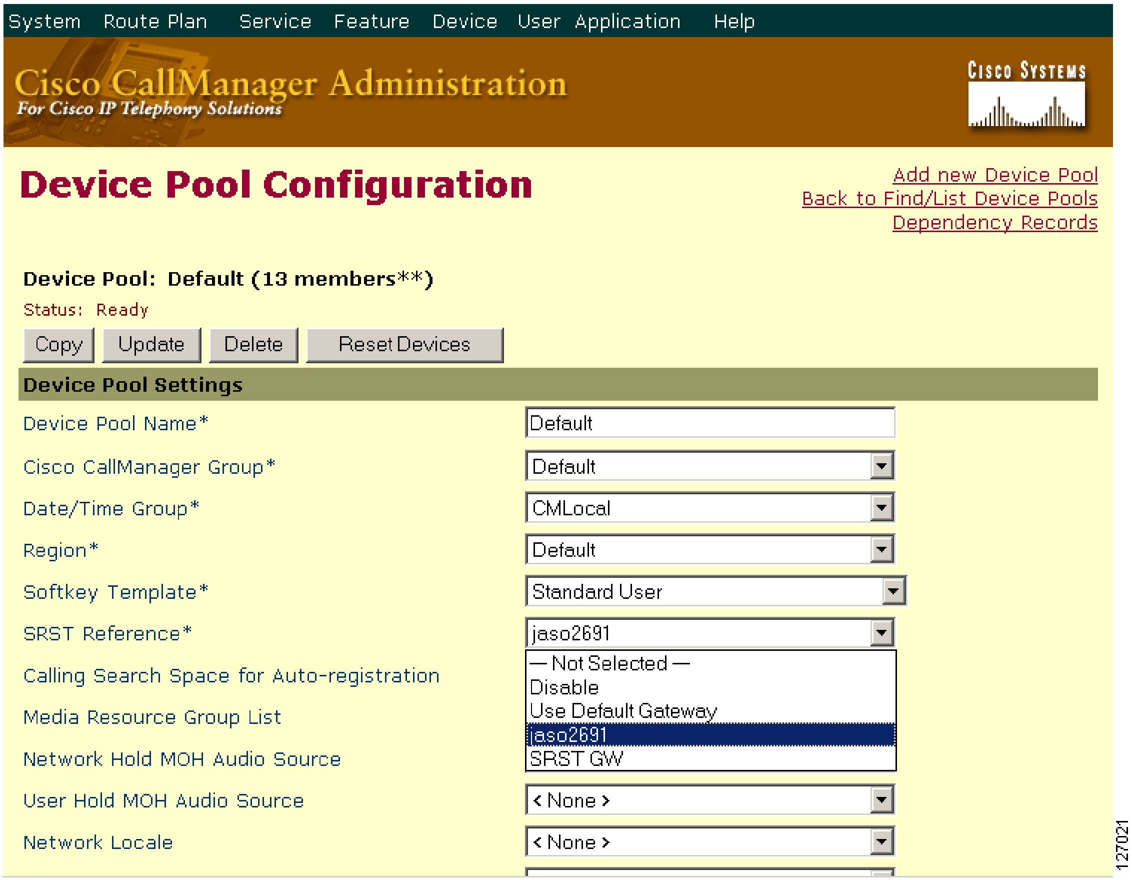
Step 1 In the menu bar in Cisco CallManager, choose **CCMAdmin > System > Device Pool**.

Step 2 Use one of the following methods to add a device pool:

- If a device pool already exists with settings that are similar to the one that you want to add, choose the existing device pool to display its settings, click **Copy**, and modify the settings as needed. Continue with Step 4.
- To add a device pool without copying an existing one, continue with Step 3.

Step 3 In the upper, right corner of the window, click the **Add New Device Pool** link. The Device Pool Configuration window displays (see Figure 7).

Figure 7 Device Pool Configuration Window



Device Pool Configuration

Device Pool: Default (13 members**)

Status: Ready

Device Pool Settings

Device Pool Name*	Default
Cisco CallManager Group*	Default
Date/Time Group*	CMLocal
Region*	Default
Softkey Template*	Standard User
SRST Reference*	jaso2691 — Not Selected — Disable Use Default Gateway jaso2691 SRST GW
Calling Search Space for Auto-registration	
Media Resource Group List	
Network Hold MOH Audio Source	
User Hold MOH Audio Source	< None >
Network Locale	< None >

Step 4 Enter the SRST reference.

Step 5 Click **Update** to save the device pool information in the database.

Configuring CAPF on Cisco CallManager

The Certificate Authority Proxy Function (CAPF) process allows supported devices, such as Cisco CallManager, to request LSC certificates from Cisco IP phones. The CAPF utility generates a key pair and certificate that are specific for CAPF, and the utility copies this certificate to all Cisco CallManager servers in the cluster.

For complete instructions on configuring CAPF in Cisco CallManager, see the *Cisco IP Phone Authentication and Encryption for Cisco CallManager* documentation.

Enabling SRST Mode on the Secure SRST Router

To configure secure SRST on the router to support the Cisco IP phone functions, use the following commands beginning in global configuration mode.

SUMMARY STEPS

1. **call-manager-fallback**
2. **secondary-dialtone** *digit-string*
3. **transfer-system** {**blind** | **full-blind** | **full-consult** | **local-consult**}
4. **ip source-address** *ip-address* [**port** *port*]
5. **max-ephones** *max-phones*
6. **max-dn** *max-directory-numbers*
7. **transfer-pattern** *transfer-pattern*
8. **exit**

DETAILED STEPS

	Command or Action	Purpose

Step 1	**`call-manager-fallback`**
	Example:
	`Router(config)# call-manager-fallback`
		Enters call-manager-fallback configuration mode.

Step 2	**`secondary-dialtone`** *`digit-string`*
	Example:
	`Router(config-cm-fallback)# secondary-dialtone 9`
		Activates a secondary dial tone when a digit string is dialed.

	Command or Action	Purpose

Step 3	**transfer-system** {**blind** | **full-blind** | **full-consult** | **local-consult**}
	Example:
	`Router(config-cm-fallback)# transfer-system full-consult`
		Defines the call-transfer method for all lines served by the Cisco SRST router.
		• **blind**—Calls are transferred without consultation with a single phone line using the Cisco proprietary method.
		• **full-blind**—Calls are transferred without consultation using H.450.2 standard methods.
		• **full-consult**—Calls are transferred with consultation using a second phone line if available. The calls fallback to **full-blind** if the second line is unavailable.
		• **local-consult**—Calls are transferred with local consultation using a second phone line if available. The calls fallback to **blind** for nonlocal consultation or nonlocal transfer target.

Step 4	**ip source-address** *ip-address* [**port** *port*] **Example:** `Router(config-cm-fallback)# ip source-address 10.1.1.22 port 2000`	Enables the router to receive messages from the Cisco IP phones through the specified IP addresses and provides for strict IP address verification. The default port number is 2000.
Step 5	**max-ephones** *max-phones* **Example:** `Router(config-cm-fallback)# max-ephones 15`	Configures the maximum number of Cisco IP phones that can be supported by the router. The maximum number is platform dependent. The default is 0. See the "Platform and Memory Support" section on page 24 for further details.
Step 6	**max-dn** *max-directory-numbers* **Example:** `Router(config-cm-fallback)# max-dn 30`	Sets the maximum number of directory numbers (DNs) or virtual voice ports that can be supported by the router. • *max-directory-numbers*—Maximum number of directory numbers or virtual voice ports supported by the router. The maximum number is platform dependent. The default is 0. See the "Platform and Memory Support" section on page 24 for further details.
Step 7	**transfer-pattern** *transfer-pattern* **Example:** `Router(config-cm-fallback)# transfer-pattern .....`	Allows transfer of telephone calls by Cisco IP phones to specified phone number patterns. • *transfer-pattern*—String of digits for permitted call transfers. Wildcards are allowed.
Step 8	**exit** **Example:** `Router(config-cm-fallback)# exit`	Exits call-manager-fallback configuration mode.

Examples

The following example enables SRST mode on your router.

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# secondary-dialtone 9
Router(config-cm-fallback)# transfer-system full-consult
Router(config-cm-fallback)# ip source-address 10.1.1.22 port 2000
Router(config-cm-fallback)# max-ephones 15
Router(config-cm-fallback)# max-dn 30
Router(config-cm-fallback)# transfer-pattern .....
Router(config-cm-fallback)# exit
```

Verifying Phone Status and Registrations

To verify or troubleshoot IP phone status and registration, complete the following steps beginning in privileged EXEC mode.

SUMMARY STEPS

1. **show ephone**
2. **show ephone offhook**
3. **show voice call status**
4. **debug ephone register**
5. **debug ephone state**

DETAILED STEPS

Step 1 **show ephone**

Use this command to display registered Cisco IP phones and their capabilities. The **show ephone** command also displays authentication and encryption status when used for secure SRST. In this example, authentication and encryption status is active with a TLS connection.

```
Router# show ephone

ephone-1 Mac:1000.1111.0002 TCP socket:[5] activeLine:0 REGISTERED in SCCP ver 5
+ Authentication + Encryption with TLS connection
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
IP:10.1.1.40 32626 7970 keepalive 390 max_line 8
button 1: dn 14 number 2002 CM Fallback CH1 IDLE

ephone-2 Mac:1000.1111.000B TCP socket:[12] activeLine:0 REGISTERED in SCCP ver
5 + Authentication + Encryption with TLS connection
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
IP:10.1.1.40 32718 7970 keepalive 390 max_line 8
button 1: dn 21 number 2011 CM Fallback CH1 IDLE

ephone-3 Mac:1000.1111.000A TCP socket:[16] activeLine:0 REGISTERED in SCCP ver
5 + Authentication + Encryption with TLS connection
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
IP:10.1.1.40 32862 7970 keepalive 390 max_line 8
button 1: dn 2 number 2010 CM Fallback CH1 IDLE
```

Step 2 **show ephone offhook**

Use this command to display Cisco IP phone status and quality for all phones that are off hook. In this example, authentication and encryption status is active with a TLS connection, and there is an active secure call.

```
Router# show ephone offhook

ephone-1 Mac:1000.1111.0002 TCP socket:[5] activeLine:1 REGISTERED in SCCP ver 5
+ Authentication + Encryption with TLS connection
mediaActive:1 offhook:1 ringing:0 reset:0 reset_sent:0 paging 0
:0
IP:10.1.1.40 32626 7970 keepalive 391 max_line 8
button 1: dn 14 number 2002 CM Fallback CH1 CONNECTED
Active Secure Call on DN 14 chan 1 :2002 10.1.1.40 29632 to 10.1.1.40 25616 via 10.1.1.40
G711Ulaw64k 160 bytes no vad
Tx Pkts 295 bytes 49468 Rx Pkts 277 bytes 46531 Lost 0
Jitter 0 Latency 0 callingDn 22 calledDn -1

ephone-2 Mac:1000.1111.000B TCP socket:[12] activeLine:1 REGISTERED in SCCP ver
5 + Authentication + Encryption with TLS connection
mediaActive:1 offhook:1 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
IP:10.1.1.40 32718 7970 keepalive 391 max_line 8
button 1: dn 21 number 2011 CM Fallback CH1 CONNECTED
Active Secure Call on DN 21 chan 1 :2011 10.1.1.40 16382 to 10.1.1.40 16382 via 10.1.1.40
G711Ulaw64k 160 bytes no vad
Tx Pkts 295 bytes 49468 Rx Pkts 277 bytes 46531 Lost 0
Jitter 0 Latency 0 callingDn -1 calledDn 11
```

Step 3 **show voice call status**

Use this command to show the call status for all voice ports on the Cisco SRST router. This command is not applicable for calls between two POTS dial peers.

```
Router# show voice call status

CallID CID ccVdb Port DSP/Ch Called # Codec Dial-peers
0x1164 2BFE 0x8619A460 50/0/35.0 2014 g711ulaw 20035/20027
0x1165 2BFE 0x86144B78 50/0/27.0 *2014 g711ulaw 20027/20035
0x1166 2C01 0x861043D8 50/0/21.0 2012 g711ulaw 20021/20011
0x1168 2C01 0x860984C4 50/0/11.0 *2012 g711ulaw 20011/20021
0x1167 2C04 0x8610EC7C 50/0/22.0 2002 g711ulaw 20022/20014
0x1169 2C04 0x860B8894 50/0/14.0 *2002 g711ulaw 20014/20022
0x116A 2C07 0x860A374C 50/0/12.0 2010 g711ulaw 20012/20002
0x116B 2C07 0x86039700 50/0/2.0 *2010 g711ulaw 20002/20012
0x116C 2C0A 0x86119520 50/0/23.0 2034 g711ulaw 20023/20020
0x116D 2C0A 0x860F9150 50/0/20.0 *2034 g711ulaw 20020/20023
0x116E 2C0D 0x8608DC20 50/0/10.0 2022 g711ulaw 20010/20008
0x116F 2C0D 0x86078AD8 50/0/8.0 *2022 g711ulaw 20008/20010
0x1170 2C10 0x861398F0 50/0/26.0 2016 g711ulaw 20026/20028
0x1171 2C10 0x8614F41C 50/0/28.0 *2016 g711ulaw 20028/20026
0x1172 2C13 0x86159CC0 50/0/29.0 2018 g711ulaw 20029/20004
0x1173 2C13 0x8604E848 50/0/4.0 *2018 g711ulaw 20004/20029
0x1174 2C16 0x8612F04C 50/0/25.0 2026 g711ulaw 20025/20030
0x1175 2C16 0x86164F48 50/0/30.0 *2026 g711ulaw 20030/20025
0x1176 2C19 0x860D8C64 50/0/17.0 2032 g711ulaw 20017/20018
0x1177 2C19 0x860E4008 50/0/18.0 *2032 g711ulaw 20018/20017
0x1178 2C1C 0x860CE3C0 50/0/16.0 2004 g711ulaw 20016/20019
0x1179 2C1C 0x860EE8AC 50/0/19.0 *2004 g711ulaw 20019/20016
0x117A 2C1F 0x86043FA4 50/0/3.0 2008 g711ulaw 20003/20024
0x117B 2C1F 0x861247A8 50/0/24.0 *2008 g711ulaw 20024/20003
0x117C 2C22 0x8608337C 50/0/9.0 2020 g711ulaw 20009/20031
0x117D 2C22 0x8616F7EC 50/0/31.0 *2020 g711ulaw 20031/20009
0x117E 2C25 0x86063990 50/0/6.0 2006 g711ulaw 20006/20001
```

```
0x117F 2C25 0x85C6BE6C 50/0/1.0 *2006 g711ulaw 20001/20006
0x1180 2C28 0x860ADFF0 50/0/13.0 2029 g711ulaw 20013/20034
0x1181 2C28 0x8618FBBC 50/0/34.0 *2029 g711ulaw 20034/20013
0x1182 2C2B 0x860C3B1C 50/0/15.0 2036 g711ulaw 20015/20005
0x1183 2C2B 0x860590EC 50/0/5.0 *2036 g711ulaw 20005/20015
0x1184 2C2E 0x8617A090 50/0/32.0 2024 g711ulaw 20032/20007
0x1185 2C2E 0x8606E234 50/0/7.0 *2024 g711ulaw 20007/20032
0x1186 2C31 0x861A56E8 50/0/36.0 2030 g711ulaw 20036/20033
0x1187 2C31 0x86185318 50/0/33.0 *2030 g711ulaw 20033/20036
18 active calls found
```

Step 4 **debug ephone register**

Use this command to debug the process of Cisco IP phone registration.

```
Router# debug ephone register

EPHONE registration debugging is enabled
*Jun 29 09:16:02.180: New Skinny socket accepted [2] (0 active)
*Jun 29 09:16:02.180: sin_family 2, sin_port 51617, in_addr 10.5.43.177
*Jun 29 09:16:02.180: skinny_socket_process: secure skinny sessions = 1
*Jun 29 09:16:02.180: add_skinny_secure_socket: pid =155, new_sock=0, ip address =
10.5.43.177
*Jun 29 09:16:02.180: skinny_secure_handshake: pid =155, sock=0, args->pid=155, ip address
= 10.5.43.177
*Jun 29 09:16:02.184: Start TLS Handshake 0 10.5.43.177 51617
*Jun 29 09:16:02.184: TLS Handshake retcode OPSSLReadWouldBlockErr
*Jun 29 09:16:03.188: TLS Handshake retcode OPSSLReadWouldBlockErr
*Jun 29 09:16:04.188: TLS Handshake retcode OPSSLReadWouldBlockErr
*Jun 29 09:16:05.188: TLS Handshake retcode OPSSLReadWouldBlockErr
*Jun 29 09:16:06.188: TLS Handshake retcode OPSSLReadWouldBlockErr
*Jun 29 09:16:07.188: TLS Handshake retcode OPSSLReadWouldBlockErr
*Jun 29 09:16:08.188: CRYPTO_PKI_OPSSL - Verifying 1 Certs

*Jun 29 09:16:08.212: TLS Handshake completes
```

Step 5 **debug ephone state**

Use this command to review call setup between two secure Cisco IP phones. The **debug ephone state** trace shows the generation and distribution of encryption and decryption keys between the two phones.

```
Router# debug ephone state

*Jan 11 18:33:09.231:%SYS-5-CONFIG_I:Configured from console by console
*Jan 11 18:33:11.747:ephone-2[2]:OFFHOOK
*Jan 11 18:33:11.747:ephone-2[2]:---SkinnySyncPhoneDnOverlays is onhook
*Jan 11 18:33:11.747:ephone-2[2]:SIEZE on activeLine 0 activeChan 1
*Jan 11 18:33:11.747:ephone-2[2]:SetCallState line 1 DN 2(-1) chan 1 ref 6 TsOffHook
*Jan 11 18:33:11.747:ephone-2[2]:Check Plar Number
*Jan 11 18:33:11.751:DN 2 chan 1 Voice_Mode
*Jan 11 18:33:11.751:dn_tone_control DN=2 chan 1 tonetype=33:DtInsideDialTone onoff=1
pid=232
*Jan 11 18:33:15.031:dn_tone_control DN=2 chan 1 tonetype=0:DtSilence onoff=0 pid=232
*Jan 11 18:33:16.039:ephone-2[2]:Skinny-to-Skinny call DN 2 chan 1 to DN 4 chan 1 instance
1
*Jan 11 18:33:16.039:ephone-2[2]:SetCallState line 1 DN 2(-1) chan 1 ref 6 TsProceed
*Jan 11 18:33:16.039:ephone-2[2]:SetCallState line 1 DN 2(-1) chan 1 ref 6 TsRingOut
*Jan 11 18:33:16.039:ephone-2[2]::callingNumber 6000

*Jan 11 18:33:16.039:ephone-2[2]::callingParty 6000

*Jan 11 18:33:16.039:ephone-2[2]:Call Info DN 2 line 1 ref 6 call state 1 called 6001
calling 6000 origcalled
```

```
*Jan 11 18:33:16.039:ephone-2[2]:Call Info DN 2 line 1 ref 6 called 6001 calling 6000
origcalled 6001 calltype 2
*Jan 11 18:33:16.039:ephone-2[2]:Call Info for chan 1
*Jan 11 18:33:16.039:ephone-2[2]:Original Called Name 6001
*Jan 11 18:33:16.039:ephone-2[2]:6000 calling
*Jan 11 18:33:16.039:ephone-2[2]:6001
*Jan 11 18:33:16.047:ephone-3[3]:SetCallState line 1 DN 4(4) chan 1 ref 7 TsRingIn
*Jan 11 18:33:16.047:ephone-3[3]::callingNumber 6000

*Jan 11 18:33:16.047:ephone-3[3]::callingParty 6000

*Jan 11 18:33:16.047:ephone-3[3]:Call Info DN 4 line 1 ref 7 call state 7 called 6001
calling 6000 origcalled
*Jan 11 18:33:16.047:ephone-3[3]:Call Info DN 4 line 1 ref 7 called 6001 calling 6000
origcalled 6001 calltype 1
*Jan 11 18:33:16.047:ephone-3[3]:Call Info for chan 1
*Jan 11 18:33:16.047:ephone-3[3]:Original Called Name 6001
*Jan 11 18:33:16.047:ephone-3[3]:6000 calling
*Jan 11 18:33:16.047:ephone-3[3]:6001
*Jan 11 18:33:16.047:ephone-3[3]:Ringer Inside Ring On
*Jan 11 18:33:16.051:dn_tone_control DN=2 chan 1 tonetype=36:DtAlertingTone onoff=1
pid=232
*Jan 11 18:33:20.831:ephone-3[3]:OFFHOOK
*Jan 11 18:33:20.831:ephone-3[3]:---SkinnySyncPhoneDnOverlays is onhook
*Jan 11 18:33:20.831:ephone-3[3]:Ringer Off
*Jan 11 18:33:20.831:ephone-3[3]:ANSWER call
*Jan 11 18:33:20.831:ephone-3[3]:SetCallState line 1 DN 4(-1) chan 1 ref 7 TsOffHook
*Jan 11 18:33:20.831:ephone-3[3][SEP000DEDAB3EBF]:Answer Incoming call from ephone-(2) DN
2 chan 1
*Jan 11 18:33:20.831:ephone-3[3]:SetCallState line 1 DN 4(-1) chan 1 ref 7 TsConnected
*Jan 11 18:33:20.831:defer_start for DN 2 chan 1 at CONNECTED
*Jan 11 18:33:20.831:ephone-2[2]:SetCallState line 1 DN 2(-1) chan 1 ref 6 TsConnected
*Jan 11 18:33:20.835:ephone-3[3]::callingNumber 6000

*Jan 11 18:33:20.835:ephone-3[3]::callingParty 6000

*Jan 11 18:33:20.835:ephone-3[3]:Call Info DN 4 line 1 ref 7 call state 4 called 6001
calling 6000 origcalled
*Jan 11 18:33:20.835:ephone-3[3]:Call Info DN 4 line 1 ref 7 called 6001 calling 6000
origcalled 6001 calltype 1
*Jan 11 18:33:20.835:ephone-3[3]:Call Info for chan 1
*Jan 11 18:33:20.835:ephone-3[3]:Original Called Name 6001
*Jan 11 18:33:20.835:ephone-3[3]:6000 calling
*Jan 11 18:33:20.835:ephone-3[3]:6001
*Jan 11 18:33:20.835:ephone-2[2]:Security Key Generation
! Ephone 2 generates a security key.

*Jan 11 18:33:20.835:ephone-2[2]:OpenReceive DN 2 chan 1 codec 4:G711Ulaw64k  duration 20
ms bytes 160
*Jan 11 18:33:20.835:ephone-2[2]:Send Decryption Key
! Ephone 2 sends the decryption key.

*Jan 11 18:33:20.835:ephone-3[3]:Security Key Generation
!Ephone 3 generates its security key.

*Jan 11 18:33:20.835:ephone-3[3]:OpenReceive DN 4 chan 1 codec 4:G711Ulaw64k  duration 20
ms bytes 160
*Jan 11 18:33:20.835:ephone-3[3]:Send Decryption Key
! Ephone 3 sends its decryption key.

*Jan 11 18:33:21.087:dn_tone_control DN=2 chan 1 tonetype=0:DtSilence onoff=0 pid=232
*Jan 11 18:33:21.087:DN 4 chan 1 Voice_Mode
*Jan 11 18:33:21.091:DN 2 chan 1 End Voice_Mode
*Jan 11 18:33:21.091:DN 2 chan 1 Voice_Mode
```

```
*Jan 11 18:33:21.095:ephone-2[2]:OpenReceiveChannelAck:IP 1.1.1.8, port=25552,
               dn_index=2, dn=2, chan=1
*Jan 11 18:33:21.095:ephone-3[3]:StartMedia 1.1.1.8 port=25552
*Jan 11 18:33:21.095:DN 2 chan 1 codec 4:G711Ulaw64k duration 20 ms bytes 160
*Jan 11 18:33:21.095:ephone-3[3]:Send Encryption Key
! Ephone 3 sends its encryption key.

*Jan 11 18:33:21.347:ephone-3[3]:OpenReceiveChannelAck:IP 1.1.1.9, port=17520,
               dn_index=4, dn=4, chan=1
*Jan 11 18:33:21.347:ephone-2[2]:StartMedia 1.1.1.9 port=17520
*Jan 11 18:33:21.347:DN 2 chan 1 codec 4:G711Ulaw64k duration 20 ms bytes 160
*Jan 11 18:33:21.347:ephone-2[2]:Send Encryption Key
!Ephone 2 sends its encryption key.*Jan 11 18:33:21.851:ephone-2[2]::callingNumber 6000

*Jan 11 18:33:21.851:ephone-2[2]::callingParty 6000
*Jan 11 18:33:21.851:ephone-2[2]:Call Info DN 2 line 1 ref 6 call state 4 called 6001
calling 6000 origcalled
*Jan 11 18:33:21.851:ephone-2[2]:Call Info DN 2 line 1 ref 6 called 6001 calling 6000
origcalled 6001 calltype 2
*Jan 11 18:33:21.851:ephone-2[2]:Call Info for chan 1
*Jan 11 18:33:21.851:ephone-2[2]:Original Called Name 6001
*Jan 11 18:33:21.851:ephone-2[2]:6000 calling
*Jan 11 18:33:21.851:ephone-2[2]:6001
```

Configuration Examples for Secure SRST

This section provides the following configuration examples.

- Secure SRST: Example, page 127
- Control Plane Policing: Example, page 132

Note IP addresses and hostnames in examples are fictitious.

Secure SRST: Example

This section provides a configuration example to match the identified configuration tasks in the previous sections. This example does not include using a third-party CA; it assumes the use of the Cisco IOS certificate server to generate your certificates.

```
Router# show running-config
.
.
.
! Define CallManager.
ccm-manager fallback-mgcp
ccm-manager mgcp
ccm-manager music-on-hold
ccm-manager config server 10.1.1.13
ccm-manager config
!
! Define root CA.
crypto pki server srstcaserver
```

```
 database level complete
 database url nvram
 issuer-name CN=srstcaserver

!
crypto pki trustpoint srstca
 enrollment url http://10.1.1.22:80
 revocation-check none
!
crypto pki trustpoint srstcaserver
 revocation-check none
 rsakeypair srstcaserver
!
! Define CTL/7970 trustpoint.
crypto pki trustpoint 7970
 enrollment terminal
 revocation-check none
!
crypto pki trustpoint PEM
 enrollment terminal
 revocation-check none
!
! Define CAPF/7960 trustpoint.
crypto pki trustpoint 7960
 enrollment terminal
 revocation-check none
!
! SRST router device certificate.
crypto pki certificate chain srstca
 certificate 02
  308201AD 30820116 A0030201 02020102 300D0609 2A864886 F70D0101 04050030
  17311530 13060355 0403130C 73727374 63617365 72766572 301E170D 30343034
  31323139 35323233 5A170D30 35303431 32313935 3232335A 30343132 300F0603
  55040513 08443042 39453739 43301F06 092A8648 86F70D01 09021612 6A61736F
  32363931 2E636973 636F2E63 6F6D305C 300D0609 2A864886 F70D0101 01050003
  4B003048 024100D7 0CC354FB 5F7C1AE7 7A25C3F2 056E0485 22896D36 6CA70C19
  C98F9BAE AE9D1F9B D4BB7A67 F3251174 193BB1A3 12946123 E5C1CCD7 A23E6155
  FA2ED743 3FB8B902 03010001 A330302E 300B0603 551D0F04 04030205 A0301F06
  03551D23 04183016 8014F829 CE97AD60 18D05467 FC293963 C2470691 F9BD300D
  06092A86 4886F70D 01010405 00038181 007EB48E CAE9E1B3 D1E7A185 D7F0D565
  CB84B17B 1151BD78 B3E39763 59EC650E 49371F6D 99CBD267 EB8ADF9D 9E43A5F2
  FB2B18A0 34AF6564 11239473 41478AFC A86E6DA1 AC518E0B 8657CEBB ED2BDE8E
  B586FE67 00C358D4 EFDD8D44 3F423141 C2D331D3 1EE43B6E 6CB29EE7 0B8C2752
  C3AF4A66 BD007348 D013000A EA3C206D CF
  quit
 certificate ca 01
  30820207 30820170 A0030201 02020101 300D0609 2A864886 F70D0101 04050030
  17311530 13060355 0403130C 73727374 63617365 72766572 301E170D 30343034
  31323139 34353136 5A170D30 37303431 32313934 3531365A 30173115 30130603
  55040313 0C737273 74636173 65727665 7230819F 300D0609 2A864886 F70D0101
  01050003 818D0030 81890281 8100C3AF EE1E4BB1 9922A8DA 2BB9DC8E 5B1BD332
  1051C9FE 32A971B3 3C336635 74691954 98E765B1 059E24B6 32154E99 105CA989
  9619993F CC72C525 7357EBAC E6335A32 2AAF9391 99325BFD 9B8355EB C10F8963
  9D8FC222 EE8AC831 71ACD3A7 4E918A8F D5775159 76FBF499 5AD0849D CAA41417
  DD866902 21E5DD03 C37D4B28 0FAB0203 010001A3 63306130 0F060355 1D130101
  FF040530 030101FF 300E0603 551D0F01 01FF0404 03020186 301D0603 551D0E04
  160414F8 29CE97AD 6018D054 67FC2939 63C24706 91F9BD30 1F060355 1D230418
  30168014 F829CE97 AD6018D0 5467FC29 3963C247 0691F9BD 300D0609 2A864886
  F70D0101 04050003 8181007A F71B25F9 73D74552 25DFD03A D8D1338F 6792C805
  47A81019 795B5AAE 035400BB F859DABF 21892B5B E71A8283 08950414 8633A8B2
  C98565A6 C09CA641 88661402 ACC424FD 36F23360 ABFF4C55 BB23C66A C80A3A57
  5EE85FF8 C1B1A540 E818CE6D 58131726 BB060974 4E1A2F4B E6195522 122457F3
  DEDBAAD7 3780136E B112A6
  quit
```

```
crypto pki certificate chain srstcaserver
 certificate ca 01
  30820207 30820170 A0030201 02020101 300D0609 2A864886 F70D0101 04050030
  17311530 13060355 0403130C 73727374 63617365 72766572 301E170D 30343034
  31323139 34353136 5A170D30 37303431 32313934 3531365A 30173115 30130603
  55040313 0C737273 74636173 65727665 7230819F 300D0609 2A864886 F70D0101
  01050003 818D0030 81890281 8100C3AF EE1E4BB1 9922A8DA 2BB9DC8E 5B1BD332
  1051C9FE 32A971B3 3C336635 74691954 98E765B1 059E24B6 32154E99 105CA989
  9619993F CC72C525 7357EBAC E6335A32 2AAF9391 99325BFD 9B8355EB C10F8963
  9D8FC222 EE8AC831 71ACD3A7 4E918A8F D5775159 76FBF499 5AD0849D CAA41417
  DD866902 21E5DD03 C37D4B28 0FAB0203 010001A3 63306130 0F060355 1D130101
  FF040530 030101FF 300E0603 551D0F01 01FF0404 03020186 301D0603 551D0E04
  160414F8 29CE97AD 6018D054 67FC2939 63C24706 91F9BD30 1F060355 1D230418
  30168014 F829CE97 AD6018D0 5467FC29 3963C247 0691F9BD 300D0609 2A864886
  F70D0101 04050003 8181007A F71B25F9 73D74552 25DFD03A D8D1338F 6792C805
  47A81019 795B5AAE 035400BB F859DABF 21892B5B E71A8283 08950414 8633A8B2
  C98565A6 C09CA641 88661402 ACC424FD 36F23360 ABFF4C55 BB23C66A C80A3A57
  5EE85FF8 C1B1A540 E818CE6D 58131726 BB060974 4E1A2F4B E6195522 122457F3
  DEDBAAD7 3780136E B112A6
  quit
crypto pki certificate chain 7970
 certificate ca 353FB24BD70F14A346C1F3A9AC725675
  308203A8 30820290 A0030201 02021035 3FB24BD7 0F14A346 C1F3A9AC 72567530
  0D06092A 864886F7 0D010105 0500302E 31163014 06035504 0A130D43 6973636F
  20537973 74656D73 31143012 06035504 03130B43 41502D52 54502D30 3032301E
  170D3033 31303130 32303138 34395A17 0D323331 30313032 30323733 375A302E
  31163014 06035504 0A130D43 6973636F 20537973 74656D73 31143012 06035504
  03130B43 41502D52 54502D30 30323082 0120300D 06092A86 4886F70D 01010105
  00038201 0D003082 01080282 010100C4 266504AD 7DC3FD8D 65556FA6 308FAE95
  B570263B 575ABD96 1CC8F394 5965D9D0 D8CE02B9 F808CCD6 B7CD8C46 24801878
  57DC4440 A7301DDF E40FB1EF 136212EC C4F3B50F BCAFBB4B CD2E5826 34521B65
  01555FE4 D4206776 03368357 83932638 D6FC953F 3A179E44 67255A73 45C69DEE
  FB4D221B 21D7A3AD 38184171 8FD8C271 42183E65 09461434 736C77CC F380EEBF
  632C7B3F A5F92AA6 A8EF3490 8724A84F 4DAF7FD7 0928F585 764D3558 3C0FE9AF
  1ED8763F A299A802 970004AD 1912D265 7DE335B4 BCB6F789 DC68B9FA C8FDF85E
  8A28AD8F 0F4883C0 77112A47 141DBEE0 948FBE53 FE67B308 D40C8029 87BD790E
  CDAB9FD7 A190C1A2 A462C5F2 4A6E0B02 0103A381 C33081C0 300B0603 551D0F04
  04030201 86300F06 03551D13 0101FF04 05300301 01FF301D 0603551D 0E041604
  1452922B E288EE2E 098A4E7E 702C56A5 9AB4D49B 96306F06 03551D1F 04683066
  3064A062 A060862D 68747470 3A2F2F63 61702D72 74702D30 30322F43 65727445
  6E726F6C 6C2F4341 502D5254 502D3030 322E6372 6C862F66 696C653A 2F2F5C5C
  6361702D 7274702D 3030325C 43657274 456E726F 6C6C5C43 41502D52 54502D30
  30322E63 726C3010 06092B06 01040182 37150104 03020100 300D0609 2A864886
  F70D0101 05050003 82010100 56838CEF C4DA3AD1 EA8FBB15 2FFE6EE5 50A1972B
  D4D7AF1F D298892C D5A2A76B C3462866 13E0E55D DC0C4B92 5AA94B6E 69277F9B
  FC73C697 11266E19 451C0FAB A55E6A28 901A48C5 B9911EE6 348A8920 0AEDE1E0
  B6EA781C FFD97CA4 B03C0E34 0E5B0649 8B0A34C9 B73A654E 09050C1F 4DA53E44
  BF78443D B08C3A41 2EEEB873 78CB8089 34F9D16E 91512F0D 3A8674AD 0991ED1A
  92841E76 36D7740E CB787F11 685B9E9D 0C67E85D AF6D05BA 3488E86D 7E2F7F65
  6918DE0F BD3C7F67 D8A33F70 9C4A596E D9F62B3B 1EDEE854 D5882AD4 3D71F72B
  8FAB7F3C 0B5F0759 D9828F83 954D7BB1 57A638EC 7D72BFF1 8933C16F 760BCA94
  4C5B1931 67947A4F 89A1BDB5
  quit
crypto pki certificate chain PEM
 certificate ca 7612F960153D6F9F4E42202032B72356
  308203A8 30820290 A0030201 02021076 12F96015 3D6F9F4E 42202032 B7235630
  0D06092A 864886F7 0D010105 0500302E 31163014 06035504 0A130D43 6973636F
  20537973 74656D73 31143012 06035504 03130B43 41502D52 54502D30 3031301E
  170D3033 30323036 32333237 31335A17 0D323330 32303632 33333633 345A302E
  31163014 06035504 0A130D43 6973636F 20537973 74656D73 31143012 06035504
  03130B43 41502D52 54502D30 30313082 0120300D 06092A86 4886F70D 01010105
  00038201 0D003082 01080282 010100AC 55BBED18 DE9B8709 FFBC8F2D 509AB83A
  21C1967F DEA7F4B0 969694B7 80CC196A 463DA516 54A28F47 5D903B5F 104A3D54
  A981389B 2FC7AC49 956262B8 1C143038 5345BB2E 273FA7A6 46860573 CE5C998D
  55DE78AA 5A5CFE14 037D695B AC816409 C6211F0B 3BBF09CF B0BBB2D4 AC362F67
  0FD145F1 620852B3 1F07E2F1 AA74F150 367632ED A289E374 AF0C5B78 CE7DFB9F
  C8EBBE54 6ECF4C77 99D6DC04 47476C0F 36E58A3B 6BCB24D7 6B6C84C2 7F61D326
  BE7CB4A6 60CD6579 9E1E3A84 8153B750 5527E865 423BE2B5 CB575453 5AA96093
  58B6A2E4 AA3EF081 C7068EC1 DD1EBDDA 53E6F0D6 E2E0486B 109F1316 78C696A3
  CFBA84CC 7094034F C1EB9F81 931ACB02 0103A381 C33081C0 300B0603 551D0F04
  04030201 86300F06 03551D13 0101FF04 05300301 01FF301D 0603551D 0E041604
  14E917B1 82C71FCF ACA91B6E F4A9269C 70AE05A0 9A306F06 03551D1F 04683066
  3064A062 A060862D 68747470 3A2F2F63 61702D72 74702D30 30312F43 65727445
  6E726F6C 6C2F4341 502D5254 502D3030 312E6372 6C862F66 696C653A 2F2F5C5C
  6361702D 7274702D 3030315C 43657274 456E726F 6C6C5C43 41502D52 54502D30
  30312E63 726C3010 06092B06 01040182 37150104 03020100 300D0609 2A864886
  F70D0101 05050003 82010100 AB64FDEB F60C32DC 360F0E10 5FE175FA 0D574AB5
  02ACDCA3 C7BBED15 A4431F20 7E9286F0 770929A2 17E4CDF4 F2629244 2F3575AF
  E90C468C AE67BA08 AAA71C12 BA0C0E79 E6780A5C F814466C 326A4B56 73938380
  73A11AED F9B9DE74 1195C48F 99454B8C 30732980 CD6E7123 8B3A6D68 80B97E00
  7F4BD4BA 0B5AB462 94D9167E 6D8D48F2 597CDE61 25CFADCC 5BD141FB 210275A2
  0A4E3400 1428BA0F 69953BB5 50D21F78 43E3E563 98BCB2B1 A2D4864B 0616BACD
  A61CD9AE C5558A52 B5EEAA6A 08F96528 B1804B87 D26E4AEE AB7AFFE9 2FD2A574
  BAFE0028 96304A8B 13FB656D 8FC60094 D5A53D71 444B3CEF 79343385 3778C193
  74A2A6CE DC56275C A20A303D
  quit
crypto pki certificate chain 7960
 certificate ca F301
  308201F7 30820160 A0030201 020202F3 01300D06 092A8648 86F70D01 01050500
  3041310B 30090603 55040613 02555331 1A301806 0355040A 13114369 73636F20
  53797374 656D7320 496E6331 16301406 03550403 130D4341 50462D33 35453038
  33333230 1E170D30 34303430 39323035 3530325A 170D3139 30343036 32303535
  30315A30 41310B30 09060355 04061302 5553311A 30180603 55040A13 11436973
  636F2053 79737465 6D732049 6E633116 30140603 55040313 0D434150 462D3335
  45303833 33323081 9F300D06 092A8648 86F70D01 01010500 03818D00 30818902
  818100C8 BD9B6035 366B44E8 0F693A47 250FF865 D76C35F7 89B1C4FD 1D122CE0
  F5E5CDFF A4A87EFF 41AD936F E5C93163 3E55D11A AF82A5F6 D563E21C EB89EBFA
  F5271423 C3E875DC E0E07967 6E1AAB4F D3823E12 53547480 23BA1A09 295179B6
  85A0E83A 77DD0633 B9710A88 0890CD4D DB55ADD0 964369BA 489043BB B667E60F
  93954B02 03010001 300D0609 2A864886 F70D0101 05050003 81810056 60FD3AB3
  6F98D2AD 40C309E2 C05B841C 5189271F 01D864E8 98BCE665 2AFBCC8C 54007A84
  8F772C67 E3047A6C C62F6508 B36A6174 B68C1D78 C2228FEA A89ECEFB CC8BA9FC
  0F30E151 431670F9 918514D9 868D1235 18137F1E 50DFD32E 1DC29CB7 95EF4096
  421AF22F 5C1D5804 B83F8E8E 95B04F45 86563BFE DF976C5B FB490A
  quit
!
!
no crypto isakmp enable
!
! Enable IPSec.
crypto isakmp policy 1
 authentication pre-share
 lifetime 28800
crypto isakmp key cisco123 address 10.1.1.13
! The crypto key should match the key configured on Cisco CallManager.
!
! The crypto IPSec configuration should match your Cisco CallManager configuration.

crypto ipsec transform-set rtpset esp-des esp-md5-hmac
!
!
crypto map rtp 1 ipsec-isakmp
 set peer 10.1.1.13
 set transform-set rtpset
 match address 116
!
!
interface FastEthernet0/0
```

```
 ip address 10.1.1.22 255.255.255.0
 duplex auto
 speed auto
 crypto map rtp
!
interface FastEthernet0/1
 no ip address
 shutdown
 duplex auto
 speed auto
!
ip classless
!
ip http server
no ip http secure-server
!
!
! Define traffic to be encrypted by IPSec.
access-list 116 permit ip host 10.1.1.22 host 10.1.1.13
!
!
control-plane
!
!
call application alternate DEFAULT
!
!
voice-port 1/0/0
!
voice-port 1/0/1
!
voice-port 1/0/2
!
voice-port 1/0/3
!
voice-port 1/1/0
 timing hookflash-out 50
!
voice-port 1/1/1
!
voice-port 1/1/2
!
voice-port 1/1/3
!
! Enable MGCP voice protocol.
mgcp
mgcp call-agent 10.1.1.13 2427 service-type mgcp version 0.1
mgcp dtmf-relay voip codec all mode out-of-band
mgcp rtp unreachable timeout 1000 action notify
mgcp package-capability rtp-package
mgcp package-capability sst-package
no mgcp package-capability fxr-package
no mgcp timer receive-rtcp
mgcp sdp simple
mgcp fax t38 inhibit
mgcp rtp payload-type g726r16 static
!
mgcp profile default
!
!
dial-peer voice 81235 pots
 application mgcpapp
 destination-pattern 81235
 port 1/1/0
 forward-digits all
!
dial-peer voice 81234 pots
 application mgcpapp
 destination-pattern 81234
 port 1/0/0
!
dial-peer voice 999100 pots
 application mgcpapp
 port 1/0/0
!
dial-peer voice 999110 pots
 application mgcpapp
 port 1/1/0
!
!
! Enable credentials service on the gateway.
credentials
 ip source-address 10.1.1.22 port 2445
 trustpoint srstca
!
!
! Enable SRST mode.
call-manager-fallback
 secondary-dialtone 9
 transfer-system full-consult
 ip source-address 10.1.1.22 port 2000
 max-ephones 15
 max-dn 30
 transfer-pattern .....
.
.
.
```

Control Plane Policing: Example

This section provides a configuration example for the security best practice of protecting the credentials service port using control plane policing. Control plane policing protects the gateway and maintains packet forwarding and protocol states despite a heavy traffic load. For more information on control planes, see the *Control Plane Policing* documentation.

```
Router# show running-config
.
.
.
! Allow trusted host traffic.
access-list 140 deny tcp host 10.1.1.11 any eq 2445

! Rate-limit all other traffic.
access-list 140 permit tcp any any eq 2445
access-list 140 deny ip any any

! Define class-map "sccp-class."
class-map match-all sccp-class
match access-group 140

policy-map control-plane-policy
class sccp-class
police 8000 1500 1500 conform-action drop exceed-action drop
```

```
! Define aggregate control plane service for the active Route Processor.
control-plane
service-policy input control-plane-policy
.
.
.
```

Where to Go Next

If you require voice mail, see the voice-mail configuration instructions in the "Integrating Voice Mail with Cisco SRST" chapter. You may also want to read the "Monitoring and Maintaining Cisco SRST" chapter.

Additional References

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

- Related Documents, page 133
- Standards, page 134
- MIBs, page 134
- RFCs, page 134
- Technical Assistance, page 134

Related Documents

Related Topic	Documents
SRST commands and specifications	• *Cisco IOS Survivable Remote Site Telephony (SRST) Command Reference (All Versions)* • *Cisco Survivable Remote Site Telephony (SRST) 3.4 Specifications for Cisco IOS Release 12.4(4)T*
Cisco security documentation	• *Media and Signaling Authentication and Encryption Feature for Cisco IOS MGCP Gateways* • *Cisco IOS Certificate Server* • *Manual Certificate Enrollment (TFTP and Cut-and-Paste)* • *Certification Authority Interoperability Commands* • *Certificate Enrollment Enhancements*
Cisco IP phones	• *Cisco IP Phone Authentication and Encryption for Cisco CallManager* • *Phone Guide Cisco IP Phone 7960 and 7940 Series* • *Cisco IP Phone 7960 and 7940 Series User Guide* • *Cisco IP Phone 7970 Guide* • *Cisco IP Phone 7970 Administration Guide for Cisco CallManager,* Release 4.x and later, "Understanding Security Features for Cisco IP Phones" section.

Related Topic	Documents
Command reference and configuration information for voice and telephony commands	• *Cisco IOS Voice Command Reference* • *Cisco IOS Debug Command Reference*
Cisco CallManager user documentation	• *Cisco CallManager* • *Cisco CallManager Security Guide* • *Cisco CallManager Administration Guide*, Release 4.1(2)

Standards

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

MIBs

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

RFCs

RFC	Title
RFC 2246	*The Transport Layer Security (TLS) Protocol Version 1.0*
RFC 3711	*The Secure Real-Time Transport Protocol (SRTP)*

Technical Assistance

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

Integrating Voice Mail with Cisco SRST

This chapter describes how to make your existing voice-mail system run on phones connected to a Cisco Survivable Remote Site Telephony (SRST) router during Cisco CallManager fallback.

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

Information About Integrating Voice Mail with Cisco SRST

Cisco SRST can send and receive voice-mail messages from Cisco Unity and other voice-mail systems during Cisco CallManager fallback. When the WAN is down, a voice-mail system with BRI or PRI access to the Cisco SRST system uses ISDN signaling (see Figure 8). Systems with Foreign Exchange Office (FXO) or Foreign Exchange Station (FXS) access connect to a PSTN and use in-band dual tone multifrequency (DTMF) signaling (see Figure 9).

Figure 8 Cisco CallManager Fallback with BRI or PRI

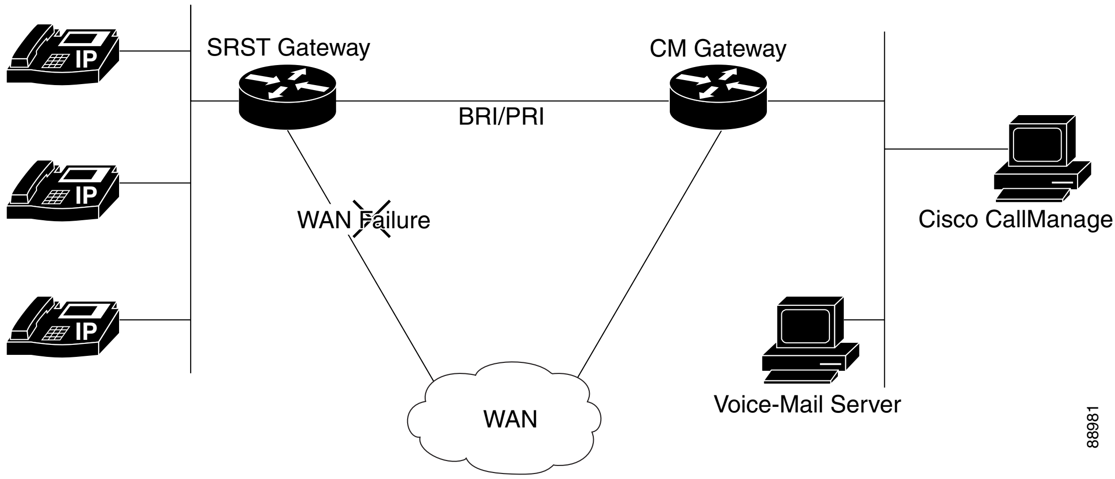
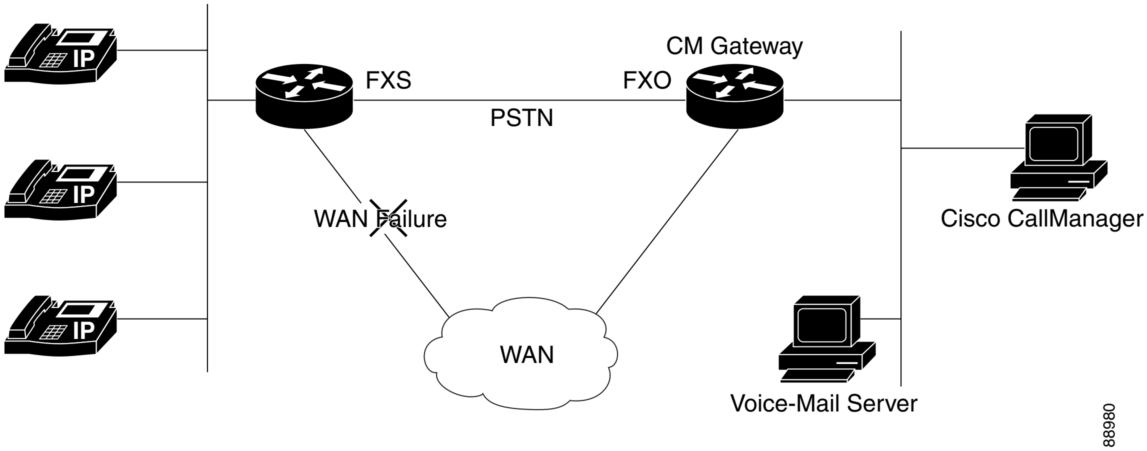


Figure 9 Cisco CallManager Fallback with PSTN



Both configurations allow phone message buttons to remain active and calls to busy or unanswered numbers to be forwarded to the dialed numbers' mailboxes.

Calls that reach a busy signal, calls that are unanswered, and calls made by pressing the message button are forwarded to the voice-mail system. To make this happen, you must configure access from the dial peers to the voice-mail system and establish routing to the voice-mail system for busy and unanswered calls and for message buttons.

If the voice-mail system is accessed over FXO or FXS, you must configure instructions (DTMF patterns) for the voice-mail system so that it can access the correct voice-mail system mailbox. If your voice-mail system is accessed over BRI or PRI, no instructions are necessary because the voice-mail system can log in to the calling phone's mailbox directly.

How to Integrate Voice Mail with Cisco SRST

This section contains the following tasks:

- Configuring Direct Access to Voice Mail, page 137 (Required)
- Configuring Message Buttons, page 140 (Required)
- Redirecting to Cisco CallManager Gateway, page 142 (Required for BRI or PRI))
- Configuring Call Forwarding to Voice Mail, page 142 (Required FXO or FXS)
- Configuring Message Waiting Indication, page 147 (Optional)

Configuring Direct Access to Voice Mail

To access voice-mail messages with FXO or FXS access, you must have POTS dial peers configured with a destination pattern that matches the voice-mail system's number. Also, you must associate the dial peer with the port to which the voice-mail system is accessed.

Both sets of configurations are done in global configuration mode and in dial-peer configuration mode. The summary and detailed steps below include only the basic commands necessary to perform this task. You may require additional commands for your particular dial-peer configuration.

For additional information about the commands in the steps below, see the *Cisco IOS Voice, Video, and Fax Command Reference,* Release 12.2T.

SUMMARY STEPS

1. **dial-peer voice** *tag* {**pots** | **voatm** | **vofr** | **voip**}
2. **destination-pattern** [+] *string* [**T**]
3. **port** {*slot-number***/***subunit-number***/***port* | *slot***/***port***:***ds0-group-no*}
4. **forward-digits** {*num-digit* | **all** | **extra**}
5. **exit**

DETAILED STEPS

	Command or Action	Purpose

Step 1	**dial-peer voice** *tag* {**pots** | **voatm** | **vofr** | **voip**}
	Example:
	`Router(config)# dial-peer voice 1002 pots`
		(FXO or FXS and BRI or PRI) Defines a particular dial peer, specifies the method of voice encapsulation, and enters dial-peer configuration mode. The **dial-peer** command provides different syntax for individual routers. This example is syntax for Cisco 3600 series routers.
		• *tag*—Digits that define a particular dial peer. Range is from 1 to 2147483647.
		• **pots**—Indicates that this is a POTS dial peer that uses VoIP encapsulation on the IP backbone.
		• **voatm**—Specifies that this is a VoATM dial peer that uses real-time AAL5 voice encapsulation on the ATM backbone network.
		• **vofr**—Specifies that this is a VoFR dial peer that uses FRF.11 encapsulation on the Frame Relay backbone network.
		• **voip**—Indicates that this is a VoIP dial peer that uses voice encapsulation on the POTS network.

Step 2	**destination-pattern** [**+**] *string* [**T**] **Example:** `Router(config-dial-peer)# destination-pattern 1100T`	(FXO or FXS and BRI or PRI) Specifies either the prefix or the full E.164 telephone number (depending on your dial plan) to be used for a dial peer. • **+**—(Optional) Character that indicates an E.164 standard number. • *string*—See Table 10. • **T**—(Optional) Control character that indicates that the destination-pattern value is a variable-length dial string.

Step 3	**port** {*slot-number***/***subunit-number***/***port* | *slot***/***port***:***ds0-group-no*}
	Example:
	`Router(config-dial-peer)# port 1/1/1`
		(FXO or FXS and BRI or PRI) Associates a dial peer with a specific voice port on Cisco 3600 series routers.
		• *slot-number*—Number of the slot in the router in which the voice interface card (VIC) is installed. Valid entries are from 0 to 3, depending on the slot in which it has been installed.
		• *subunit-number*—Subunit on the VIC in which the voice port is located. Valid entries are 0 or 1.
		• *port*—Voice port number. Valid entries are 0 and 1.
		• *ds0-group-no*—Specifies the DS0 group number. Each defined DS0 group number is represented on a separate voice port. This allows you to define individual DS0s on the digital T1/E1 card.

	Command or Action	Purpose

Step 4	`forward-digits {num-digit | all | extra}`
	Example:
	`Router(config-dial-peer)# forward-digits all`
		(Optional for FXO or FXS) Specifies which digits to forward for voice calls.
		• *num-digit*—The number of digits to be forwarded. If the number of digits is greater than the length of a destination phone number, the length of the destination number is used. Range is 0 to 32. Setting the value to 0 is equivalent to entering the **no forward-digits** command.
		• **all**—Forwards all digits. If **all** is entered, the full length of the destination pattern is used.
		• **extra**—If the length of the dialed digit string is greater than the length of the dial-peer destination pattern, the extra right-justified digits are forwarded. However, if the dial-peer destination pattern is variable length and ends with the character "T" (for example: T, 123T, 123...T), extra digits are not forwarded.

Step 5	`exit`
	Example:
	`Router(config-dial-peer)# exit`
		(FXO or FXS and BRI or PRI) Exits dial-peer configuration mode.

Table 10 Valid Entries for the string Argument in the destination-pattern Command

Entry	Description
Digits 0 through 9	—
Letters A through D	—
Asterisk (*) and pound sign (#)	These appear on standard touch-tone dial pads.
Comma (,)	Inserts a pause between digits.
Period (.)	Matches any entered digit (this character is used as a wildcard).
Percent sign (%)	Indicates that the preceding digit occurred zero or more times; similar to the wildcard usage.
Plus sign (+)	Indicates that the preceding digit occurred one or more times. **Note** The plus sign used as part of a digit string is different from the plus sign that can be used in front of a digit string to indicate that the string is an E.164 standard number.
Circumflex (^)	Indicates a match to the beginning of the string. Parentheses (( )), which indicate a pattern and are the same as the regular expression rule.
Dollar sign ($)	Matches the null string at the end of the input string.
Backslash symbol (\)	Is followed by a single character and matches that character. Can be used with a single character with no other significance (matching that character).
Question mark (?)	Indicates that the preceding digit occurred zero or one time.
Brackets ([ ])	Indicates a range. A range is a sequence of characters enclosed in the brackets; only numeric characters from 0 to 9 are allowed in the range.

Examples

The following FXO and FXS example sets up a POTS dial peer named 1102, matches dial-peer 1102 to voice-mail extension 1101, and assigns dial-peer 1102 to voice-port 1/1/1 where the voice-mail system is connected. Other dial peers are configured for direct access to voice mail.

```
voice-port 1/1/1
 timing digit 250
 timing inter-digit 250

dial-peer voice 1102 pots
 destination-pattern 1101
 port 1/1/1
 forward-digits all

dial-peer voice 1103 pots
 destination-pattern 1101
 port 1/1/1
 forward-digits all

dial-peer voice 1104 pots
 destination-pattern 1101
 port 1/1/1
 forward-digits all
```

The following example sets up a POTS dial peer named 1102 to go directly to 1101 through port 2/0:23.

```
controller T1 2/0
 framing esf
 clock source line primary
 linecode b8zs
 cablelength short 133
 pri-group timeslots 21-24

interface Serial2/0:23
 no ip address
 no logging event link-status
 isdn switch-type primary-net5
 isdn incoming-voice voice
 isdn T309-enable
 no cdp enable

voice-port 2/0:23

dial-peer voice 1102 pots
 destination-pattern 1101T
 port 2/0:23
```

Configuring Message Buttons

To activate the message buttons on Cisco IP phones connected to the Cisco SRST router during Cisco CallManager fallback, you must program a speed-dial number to the voice-mail system. The speed-dial number is dialed when message buttons on phones connected to the Cisco SRST router are pressed during Cisco CallManager fallback. In addition, call forwarding must be configured so that calls to busy and unanswered numbers are sent to the voice-mail number.

This configuration is required for FXO or FXS and BRI or PRI.

SUMMARY STEPS

1. **call-manager-fallback**
2. **voicemail** *phone-number*
3. **call-forward busy** *directory-number*
4. **call-forward noan** *directory-number* **timeout** *seconds*
5. **exit**

DETAILED STEPS

	Command or Action	Purpose

Step 1	**`call-manager-fallback`**
	Example:
	`Router(config)# call-manager-fallback`
		Enters call-manager-fallback configuration mode.

Step 2	**`voicemail`** *`phone-number`*
	Example:
	`Router(config-cm-fallback)# voicemail 5550100`
		Configures the telephone number that is dialed when the message button on a Cisco IP phone is pressed.
		• *phone-number*—Phone number configured as a speed-dial number for retrieving messages.

Step 3	**`call-forward busy`** *`directory-number`*
	Example:
	`Router(config-cm-fallback)# call-forward busy 2000`
		Configures call forwarding to another number when the Cisco IP phone is busy.
		• *directory-number*—Selected directory number representing a fully qualified E.164 number. This number can contain "." wildcard characters that correspond to the right-justified digits in the directory number extension.

Step 4	**`call-forward noan`** *`directory-number`* **`timeout`** *`seconds`*
	Example:
	`Router(config-cm-fallback)# call-forward noan 2000 timeout 10`
		Configures call forwarding to another number when no answer is received from the Cisco IP phone.
		• *directory-number*—Selected directory number representing a fully qualified E.164 number. This number can contain "." wildcard characters that correspond to the right-justified digits in the directory number extension.
		• **timeout** *seconds*—Sets the waiting time, in seconds, before the call is forwarded to another phone. The *seconds* range is from 3 to 60000.

Step 5	**`exit`**
	Example:
	`Router(config-cm-fallback)# exit`
		Exits call-manager-fallback configuration mode.

Examples

The following example specifies 1101 as the speed-dial number that is issued when message buttons are pressed on Cisco IP phones connected to the Cisco SRST router. All busy and unanswered calls are configured to be forwarded to the voice-mail number (1101).

```
call-manager-fallback
 voicemail 1101
 call-forward busy 1101
 call-forward noan 1101 timeout 3
```

Redirecting to Cisco CallManager Gateway

Note The following task is required for voice-mail systems with BRI or PRI access.

In addition to supporting message buttons for retrieving personal messages, Cisco SRST allows the automatic forwarding of calls to busy and unanswered numbers to voice-mail systems. Voice-mail systems with BRI or PRI access can log in to the calling phone's mailbox directly. For this to happen, some Cisco CallManager configuration is recommended. If your voice-mail system supports Redirected Dialed Number Identification Service (RDNIS), RDNIS must be included in the outgoing SETUP message to Cisco CallManager to declare the last redirected number and the originally dialed number to and from configured devices and applications.

Step 1 From any page in Cisco CallManager, click **Device** and **Gateway.**

Step 2 From the Find and List Gateways page, click **Find**.

Step 3 From the Find and List Gateways page, choose a device name.

Step 4 From the Gateway Configuration page, check **Redirecting Number IE Delivery - Outgoing**.

Configuring Call Forwarding to Voice Mail

Note The following task is required for voice-mail systems with FXO or FXS access.

In addition to supporting message buttons for retrieving personal messages, Cisco SRST allows the automatic forwarding of calls to busy or unanswered numbers to voice-mail systems. The forwarded calls can be routed to almost any location in the voice-mail system. Typically, calls are forwarded to a location in the called number's mailbox where the caller can leave messages.

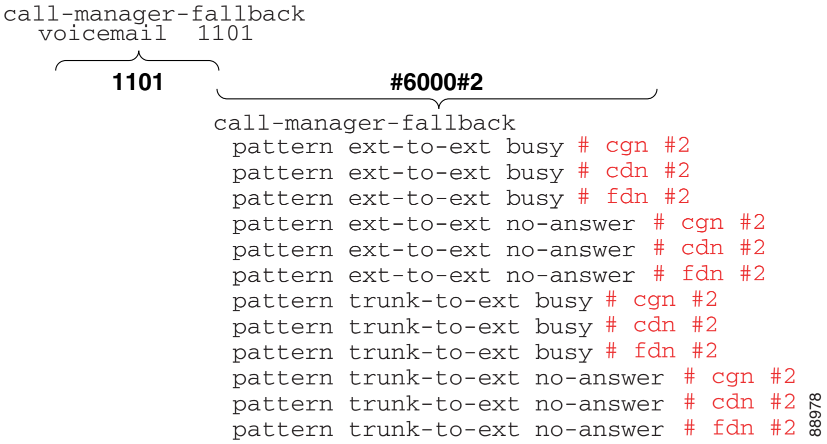
Call Routing Instructions Using DTMF Digit Patterns

Cisco SRST call-routing instructions are required so that forwarded calls can be sent to the correct voice mailboxes. These instructions consist of DTMF digits configured in patterns that match the dial sequences required by the voice-mail system to get to a particular voice-mail location. For example, a voice-mail system may be designed so that callers must do the following to leave a message:

1. Dial the central voice-mail number (1101) and press #.
2. Dial an extension number (6000) and press #.
3. Dial 2 to select the menu option for leaving messages in the extension number's mailbox.

For Cisco SRST to forward a call to a busy or unanswered number to extension 6000's mailbox, it must be programmed to issue a sequence of 1101#6000#2. As shown in Figure 10, this is accomplished through the **voicemail** and **pattern** commands.

Figure 10 How Voice-Mail Dial Sequence 1101#6000#2 Is Configured in Cisco SRST



```
call-manager-fallback
  voicemail  1101
```

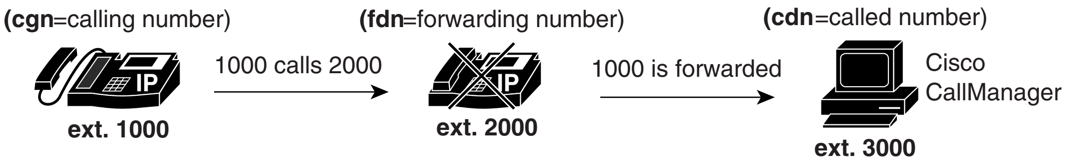
```
call-manager-fallback
 pattern ext-to-ext busy # cgn #2
 pattern ext-to-ext busy # cdn #2
 pattern ext-to-ext busy # fdn #2
 pattern ext-to-ext no-answer # cgn #2
 pattern ext-to-ext no-answer # cdn #2
 pattern ext-to-ext no-answer # fdn #2
 pattern trunk-to-ext busy # cgn #2
 pattern trunk-to-ext busy # cdn #2
 pattern trunk-to-ext busy # fdn #2
 pattern trunk-to-ext no-answer # cgn #2
 pattern trunk-to-ext no-answer # cdn #2
 pattern trunk-to-ext no-answer # fdn #2
```

The # cgn #2, # cdn #2, and # fdn #2 portions of the **pattern** commands shown in Figure 10 are DTMF digit patterns. These patterns are composed of tags and tokens. Tags are sets of characters representing DTMF tones. Tokens consist of three command keywords (**cgn**, **cdn**, and **fdn**) that declare the state of an incoming call transferred to voice mail.

A tag can be up to three character from the DTMF tone set (A to D, 0 to 9, # and *). Voice-mail systems can use limited sets of DTMF tones. For example, Cisco Unity uses all DTMF tones but A to D. Tones can be defined in multiple ways. For example, when the star (*) is placed in front of a token by itself, it can mean "dial the following token number," or, if it is at the end of a token, it can mark the end of a token number. If the asterisk is between other tag characters, it can mean dial *. The use of tags depends on how DTMF tones are defined by your voice-mail system.

Tokens tell Cisco SRST what telephone number in the call forwarding chain to use in the pattern. As shown in Figure 11, there are three kinds of tokens that correspond to three possible call states during voice-mail forwarding.

Figure 11 How Numbers Are Extracted from Tokens



```
pattern ext-to-ext busy # cdn # 2 = pattern ext-to-ext busy # 3000 # 2
pattern ext-to-ext busy # fdn # 2 = pattern ext-to-ext busy # 2000 # 2
pattern ext-to-ext busy # cgn # 2 = pattern ext-to-ext busy # 1000 # 2
```

Sets of tags and tokens or patterns activate a voice-mail system when

- A user presses the message button on a phone (**pattern direct** command).
- An internal extension attempts to connect to a busy extension and the call is forwarded to voice mail (**pattern ext-to-ext busy** command).
- An internal extension fails to connect to an extension and the call is forwarded to voice mail (**pattern ext-to-ext no-answer** command).
- An external trunk call reaches a busy extension and the call is forwarded to voice mail (**pattern trunk-to-ext busy** command).
- An external trunk call reaches an unanswered extension and the call is forwarded to voice mail (**pattern trunk-to-ext no-answer** command).

Prerequisites

- FXO hairpin-forwarded calls to voice-mail systems must have disconnect supervision from the central office. For further information, see the *FXO Answer and Disconnect Supervision* document.
- To configure patterns that your voice-mail system will interpret correctly, you must know how the system routes voice-mail calls and interprets DTMF tones (see the "Call Routing Instructions Using DTMF Digit Patterns" section on page 143).

  You can find information about how Cisco Unity handles voice-mail calls in the *How to Transfer a Caller Directly into a Cisco Unity Mailbox* document. Additional call-handling information can be found in the "Subscriber and Operator Orientation" chapters of any Cisco Unity system administration guide book.

  For other voice-mail systems, see the analog voice mail integration configuration guide or information about the system's call handling.

SUMMARY STEPS

1. **vm-integration**
2. **pattern direct** *tag1* {**CGN** | **CDN** | **FDN**} [*tag2* {**CGN** | **CDN** | **FDN**}] [*tag3* {**CGN** | **CDN** | **FDN**}] [*last-tag*]
3. **pattern ext-to-ext busy** *tag1* {**CGN** | **CDN** | **FDN**} [*tag2* {**CGN** | **CDN** | **FDN**}] [*tag3* {**CGN** | **CDN** | **FDN**}] [*last-tag*]
4. **pattern ext-to-ext no-answer** *tag1* {**CGN** | **CDN** | **FDN**} [*tag2* {**CGN** | **CDN** | **FDN**}] [*tag3* {**CGN** | **CDN** | **FDN**}] [*last-tag*]

5. **pattern trunk-to-ext busy** *tag1* {**CGN** | **CDN** | **FDN**} [*tag2* {**CGN** | **CDN** | **FDN**}] [*tag3* {**CGN** | **CDN** | **FDN**}] [*last-tag*]
6. **pattern trunk-to-ext no-answer** *tag1* {**CGN** | **CDN** | **FDN**} [*tag2* {**CGN** | **CDN** | **FDN**}] [*tag3* {**CGN** | **CDN** | **FDN**}] [*last-tag*]

DETAILED STEPS

	Command or Action	Purpose
Step 1	**vm-integration** **Example:** `Router(config)# vm-integration`	Enters voice-mail integration mode and enables voice-mail integration with DTMF and analog voice-mail systems.
Step 2	**pattern direct** *tag1* {**CGN** | **CDN** | **FDN**} [*tag2* {**CGN** | **CDN** | **FDN**}] [*tag3* {**CGN** | **CDN** | **FDN**}] [*last-tag*] **Example:** `Router(config-vm-int)# pattern direct 2 CGN *`	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when the user presses the messages button on the phone. • *tag1*—Alphanumeric string fewer than four DTMF digits in length. The alphanumeric string consists of a combination of four letters (A, B, C, and D), two symbols (* and #), and ten digits (0 to 9). The tag numbers match the numbers defined in the voice-mail system's integration file, immediately preceding either the number of the calling party, the number of the called party, or a forwarding number. • *tag2* and *tag3*—(Optional) See *tag1*. • *last-tag*—See *tag1*. This tag indicates the end of the pattern. • **CGN**—Calling number (CGN) information is sent to the voice-mail system. • **CDN**—Called number (CDN) information is sent to the voice-mail system. • **FDN**—Forwarding number (FDN) information is sent to the voice-mail system.
Step 3	**pattern ext-to-ext busy** *tag1* {**CGN** | **CDN** | **FDN**} [*tag2* {**CGN** | **CDN** | **FDN**}] [*tag3* {**CGN** | **CDN** | **FDN**}] [*last-tag*] **Example:** `Router(config-vm-int)# pattern ext-to-ext busy 7 FDN * CGN *`	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an internal extension attempts to connect to a busy extension and the call is forwarded to voice mail. For argument and keyword information, see Step 2.
Step 4	**pattern ext-to-ext no-answer** *tag1* {**CGN** | **CDN** | **FDN**} [*tag2* {**CGN** | **CDN** | **FDN**}] [*tag3* {**CGN** | **CDN** | **FDN**}] [*last-tag*] **Example:** `Router(config-vm-int)# pattern ext-to-ext no-answer 5 FDN * CGN *`	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an internal extension fails to connect to an extension and the call is forwarded to voice mail. For argument and keyword information, see Step 2.

	Command or Action	Purpose
Step 5	**pattern trunk-to-ext busy** *tag1* {**CGN** | **CDN** | **FDN**} [*tag2* {**CGN** | **CDN** | **FDN**}] [*tag3* {**CGN** | **CDN** | **FDN**}] [*last-tag*] **Example:** `Router(config-vm-int)# pattern trunk-to-ext busy 6 FDN * CGN *`	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an external trunk call reaches a busy extension and the call is forwarded to voice mail. For argument and keyword information, see Step 2.
Step 6	**pattern trunk-to-ext no-answer** *tag1* {**CGN** | **CDN** | **FDN**} [*tag2* {**CGN** | **CDN** | **FDN**}] [*tag3* {**CGN** | **CDN** | **FDN**}] [*last-tag*] **Example:** `Router(config-vm-int)# pattern trunk-to-ext no-answer 4 FDN * CGN *`	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when an external trunk call reaches an unanswered extension and the call is forwarded to voice mail. For argument and keyword information, see Step 2.

Examples

For the following configuration, if the voice-mail number is 1101, and 3001 is a phone with a message button, 1101*3001 would be dialed automatically when the 3001 message button is pressed. Under these circumstances, 3001 is considered to be a calling number or inbound call number.

```
vm-integration
 pattern direct * CGN
```

For the following configuration, if 3001 calls 3006 and 3006 does not answer, the SRST router will forward 3001 to the voice-mail system (1101) and send to the voice-mail system the DTMF pattern # 3006 #2. This pattern is intended to select voice mailbox number 3006 (3006's voice mailbox). For this pattern to be sent, 3001 must be a forwarding number.

```
vm-integration
 pattern ext-to-ext no-answer # FDN #2
```

For the following configuration, if 3006 is busy and 3001 calls 3006, the SRST router will forward 3001 to the voice-mail system (1101) and send to the voice-mail system the DTMF pattern # 3006 #2. This pattern is intended to select voice mailbox number 3006 (3006's voice mailbox). For this pattern to be sent, 3001 must be a forwarding number.

```
vm-integration
 pattern ext-to-ext busy # FDN #2
```

Configuring Message Waiting Indication

The MWI relay mechanism is initiated after someone leaves a voice-mail message on the remote voice-mail message system. MWI relay is required when one Cisco Unity Voice Mail system is shared by multiple Cisco SRST routers. SRST routers use the SIP Subscribe and Notify methods for MWI. See the *Configuring Cisco IOS SIP Configuration Guide* for more information on SIP MWI and the Subscribe and Notify methods. The SRST router that is the SIP MWI relay server acts as the SIP notifier. The other remote routers act as the SIP subscribers.

SUMMARY STEPS

1. **call-manager-fallback**
2. **mwi relay**
3. **mwi reg-e164**
4. **exit**
5. **sip-ua**
6. **mwi-server** {**ipv4:***destination-address* | **dns:***host-name*} [**expires** *seconds*] [port *port*] [**transport** {**tcp** | **udp**}] [**unsolicited**]
7. **exit**

DETAILED STEPS

	Command	Purpose
Step 1	**call-manager-fallback** **Example:** `Router(config)# call-manager-fallback`	Enters call-manager-fallback configuration mode.
Step 2	**mwi relay** **Example:** `Router(config-cm-fallback)# mwi relay`	Enables the SRST router to relay MWI information to remote Cisco IP phones.
Step 3	**mwi reg-e164** **Example:** `Router(config-cm-fallback)# mwi reg-e164`	Registers E.164 numbers rather than extension numbers with a SIP proxy or registrar.
Step 4	**exit** **Example:** `Router(config-cm-fallback)# exit`	Exits call-manager-fallback configuration mode.
Step 5	**sip-ua** **Example:** `Router(config)# sip-ua`	Enters SIP user-agent configuration mode.

	Command	Purpose
Step 6	**mwi-server** {**ipv4:***destination-address* | **dns:***host-name*} [**expires** *seconds*] [**port** *port*] [**transport** {**tcp** | **udp**}] [**unsolicited**] **Example:** `Router(config-sip-ua)# mwi-server ipv4:10.0.2.254`	Configures voice-mail server settings on a voice gateway or user agent. The IP address and port for the SIP-based MWI server should be in the same LAN as the voice-mail server. The MWI server is a Cisco SRST router. Keywords and arguments are as follows: • **ipv4:***destination-address*—IP address of the voice-mail server. • **dns:***host-name*—Host device housing the domain name server that resolves the name of the voice-mail server. The argument should contain the complete hostname to be associated with the target address; for example, **dns:test.cisco.com**. • **expires** *seconds*—Subscription expiration time, in seconds. Range is from 1 to 999999. Default is 3600. • **port** *port*—Port number on the voice-mail server. Default is 5060. • **transport**—Transport protocol to the voice-mail server. Valid values are tcp and udp. Default is UDP. • **unsolicited**—Requires the voice-mail server to send a SIP notification message to the voice gateway or UA if the mailbox status changes. Removes the requirement that the voice gateway subscribe for MWI service.
Step 7	**exit** **Example:** `Router(config-sip-ua)# exit`	Exits SIP user-agent configuration mode.

Configuration Examples

This section provides the following configuration examples:

- Configuring Local Voice-Mail System (FXO and FXS): Example, page 149
- Configuring Central Location Voice-Mail System (FXO and FXS): Example, page 150
- Configuring Voice-Mail Access over FXO and FXS: Example, page 150
- Configuring Voice-Mail Access over BRI and PRI: Example, page 151

Configuring Local Voice-Mail System (FXO and FXS): Example

The "Dial-Peer Configuration for Integration of Voice-Mail with Cisco SRST" section of the example below shows a legacy dial-peer configuration for a local voice-mail system. The "Cisco SRST Voice-Mail Integration Pattern Configuration" section must be compatible with your voice-mail system configuration.

```
! Dial-Peer Configuration for Integration of Voice-Mail with Cisco SRST
!
dial-peer voice 101 pots
 destination-pattern 14011
 port 3/0/0
!
dial-peer voice 102 pots
 preference 1
 destination-pattern 14011
 port 3/0/1
!
dial-peer voice 103 pots
 preference 2
 destination-pattern 14011
 port 3/1/0
!
dial-peer voice 104 pots
 destination-pattern 14011
 port 3/1/1
!
! Cisco SRST configuration
!
call-manager-fallback
 max-ephones 24
 max-dn 144
 ip source-address 1.4.214.104 port 2000
 voicemail 14011
 call-forward busy 14011
 call-forward noan 14011 timeout 3

! Cisco SRST Voice-Mail Integration Pattern Configuration
!
vm-integration
 pattern direct 2 CGN *
 pattern ext-to-ext no-answer 5 FDN * CGN *
 pattern ext-to-ext busy 7 FDN * CGN *
 pattern trunk-to-ext no-answer 4 FDN * CGN *
 pattern trunk-to-ext busy 6 FDN * CGN *
```

Configuring Central Location Voice-Mail System (FXO and FXS): Example

The "Dial-Peer Configuration for Integration of Voice-Mail with Cisco SRST in Central Location" section of the example shows a legacy dial-peer configuration for a central voice-mail system. The "Cisco SRST Voice-Mail Integration Pattern Configuration" section must be compatible with your voice-mail system configuration.

Note Message waiting indicator (MWI) integration is not supported for PSTN access to voice-mail systems at central locations.

```
! Dial-Peer Configuration for Integration of Voice-Mail with Cisco SRST in Central
! Location
!
dial-peer voice 101 pots
 destination-pattern 14011
 port 3/0/0
!
! Cisco SRST configuration
!
call-manager-fallback
 max-ephones 24
 max-dn 144
 ip source-address 1.4.214.104 port 2000
 voicemail 14011
 call-forward busy 14011
 call-forward noan 14011 timeout 3
!
! Cisco SRST Voice-Mail Integration Pattern Configuration
!
vm-integration
 pattern direct 2 CGN *
 pattern ext-to-ext no-answer 5 FDN * CGN *
 pattern ext-to-ext busy 7 FDN * CGN *
 pattern trunk-to-ext no-answer 4 FDN * CGN *
 pattern trunk-to-ext busy 6 FDN * CGN *
```

Configuring Voice-Mail Access over FXO and FXS: Example

The following example shows how to configure the Cisco SRST router to forward unanswered calls to voice mail. In this example, the voice-mail number is 1101, the voice-mail system is connected to FXS voice port 1/1/1, and the voice mailbox numbers are 3001, 3002, and 3006.

```
voice-port 1/1/1
 timing digit 250
 timing inter-digit 250

dial-peer voice 1102 pots
 destination-pattern 1101T
 port 1/1/1

call-manager-fallback
 timeouts interdigit 5
 ip source-address 1.6.0.199 port 2000
 max-ephones 24
 max-dn 24
 transfer-pattern 3...
 voicemail 1101
 call-forward busy 1101
 call-forward noan 1101 timeout 3
 moh minuet.au

vm-integration
 pattern direct * CGN
 pattern ext-to-ext no-answer # FDN #2
 pattern ext-to-ext busy # FDN #2
 pattern trunk-to-ext no-answer # FDN #2
 pattern trunk-to-ext busy # FDN #2
```

Configuring Voice-Mail Access over BRI and PRI: Example

The following example shows how to configure the Cisco SRST router to forward unanswered calls to voice mail. In this example, the voice-mail number is 1101, the voice-mail system is connected to a BRI or PRI voice port, and the voice mailbox numbers are 3001, 3002, and 3006.

```
controller T1 2/0
 framing esf
 clock source line primary
 linecode b8zs
 cablelength short 133
 pri-group timeslots 21-24

interface Serial2/0:23
 no ip address
 no logging event link-status
 isdn switch-type primary-net5
 isdn incoming-voice voice
 isdn T309-enable
 no cdp enable

voice-port 2/0:23

dial-peer voice 1102 pots
 destination-pattern 1101T
 direct-inward-dial
 port 2/0:23

call-manager-fallback
 timeouts interdigit 5
 ip source-address 1.6.0.199 port 2000
 max-ephones 24
 max-dn 24
 transfer-pattern 3...
 voicemail 1101
 call-forward busy 1101
 call-forward noan 1101 timeout 3
 moh minuet.au
```

Where to Go Next

For information about monitoring and maintaining Cisco SRST, go to the "Monitoring and Maintaining Cisco SRST" chapter.

Monitoring and Maintaining Cisco SRST

To monitor and maintain Cisco Survivable Remote Site Telephony (SRST), use the following commands in the privileged EXEC and mode.

Command	Purpose
`Router# show running-config`	Displays the configuration.
`Router# show call-manager-fallback all`	Displays the detailed configuration of all the Cisco IP phones, voice ports, and dial peers of the Cisco SRST router.
`Router# show call-manager-fallback dial-peer`	Displays the output of the dial peers of the Cisco SRST router.
`Router# show call-manager-fallback ephone-dn`	Displays Cisco IP phone destination numbers when in call manager fallback mode.
`Router# show call-manager-fallback voice-port`	Displays output for the voice ports.
`Router# show ephone phone`	Displays Cisco IP phone status.
`Router# show ephone offhook`	Displays Cisco IP phone status for all phones that are off hook.
`Router# show ephone registered`	Displays Cisco IP phone status for all phones that are currently registered.
`Router# show ephone remote`	Displays Cisco IP phone status for all nonlocal phones (phones that have no Address Resolution Protocol [ARP] entry).
`Router# show ephone ringing`	Displays Cisco IP phone status for all phones that are ringing.
`Router# show ephone summary`	Displays a summary of all Cisco IP phones.
`Router# show ephone telephone-number phone-number`	Displays Cisco IP phone status for a specific phone number.
`Router# show ephone unregistered`	Displays Cisco IP phone status for all unregistered phones.
`Router# show ephone-dn tag`	Displays Cisco IP phone destination numbers.
`Router# show ephone-dn summary`	Displays a summary of all Cisco IP phone destination numbers.
`Router# show ephone-dn loopback`	Displays Cisco IP phone destination numbers in loopback mode.
`Router# show voice port summary`	Displays a summary of all voice ports.
`Router# show dial-peer voice summary`	Displays a summary of all voice dial peers.

Appendix A: Preparing Cisco SRST Support for SIP

Cisco Survivable Remote Site Telephony (SRST) supports incoming and outgoing Session Initiation Protocol (SIP) calls to and from IP phones and router voice gateway voice ports, but does not support direct attachment of SIP phones to Cisco SRST. SIP may be used in situations where the SRST router is separate from the PSTN gateway and the SRST and PSTN gateways are linked together using SIP (instead of H.323).

Special configurations to support SIP calls are described in this appendix. For more information about SIP, see the *Cisco IOS SIP Configuration Guide*.

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

DTMF Relay for SIP Applications and Voice Mail

DTMF relay for SIP applications can be used in two voice-mail situations:

- DTMF Relay Using SIP RFC 2833, page 155
- DTMF Relay Using SIP Notify (Nonstandard), page 157

DTMF Relay Using SIP RFC 2833

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

The SIP DTMF relay method is needed in the following situations:

- When SIP is used to connect a Cisco SRST system to a remote SIP-based IVR or voice-mail application, such as Cisco Unity.
- When SIP is used to connect a Cisco SRST system to a remote SIP-PSTN voice gateway that goes through the PSTN to a voice-mail or IVR application.

Note The need to use out-of-band DTMF relay conversion is limited to SCCP phones. SIP phones natively support in-band DTMF relay as specified in RFC 2833.

To enable SIP DTMF relay using RFC 2833, the commands in this section must be used on both originating and terminating gateways.

SUMMARY STEPS

1. **dial-peer voice** *tag* **voip**
2. **dtmf-relay rtp-nte**
3. **exit**
4. **sip-ua**
5. **notify telephone-event max-duration** *time*
6. **exit**

DETAILED STEPS

	Command or Action	Purpose

Step 1	**`dial-peer voice`** *`tag`* **`voip`**
	Example:
	`Router(config)# dial-peer voice 2 voip`
		Enters dial-peer configuration mode.

Step 2	**`dtmf-relay rtp-nte`**
	Example:
	`Router(config-dial-peer)# dtmf-relay rtp-nte`
		Forwards DTMF tones by using Real-Time Transport Protocol (RTP) with the Named Telephone Event (NTE) payload type.

Step 3	**`exit`**
	Example:
	`Router(config-dial-peer)# exit`
		Exits dial-peer configuration mode.

Step 4	**`sip-ua`**
	Example:
	`Router(config)# sip-ua`
		Enables SIP user-agent configuration mode.

	Command or Action	Purpose

Step 5	**`notify telephone-event max-duration`** *`time`*
	Example:
	`Router(config-sip-ua)# notify telephone-event max-duration 2000`
		Configures the maximum time interval allowed between two consecutive NOTIFY messages for a single DTMF event.
		• **max-duration** *time*—Time interval between consecutive NOTIFY messages for a single DTMF event, in milliseconds. Range is from 500 to 3000. Default is 2000.

Step 6	**`exit`**
	Example:
	`Router(config-sip-ua)# exit`
		Exits SIP user-agent configuration mode.

Troubleshooting Tips

The dial-peer section of the **show running-config** command output displays DTMF relay status when it is configured, as shown in this excerpt:

```
dial-peer voice 123 voip
 destination-pattern [12]...
 monitor probe icmp-ping
 session protocol sipv2
 session target ipv4:10.8.17.42
 dtmf-relay rtp-nte
```

DTMF Relay Using SIP Notify (Nonstandard)

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

SUMMARY STEPS

1. **dial-peer voice** *tag* **voip**
2. **dtmf-relay sip-notify**
3. **exit**
4. **sip-ua**
5. **notify telephone-event max-duration** *time*
6. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**dial-peer voice** *tag* **voip** **Example:** `Router(config)# dial-peer voice 2 voip`	Enters dial-peer configuration mode.
Step 2	**dtmf-relay sip-notify** **Example:** `Router(config-dial-peer)# dtmf-relay sip-notify`	Forwards DTMF tones using SIP NOTIFY messages.
Step 3	**exit** **Example:** `Router(config-dial-peer)# exit`	Exits dial-peer configuration mode.
Step 4	**sip-ua** **Example:** `Router(config)# sip-ua`	Enables SIP user-agent configuration mode.
Step 5	**notify telephone-event max-duration** *time* **Example:** `Router(config-sip-ua)# notify telephone-event max-duration 2000`	Configures the maximum time interval allowed between two consecutive NOTIFY messages for a single DTMF event. • **max-duration** *time*—Time interval between consecutive NOTIFY messages for a single DTMF event, in milliseconds. Range is from 500 to 3000. Default is 2000.
Step 6	**exit** **Example:** `Router(config-sip-ua)# exit`	Exits SIP user-agent configuration mode.

Troubleshooting Tips

The **show sip-ua status** command output displays the time interval between consecutive NOTIFY messages for a telephone event. In the following example, the time interval is 2000 ms.

```
Router# show sip-ua status

SIP User Agent Status
SIP User Agent for UDP :ENABLED
SIP User Agent for TCP :ENABLED
SIP User Agent bind status(signaling):DISABLED
SIP User Agent bind status(media):DISABLED
SIP early-media for 180 responses with SDP:ENABLED
SIP max-forwards :6
SIP DNS SRV version:2 (rfc 2782)
NAT Settings for the SIP-UA
Role in SDP:NONE
Check media source packets:DISABLED
Maximum duration for a telephone-event in NOTIFYs:2000 ms
```

```
SIP support for ISDN SUSPEND/RESUME:ENABLED
Redirection (3xx) message handling:ENABLED

SDP application configuration:
 Version line (v=) required
 Owner line (o=) required
 Timespec line (t=) required
 Media supported:audio image
 Network types supported:IN
 Address types supported:IP4
 Transport types supported:RTP/AVP udptl
```

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