



Cisco Survivable Remote Site Telephony Version 2.1

Cisco Software Release 12.2(15)T

March 17, 2003

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Preface

This preface discusses the objectives, audience, organization, and conventions of this document. It also provides sources for obtaining documentation from Cisco Systems.

Documentation Objectives

This document describes the tasks and commands necessary to configure and maintain Cisco Survivable Remote Site (SRS) Telephony for systems with Cisco CallManager V3.0.5 and higher.

Audience

This document is intended primarily for users who configure and maintain Cisco SRS Telephony but who may not be familiar with the tasks, the relationship between tasks, or the Cisco IOS software commands necessary to perform particular tasks. This configuration guide is also intended for those users experienced with Cisco SRS Telephony who need to know about new features, new configuration options, and new software characteristics in the current Cisco IOS software release.

Documentation Organization

This document consists of the chapters listed in Table 1.

Table 1 Chapter Contents of Cisco SRS Telephony Document

	Title	Contents
Chapter 1	Cisco SRS Telephony Overview	Product description, history, and reference information
Chapter 2	Cisco SRS Telephony Configuration	<p>Prerequisites, restrictions, background information about Cisco SRS Telephony, and descriptions of how to perform basic, required configuration, and additional configuration tasks for the following features:</p> <ul style="list-style-type: none"> • Unmatched dial-peer routing • Cisco IP phone date and time display formats • Phone setting based on MAC addresses • Keepalive intervals • Default destinations for incoming calls • Global prefixes • Call transfers from Cisco IP phones to other phone numbers • Trunk access codes • Message button phone numbers • Class of restriction (COR) on the dial peers associated with directory numbers • Call forwarding during a busy signal or no answer • Translation rules for numbers dialed on Cisco IP phones • Interdigit timeout value for all Cisco IP phones attached to the router • Music on hold • Dial-peer hunting • Additional language options for IP phone display
Chapter 3	Command Reference	Cisco SRS Telephony commands

- “Cisco SRS Telephony Configuration,” which Documentation Conventions

Within Cisco IOS software documentation, the term *router* is generally used to refer to a variety of Cisco products (for example, routers, access servers, and switches). Routers, access servers, and other networking devices that support Cisco IOS software are shown interchangeably within examples. These products are used only for illustrative purposes; that is, an example that shows one product does not necessarily indicate that other products are not supported.

The Cisco IOS documentation set uses the following conventions:

Convention	Description
^ or Ctrl	The ^ and Ctrl symbols represent the Control key. For example, the key combination ^D or Ctrl-D means hold down the Control key while you press the D key. Keys are indicated in capital letters but are not case sensitive.
<i>string</i>	A string is a nonquoted set of characters shown in italics. For example, when setting an SNMP community string to public, do not use quotation marks around the string or the string will include the quotation marks.

Command syntax descriptions use the following conventions:

Convention	Description
boldface	Boldface text indicates commands and keywords that you enter literally as shown.
<i>italics</i>	Italic text indicates arguments for which you supply values.
[x]	Square brackets enclose an optional element (keyword or argument).
	A vertical line indicates a choice within an optional or required set of keywords or arguments.
[x y]	Square brackets enclosing keywords or arguments separated by a vertical line indicate an optional choice.
{x y}	Braces enclosing keywords or arguments separated by a vertical line indicate a required choice.

Nested sets of square brackets or braces indicate optional or required choices within optional or required elements. For example:

Convention	Description
[x {y z}]	Braces and a vertical line within square brackets indicate a required choice within an optional element.

Examples use the following conventions:

Convention	Description
screen	Examples of information displayed on the screen are set in Courier font.
boldface screen	Examples of text that you must enter are set in Courier bold font.
< >	Angle brackets enclose text that is not printed to the screen, such as passwords.
!	An exclamation point at the beginning of a line indicates a comment line. (Exclamation points are also displayed by the Cisco IOS software for certain processes.)
[]	Square brackets enclose default responses to system prompts.

The following conventions are used to attract the attention of the reader:



Caution

Means *reader be careful*. In this situation, you might do something that could result in equipment damage or loss of data.

**Note**

Means *reader take note*. Notes contain helpful suggestions or references to materials not contained in this manual.

**Timesaver**

Means the *described action saves time*. You can save time by performing the action described in the paragraph.

Obtaining Documentation

These sections explain how to obtain documentation from Cisco Systems.

World Wide Web

You can access the most current Cisco documentation on the World Wide Web at this URL:

<http://www.cisco.com>

Translated documentation is available at this URL:

http://www.cisco.com/en/US/swassets/sw274/sitewide_country_language_selector.html

Documentation CD-ROM

Cisco documentation and additional literature are available in a Cisco Documentation CD-ROM package, which is shipped with your product. The Documentation CD-ROM is updated monthly and may be more current than printed documentation. The CD-ROM package is available as a single unit or through an annual subscription.

Ordering Documentation

You can order Cisco documentation in these ways:

- Registered Cisco.com users (Cisco direct customers) can order Cisco product documentation from the Networking Products MarketPlace:
http://www.cisco.com/cgi-bin/order/order_root.pl
- Registered Cisco.com users can order the Documentation CD-ROM through the online Subscription Store:
<http://www.cisco.com/go/subscription>
- Nonregistered Cisco.com users can order documentation through a local account representative by calling Cisco Systems Corporate Headquarters (California, U.S.A.) at 408 526-7208 or, elsewhere in North America, by calling 800 553-NETS (6387).

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San Jose, CA 95134-9883

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Cisco.com

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- Resolve technical issues with online support
- Download and test software packages
- Order Cisco learning materials and merchandise
- Register for online skill assessment, training, and certification programs

If you want to obtain customized information and service, you can self-register on Cisco.com. To access Cisco.com, go to this URL:

<http://www.cisco.com>

Technical Assistance Center

The Cisco Technical Assistance Center (TAC) is available to all customers who need technical assistance with a Cisco product, technology, or solution. Two levels of support are available: the Cisco TAC Web Site and the Cisco TAC Escalation Center.

Cisco TAC inquiries are categorized according to the urgency of the issue:

- Priority level 4 (P4)—You need information or assistance concerning Cisco product capabilities, product installation, or basic product configuration.
- Priority level 3 (P3)—Your network performance is degraded. Network functionality is noticeably impaired, but most business operations continue.



Cisco SRS Telephony Overview

This chapter provides an overview that describes what Cisco SRS Telephony is and what it does. In addition is information regarding IP phone, platform, and Cisco CallManager Version support; specifications; restrictions; and where to find additional reference documents.

Contents

- Cisco SRS Telephony Description, page 1-1
- Cisco IP Phone, Platform, Cisco CallManager, Signal, and Switch Support, page 1-4
- Specifications, page 1-7
- Features, page 1-9
- Restrictions, page 1-14
- Additional References, page 1-16

Cisco SRS Telephony Description

The SRS Telephony feature provides Cisco CallManager (CCM) with fallback support for Cisco IP phones attached to a Cisco router on your local network. The SRS Telephony feature enables routers to provide call-handling support for Cisco IP phones when they lose connection to remote primary, secondary, or tertiary CCM installations, or when the WAN connection is down.

CCM supports Cisco IP phones at remote sites attached to Cisco multiservice routers across the WAN. Prior to the SRS Telephony feature, when the WAN connection between a router and CCM failed, or connectivity with CCM was lost for some reason, Cisco IP phones on the network became unusable for the duration of the failure. The SRS Telephony feature overcomes this problem and ensures that the Cisco IP phones offer continuous (yet, minimal) service by providing call-handling support for Cisco IP phones directly from the SRS Telephony 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 registered with the router. When the WAN link or connection to the primary CCM is restored, call-handling reverts back to the primary CCM.

When Cisco IP phones lose contact with primary, secondary, and tertiary CCMs, they must establish a connection to a local SRS Telephony router in order to ensure call-processing capability necessary to place and receive calls. The Cisco IP phone retains the IP address of the local SRS Telephony router as a default router in the Network Configuration area of the Settings menu. This list supports a maximum

of five default router entries; however, CCM accommodates a maximum of three entries. When a secondary CCM is not available on the network, the local SRS Telephony router's IP address is retained as the standby connection for CCM during normal operation.

When the WAN link fails, calls in progress are sustained for the duration of the call. Calls in transition and calls that have not yet connected are dropped and must be reinitiated once Cisco IP phones reestablish connection to their local SRS Telephony router. Telephone service remains unavailable from the time connection to the remote CCM is lost until the Cisco IP phone establishes connection to the SRS Telephony router.

**Note**

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

The time taken to reestablish connection to a remote CCM depends in part on the keepalive period set by CCM, itself. Typically, it takes three times the keepalive period for a phone to discover that its connection to CCM has failed. The default keepalive period is 30 seconds. If the phone has an active standby connection established with an SRS Telephony router, the fallback process takes 10 to 20 seconds after connection with CCM is lost. An active standby connection to an SRS Telephony router exists only if the phone has the location of a single CCM in its CallManager list. Otherwise, the phone activates a standby connection to its secondary CCM.

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

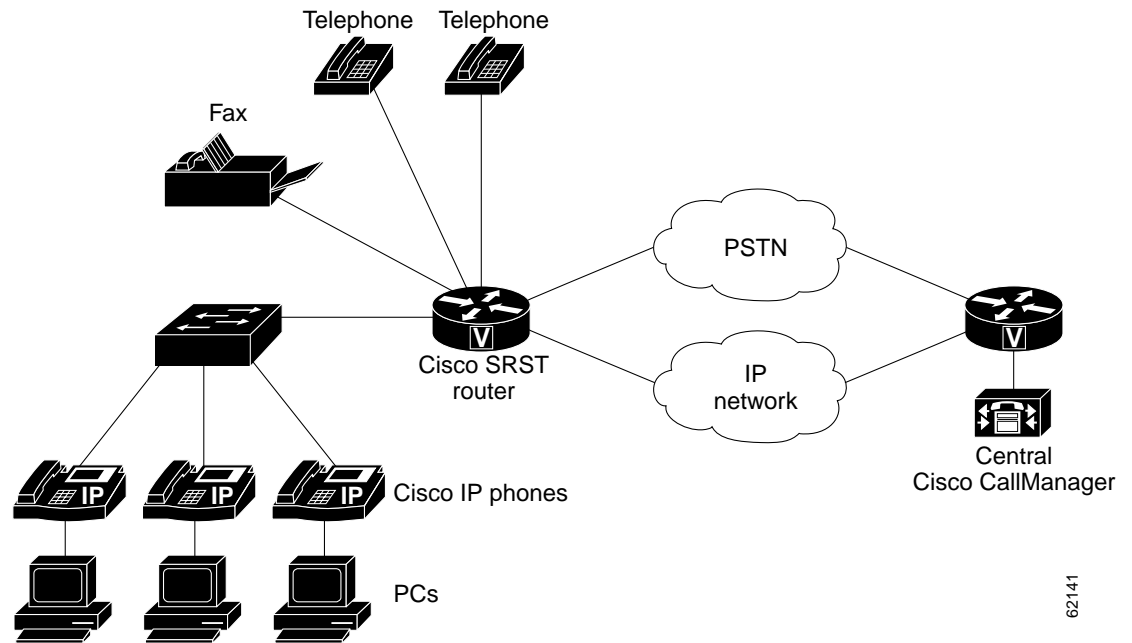
**Note**

During a WAN connection failure, when SRS Telephony is enabled, Cisco IP phones display a message informing you that they are operating in CCM 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 CCM fallback mode. When the CCM is restored, the message goes away and full Cisco IP phone functionality is restored.

While in CallManager fallback mode, Cisco IP phones periodically attempt to reestablish a connection with CCM at the central office. When a connection is reestablished with CCM, Cisco IP phones automatically cancel their registration with the SRS Telephony router. A Cisco IP phone cannot reestablish a connection with the primary CCM at the central office if it is currently engaged in an active call.

Figure 1-1 shows a branch office with several Cisco IP phones connected to an SRS Telephony router. The router features connections to both a WAN link and the public switched telephone network (PSTN). The Cisco IP phones connect to their primary CCM at the central office via this WAN link.

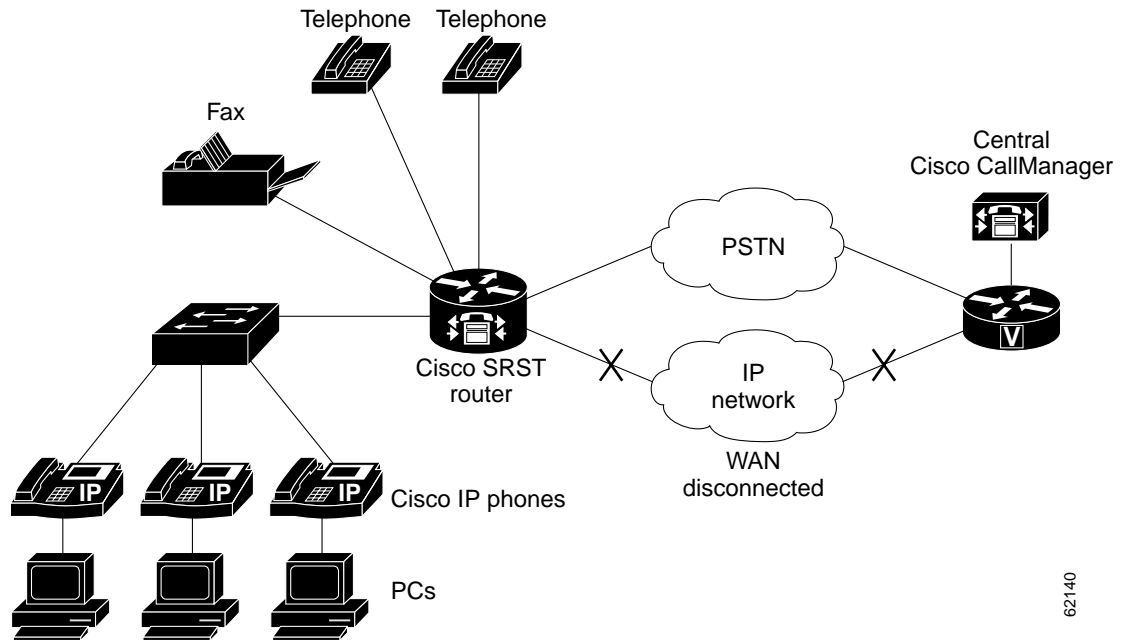
Figure 1-1 Branch Office Cisco IP Phones Connected to a Remote Central CCM



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Figure 1-2 shows the same branch office telephone network with the WAN connection down. In this situation, the Cisco IP phones use the SRS Telephony router as a fallback for their primary CCM. The branch office Cisco IP phones are connected to the PSTN through the SRS Telephony router and are able to make and receive “off-net” calls.

Figure 1-2 Branch Office Cisco IP Phones Operating in SRS Telephony Mode



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Cisco IP Phone, Platform, Cisco CallManager, Signal, and Switch Support

The following sections provide information about Cisco SRST Version 2.1:

- Finding Cisco IOS Software Releases that Support Cisco SRST, page 1-5
- Cisco IP Phone Support, page 1-5
- Platform Support, page 1-6
- Cisco CallManager Compatibility, page 1-6
- Signal Support, page 1-7
- Switch Support, page 1-7

Finding Cisco IOS Software Releases that Support Cisco SRST

Cisco SRST 2.1 was introduced in Cisco IOS release 12.2(11)YT and subsequently incorporated into Cisco IOS release 12.2(15)T. Other Cisco IOS software releases may inherit versions of Cisco SRST subsequently. 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

The following IP phones are supported by Cisco SRST 2.1:

Table 1-1 Cisco SRST 2.1 P Phone Support

Cisco SRST Version	Cisco IOS Release ¹	Supported Cisco IP Phones
Version 2.1	12.2(15)T 12.2(11)YT	<ul style="list-style-type: none"> • Cisco Analog Telephone Adaptors (ATA) 186 and Cisco ATA 188 Version 2.16 and higher with Cisco Call Manager 3.3 and higher² • Cisco IP Phone 7905 and Cisco IP Phone 7905G • Cisco IP Phone 7910 • Cisco IP Phone Expansion Module 7914 • Cisco IP Conference Station 7935 • Cisco IP Phone 7940 and Cisco IP Phone 7940G • Cisco IP Phone 7960 and Cisco IP Phone 7960G • Cisco VG248 Analog Phone Gateway Version 1.2(1) and higher³

Table 1-1 Cisco SRST 2.1 P Phone Support (continued)

Cisco SRST Version	Cisco IOS Release ¹	Supported Cisco IP Phones
Version 2.1	12.2(15)T1	<ul style="list-style-type: none"> • Cisco Analog Telephone Adaptor (ATA) 186 and Cisco ATA 188 Version 2.16 and higher with Cisco Call Manager 3.3 and higher² • Cisco IP Phone 7902G • Cisco IP Phone 7905 and Cisco IP Phone 7905G • Cisco IP Phone 7910 • Cisco IP Phone 7912G • Cisco IP Phone Expansion Module 7914 • Cisco IP Conference Station 7935 • Cisco IP Phone 7940 and Cisco IP Phone 7940G • Cisco IP Phone 7960 and Cisco IP Phone 7960G • Cisco VG248 Analog Phone Gateway Version 1.2(1) and higher³

1. Includes subsequent rebuilds (T1, T2, T3, and so forth) that support Cisco SRST.
2. Cisco SRST supports Cisco ATA 186 and Cisco ATA 188 using Skinny Client Control Protocol (SCCP) for voice calls only.
3. During Cisco CallManager fallback, Cisco SRST considers the Cisco VG248 to be a group of Cisco IP phones. Each of the 48 ports on the Cisco VG248 is treated by Cisco SRST as a separate Cisco IP phone. For more information, see the “Cisco IP Phone Expansion Module 7914 Support” section on page 1-14.

Platform Support

See the “Specifications” section on page 1-7.

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, refer to the online release notes or, if supported, Cisco Feature Navigator.

Cisco CallManager Compatibility

See the Cisco Call Manager Compatibility Matrix.

Signal Support

Cisco SRST 2.1 supports T1 and E1 signals.

Switch Support

Cisco SRST 2.1 supports all Primary Rate Interface (PRI) and Basic Rate Interface (BRI) switches are supported, including the following:

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

Specifications

Table 2 provides information about the maximum number of Cisco IP Phones, maximum number of directory numbers (DNs) or virtual voice ports, and memory requirements for Cisco SRST 2.1.



Note

Although Cisco IOS can provide higher number of DN's for some of these platforms, the higher limitation may not apply to your platform due to memory restrictions. We recommend that you follow the guidelines in the Specifications tables to configure your network.

Table 2 Specifications for Cisco SRST 2.1

Cisco Platform	Maximum Cisco IP Phones	Maximum DN's or Virtual Voice Ports	Minimum DRAM	Recommended DRAM ¹	Minimum Flash	Recommended Flash
Cisco 1751 router ²	24	120	64 MB	64 MB	16 MB	32 MB
Cisco 1760 router	24	120	64 MB	64 MB	16 MB	32 MB

Table 2 Specifications for Cisco SRST 2.1 (continued)

Cisco Platform	Maximum Cisco IP Phones	Maximum DNs or Virtual Voice Ports	Minimum DRAM	Recommended DRAM ¹	Minimum Flash	Recommended Flash
Cisco 2600-XM routers ³	24	120	96 MB	128 MB	32 MB	—
Cisco 2650 and Cisco 2651 routers	48	192	96 MB	128 MB	16 MB	32 MB
Cisco 2650-XM and Cisco 2651-XM routers	48	192	96 MB	128 MB	32 MB	—
Cisco 2691 router	72	288	96 MB	128 MB	32 MB	—
Cisco 3640 router and Cisco 3640A router	72	288	96 MB	128 MB	32 MB	—
Cisco 3660 router	240	960	96 MB	128 MB	16 MB	32 MB
Cisco 3725 router	144	576	128 MB	—	32 MB	—
Cisco 3745 router	240	960	128 MB	—	32 MB	—
Cisco 7200 routers NPE-225 ⁴	200	800	256 MB	256 MB	32 MB	32 MB
Cisco 7200 routers NPE-300 and NPE-400 ⁴	240	960	256 MB	256 MB	32 MB	32 MB
Cisco 7200 routers NPE-400 ⁴	480	960	512 MB	512 MB	32 MB	32 MB
Cisco 7200 routers G1 ⁴	480	960	512 MB	512 MB	32 MB	32 MB

1. These DRAM recommendations were tested in Cisco IOS release 12.2(8)T for the maximum number of IP phones and virtual ports listed in this table. The DRAM recommendations for the number of IP phones and virtual ports listed have not been verified for subsequent Cisco IOS releases that have inherited Cisco SRST. Cisco SRST configurations running post Cisco IOS release 12.2(8)T software may require additional memory.
2. The Cisco 1750 platform does not support quality of service (QoS) features on the asymmetric digital subscriber line (ADSL) link, Cisco Hoot and Holler over IP applications, and G.SHDSL WAN card supported in the current.
3. Cisco 2610-XM, Cisco 2611-XM, Cisco 2620-XM, and Cisco 2621-XM.
4. These DRAM recommendations were tested in Cisco IOS release 12.2(13)T for the maximum number of IP phones and virtual ports listed in this table. The DRAM recommendations for the number of IP phones and virtual ports listed have not been verified for subsequent Cisco IOS releases that have inherited Cisco SRST.

Features

Table 1-3 provides a summary of the features in the order in which they were introduced.

Table 1-3 Feature History from Cisco SRS Telephony V1.0 to Present Version

Release	Cisco SRST Version	Modification
12.1(5)YD	Version 1.0	<ul style="list-style-type: none"> • Cisco SRS Telephony introduced on the Cisco 2600 series and Cisco 3600 series multiservice routers, and Cisco IAD2420 series integrated access devices • Cisco IP phones able to establish a connection with an SRS Telephony router in the event of a WAN link to Cisco CallManager failure • Graying out of all Cisco IP phone function keys that are not supported during SRS Telephony operation • Extension-to-extension dialing • Direct Inward Dialing (DID) • Direct Outward Dialing (DOD) • Calling party ID (Caller ID/ANI) display • Calling party name display • Last number redial • Preservation of local extension-to-extension calls when WAN link fails • Preservation of local extension to Public Switched Telephone Network (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
12.1(5)YD1	Version 1.0	Support was added for 144 Cisco IP phones on the Cisco 3660 multiservice routers.

Table 1-3 Feature History from Cisco SRS Telephony V1.0 to Present Version

12.2(2)XT	Version 2.0	<ul style="list-style-type: none"> • Cisco SRS Telephony 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
12.2(8)T	Version 2.0	Cisco SRS Telephony was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745 routers, and Cisco MC3810-V3 concentrators.
12.2(8)T1	Version 2.0	Cisco SRS Telephony was implemented on the Cisco 2600-XM and Cisco 2691 routers.
12.2(11)T	Version 2.01	<ul style="list-style-type: none"> • Cisco SRS Telephony was implemented on the Cisco 1760 routers and support for 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
12.2(13)T	Version 2.02	<ul style="list-style-type: none"> • Unity Voice Mail Integration Using In-Band DTMF Signaling across the PSTN, page 1-12 • Increase in Directory Number Maximums, page 1-12. • Cisco IP Phone Conference Station 7935 Support, page 1-11 • Cisco SRS Telephony was implemented on the Cisco Catalyst 4224 access gateway switches, Cisco Catalyst 4000 family access gateway modules, Cisco Catalyst 4224 access gateway switch, and Cisco 7200 routers (NPE-225, NPE-300, and NPE400). See the Specifications for Cisco SRST 2.1, page 1-7 • Support was removed for the Cisco MC3810-V3 concentrator.

Table 1-3 Feature History from Cisco SRS Telephony V1.0 to Present Version

12.2(11)YT	Version 2.1	<ul style="list-style-type: none"> • Additional Language Options for IP Phone Display, page 1-13 • Cisco IP Phone 7905G Support • Cisco IP Phone Expansion Module 7914 Support, page 1-14 • dialplan-pattern Command Enhancement, page 1-14 • Support for Cisco VG248 Analog Phone Gateway Version 1.2(1) and Higher
12.2(15)T		—
12.2(15)T1		<ul style="list-style-type: none"> • Cisco IP Phone 7902G Support • Cisco IP Phone 7912G Support

Information about New Features in Cisco SRS Telephony V.2.02

Cisco SRS Telephony Version 2.1 introduces the new features described in the following sections:

- Cisco IP Phone Conference Station 7935 Support
- Increase in Directory Number Maximums
- Unity Voice Mail Integration Using In-Band DTMF Signaling across the PSTN

Cisco IP Phone Conference Station 7935 Support

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

The Cisco 7935 full-duplex design offers superior voice quality, eliminating echoes, clipped words, and reverberations for more natural conversation. It features superior sound quality with a digitally tuned speaker and three microphones, allowing conference participants to move around while speaking. In addition to the regular telephony keypad, 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 Number Maximums

Directory numbers were increased for the platforms shown in Table 1-4.

Table 1-4 *Directory Number Changes 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 2600-XM	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

Unity voice mail is integrated with Cisco SRS Telephony, introducing six new commands:

- pattern direct, page 3-56
- pattern ext-to-ext busy, page 3-58
- pattern ext-to-ext no-answer, page 3-60
- pattern trunk-to-ext busy, page 3-62
- pattern trunk-to-ext no-answer, page 3-64
- vm-integration, page 3-97

For further information, see the “Configuring DTMF Patterns on the Router” section on page 2-9.

Information about New Features in Cisco SRS Telephony V.2.1

Cisco SRS Telephony Version 2.1 introduces the new features described in the following sections:

- Additional Language Options for IP Phone Display
- Cisco IP Phone 7902G Support
- Cisco IP Phone 7905G Support
- Cisco IP Phone 7912G Support
- Cisco IP Phone Expansion Module 7914 Support
- dialplan-pattern Command Enhancement
- 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 7940 and Cisco IP Phone 7960 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 SRS Telephony running under Cisco CallManager V3.2.

For configuration information, see the “Configuring Cisco SRS Telephony Optional Settings” section on page 2-4.

Cisco IP Phone 7902G Support

The Cisco IP Phone 7902G is an entry-level IP phone addressing 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 CiscoIPPhones, 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](http://www.cisco.com) and click Products & Service, IP Phone, Cisco 7900 Series IP Phones, Product Literature, and 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 cost-effective, basic IP phone providing a core set of business features. It 7905G 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. This capability gives the network administrator centralized power control, which translates into greater network availability.

No configuration is necessary.

For more information, refer to 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 & Service, IP Phone, Cisco 7900 Series IP Phones, Product Literature, and Data Sheets.

Cisco IP Phone Expansion Module 7914 Support

The Cisco IP Phone 7914 Expansion Module attaches to your Cisco IP Phone 7960, adding 14 line appearances and/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 and/or speed dial numbers, or a total of 34 line appearances and/or speed dial numbers.

No configuration is necessary.

For more information, refer to the Cisco IP Phone 7914 Expansion Module Quick Start Guide.

dialplan-pattern Command Enhancement

A new keyword has been added to the **dial-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 "dialplan-pattern" section on page 3-39.

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), which 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.

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.

Restrictions

Table 1-5 provides a history of restrictions from Cisco SRS Telephony Version 1.0 to the present version.

Table 1-5 Restrictions History from Cisco SRS Telephony V1.0 to Present Version

Cisco SRS Telephony Version	Cisco IOS Release	Restrictions
Version 1.0	12.1(5)YD	<ul style="list-style-type: none"> This feature 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. This feature does not support Centralized Automatic Message Accounting (CAMA) trunks on the Cisco 3660 routers. <p>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, please contact your local Cisco account team for help in understanding and meeting the CAMA regulatory requirements.</p>
	12.1(5)YD1	
	12.2(2)XB	
	12.2(2)XG	
Version 2.0	12.2(2)XT	<ul style="list-style-type: none"> All of the restrictions in Cisco SRS Telephony Version 1.0
Version 2.0	12.2(8)T	<ul style="list-style-type: none"> Call transfer is supported only on the following: <ul style="list-style-type: none"> Voice over IP (VoIP) H.323, Voice over Frame Relay (VoFR), and Voice over ATM (VoATM) between Cisco gateways running Cisco IOS Release 12.2(11)T and using the H.323 nonstandard information element FXO and FXS loopstart (analog) FXO and FXS groundstart (analog) E&M (analog) and DID (analog) T1 CAS with FX0 and FXS groundstart signalling T1 CAS with E&M signalling All PRI and BRI switch types Graying out of all Cisco IP phone function keys that are not supported during SRS Telephony operation <ul style="list-style-type: none"> CFwdAll (call forward all) MeetMe PickUp GPickUp (group pickup) Park Confrn (conference) Although the Cisco IAD2420 series IADs support the SRS Telephony feature, this feature is not recommended as a solution for enterprise branch offices. The Cisco 1750 and Cisco 1751 platforms do not support quality of service (QoS) features on an asymmetric digital subscriber line (ADSL) link, Cisco Hoot and Holler over IP applications, and G.SHDSL WAN card supported in the current Cisco 1700 image sets.
Version 2.0	12.2(8)T1	
Version 2.01	12.2(11)T	
Version 2.02	12.2(13)T	
Version 2.1	12.2(11)YT	
Version 2.1	12.2(15)T	

Additional References

The following sections provide additional references related to Cisco SRS Telephony Version 2.02:

- Related Documents, page 1-16
- Standards, page 1-16
- MIBs, page 1-17
- RFCs, page 1-17
- Technical Assistance, page 1-17

Related Documents

Related Topic	Documents
Cisco IP phones	<ul style="list-style-type: none"> • <i>Getting Started with the Cisco IP Phone 7910</i> • <i>Cisco IP Phone 7914 Expansion Module Quick Start Guide</i> • Cisco IP Conference Station 7935 documents • <i>At a Glance Cisco IP Phone 7960 and 7940 Series</i> • <i>Cisco IP Phone 7960 and 7940 Series User Guide</i>
Command reference and configuration information for voice and telephony commands	<ul style="list-style-type: none"> • <i>Cisco IOS Voice, Video, and Fax Command Reference, Release 12.2 T</i> • <i>Cisco IOS Voice, Video, and Fax Configuration Guide, Release 12.2</i> • <i>Cisco IOS Debug Command Reference, Release 12.2</i>
Cisco CallManager user documentation	<ul style="list-style-type: none"> • <i>Cisco CallManager</i>
Dynamic Host Control Protocol (DHCP)	<ul style="list-style-type: none"> • <i>Cisco IOS DHCP Server</i>

Standards

Standards ¹	Title
No new or modified standards are supported by this feature, and support for existing standards has not been modified by this feature.	—

1. Not all supported standards are listed.

MIBs

MIBs ¹	MIBs Link
No new or modified MIBs are supported by this feature, and support for existing standards has not been modified by this feature.	To obtain lists of supported MIBs by platform and Cisco IOS release, and to download MIB modules, go to the Cisco MIB website on Cisco.com at the following URL: http://www.cisco.com/public/sw-center/netmgmt/cmtk/mibs.shtml

1. Not all supported MIBs are listed.

To locate and download MIBs for selected platforms, Cisco IOS releases, and feature sets, use Cisco MIB Locator found at the following URL:

<http://tools.cisco.com/ITDIT/MIBS/servlet/index>

If Cisco MIB Locator does not support the MIB information that you need, you can also obtain a list of supported MIBs and download MIBs from the Cisco MIBs page at the following URL:

<http://www.cisco.com/public/sw-center/netmgmt/cmtk/mibs.shtml>

To access Cisco MIB Locator, 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>

RFCs

RFCs ¹	Title
No new or modified RFCs are supported by this feature, and support for existing RFCs has not been modified by this feature.	—

1. Not all supported RFCs are listed.

Technical Assistance

Description	Link
Technical Assistance Center (TAC) home page, containing 30,000 pages of searchable technical content, including links to products, technologies, solutions, technical tips, tools, and lots more. Registered Cisco.com users can log in from this page to access even more content.	http://www.cisco.com/public/support/tac/home.shtml



Cisco SRS Telephony Configuration

This chapter explains the required and optional tasks for configuring Cisco SRS Telephony.

Contents

- Prerequisites for SRS Telephony Configuration, page 2-1
- How to Configure Cisco SRS Telephony, page 2-2
- Monitoring and Maintaining SRS Telephony, page 2-11
- Configuration Examples, page 2-12

Prerequisites for SRS Telephony Configuration

The following are prerequisites that must be met before configuration:

- The Cisco SRS Telephony system must be configured with the correct IP phones, platforms, and Cisco CallManager (CCM) version. See the “Cisco IP Phone, Platform, Cisco CallManager, Signal, and Switch Support” section on page 1-4
- The correct memory requirements must be installed. Memory requirements are dependent on the platform and the number of supported Cisco IP phones. See the “Specifications” section on page 1-7 for details.
- The correct IP phone firmware must be installed. With CCM V3.1 and later, Cisco SRS Telephony is standard in IP phone firmware. Refer to CCM documentation for firmware requirements. For CCM 3.0.5, you must use the firmware version P003E302 or P004E302.
- IP routing must be enabled.
- The SRS Telephony router must be configured as the default router for the Cisco IP phones.



Note You must purchase a feature license to turn on this new feature. You also need an account on Cisco.com to access the Cisco IP phone load versions.

How to Configure Cisco SRS Telephony

This section contains the following procedures:

- Configuring SRS Telephony on Routers to Support IP Phone Functions, page 2-2 (required)
- Configuring Cisco SRS Telephony Optional Settings, page 2-4 (optional)
- Configuring Cisco SRS Telephony for Unity Voice-Mail Integration, page 2-9 (optional)

Configuring SRS Telephony on Routers to Support IP Phone Functions



Tip

When the SRS Telephony feature is enabled, Cisco IP phones do not need to be reconfigured while in Cisco CallManager fallback mode because phones retain the same configuration that was used with Cisco CallManager.

To configure SRS Telephony on the routers 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*
4. **max-ephones** *max-phones*
5. **limit-dn** {**7910** | **7940** | **7960**} *max-lines*

DETAILED STEPS

	Command	Purpose
Step 1	<code>call-manager-fallback</code>	Enables SRS Telephony feature support and enters Cisco CallManager fallback mode.
	Example: Router(config)#	
Step 2	<code>ip source-address ip-address [port port] [any-match strict-match]</code>	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.
	Example: Router(config-cm-fallback)# <code>ip source-address 10.6.21.4 port 2002 strict-match</code>	

	Command	Purpose
Step 3	<p>max-dn <i>max-directory-numbers</i></p> <p>Example: Router(config-cm-fallback)# max-dn 12</p>	<p>Sets the maximum number of directory numbers or virtual voice ports that can be supported by the router. The default is 0. The maximum number is platform dependent. See the “Specifications” section on page 1-7 for further details.</p> <p>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.</p>
Step 4	<p>max-ephones <i>max-phones</i></p> <p>Example: Router(config-cm-fallback)# max-ephones 24</p>	<p>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 “Specifications” section on page 1-7 for further details.</p> <p>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.</p>
Step 5	<p>limit-dn {7910 7940 7960} <i>max-lines</i></p> <p>Example: Router(config-cm-fallback)# limit-dn 7910 2</p>	<p>Limits the directory number lines on Cisco IP phones during CallManager fallback mode.</p> <p>Note You must configure this command during initial SRS Telephony router configuration, before any phone actually registers with the SRS Telephony router. However, you can modify the number of lines at a later time.</p> <p>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 last line number greater than this limit, warning information is displayed for phone reset.</p>

Verifying Cisco SRS Telephony

To verify that the SRS Telephony feature is enabled, perform the following steps:

- Step 1 Enter the **show run** command to verify the configuration.
- Step 2 Enter the **show call-manager-fallback all** command to verify that SRS Telephony 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 SRS Telephony router.
- Step 4 To temporarily block the TCP port 2000 Skinny Client Control Protocol (SCCP) connection for one of the Cisco IP phones to force the Cisco IP phone to lose its connection to the Cisco CallManager and register with the SRS Telephony 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 SRS Telephony 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 SRS Telephony router.
- d. Use the **show ephone** command to display the Cisco IP phones that have registered to the SRS Telephony router.

Troubleshooting Tips

To troubleshoot the SRS Telephony feature, perform the following steps:

-
- | | |
|--------|---|
| Step 1 | To set keepalive debugging for the Cisco IP phone, use the debug ephone keepalive command. |
| Step 2 | To set registration debugging for the Cisco IP phone, use the debug ephone register command. |
| Step 3 | To set state debugging for the Cisco IP phone, use the debug ephone state command. |
-

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

Configuring Cisco SRS Telephony Optional Settings

Although following baseline Cisco SRS Telephony settings are not required, they are worth considering for possible configuration of the following features:

- Unmatched dial-peer routing
- Cisco IP phone date and time display formats
- Keepalive intervals
- Default destinations for incoming calls
- Global prefixes
- Call transfers from Cisco IP phones to other phone numbers
- Trunk access codes
- Message button phone numbers
- Class of restriction (COR) on the dial peers associated with directory numbers
- Call forwarding during a busy signal or no answer
- Translation rules for numbers dialed on Cisco IP phones
- Interdigit timeout value for all Cisco IP phones attached to the router
- Music on hold
- Dial-peer hunting
- Additional language options for IP phone display

SUMMARY STEPS

1. **call-manager-fallback**
2. **alias** *tag number-pattern to alternate-number preference preference-value*
3. **date-format** {*mm-dd-yy* | *dd-mm-yy*}
4. **time-format** {*12* | *24*}
5. **default-destination** *telephone number*
6. **keepalive** *seconds*
7. **dialplan-pattern** *tag prefix-pattern extension-length length* [**no-reg**]
8. **transfer-pattern** *transfer-pattern*
9. **access-code** {**{fxo | e&m}** *dial-string* | **{bri | pri}** *dial-string*} [**direct-inward-dial**]
10. **voicemail** *phone-number*
11. **cor** {**incoming** | **outgoing**} *cor-list-name* {*cor-list-number starting-number - ending-number* | **default**}
12. **call-forward busy** *directory-number*
13. **call-forward noan** *directory-number timeout seconds*
14. **translate** {**called** | **calling**} *translation-rule-tag*
15. **timeouts interdigit** *seconds*
16. **moh** *filename*
17. **user-locale** *country-code*

DETAILED STEPS

	Command	Purpose
Step 1	call-manager-fallback Example: Router(config)#	Enables SRS Telephony feature support and enters CallManager fallback mode.
Step 2	alias <i>tag number-pattern to alternate-number preference preference-value</i> Example: Router(config-cm-fallback)# alias 1 60.. to 5001 preference 2	(Optional) Allows routing of unmatched call destination to specific extension numbers with an associated dial-peer preference parameter.
Step 3	date-format { <i>mm-dd-yy</i> <i>dd-mm-yy</i> }	(Optional) Sets the date display format on all Cisco IP phones registered with the router. The default is mm-dd-yy.
Step 4	time-format { <i>12</i> <i>24</i> }	(Optional) Sets the time display format on all Cisco IP phones registered with the router. The default is 12 hours.
	Example: Router(config-cm-fallback)# time-format 24	

	Command	Purpose
Step 5	<p><code>keepalive seconds</code></p> <p>Example: Router(config-cm-fallback)# <code>keepalive 60</code></p>	(Optional) Configures the time interval between Cisco IP phones keepalive messages sent to the router when SRS Telephony is enabled. The range is from 10 to 65535 seconds. The default timeout is 30 seconds.
Step 6	<p><code>default-destination telephone number</code></p> <p>Example: Router(config-cm-fallback)# <code>default-destination 40802</code></p>	(Optional) Assigns the default destination number for incoming telephone calls.
Step 7	<p><code>dialplan-pattern tag prefix-pattern</code> <code>extension-length length [no-reg]</code></p> <p>Example: Router(config-cm-fallback)# <code>dialplan-pattern 1 40855550.. extension-length 4 no-reg</code></p>	<p>(Optional) Creates a global prefix that can be used to expand the abbreviated extension numbers into fully qualified E.164 numbers. The extension-length keyword enables the system to convert a full E.164 telephone number back to an extension number for the purposes of caller-ID display, received, and missed call lists.</p> <p>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 some specific numbers to the gatekeeper so that those numbers can be used for other telephony services.</p>
Step 8	<p><code>transfer-pattern transfer-pattern</code></p> <p>Example: Router(config-cm-fallback)# <code>transfer-pattern 52540..</code></p>	(Optional) Allows telephone call transfer from Cisco IP phones to other phone numbers both within the local IP network and outside of the local IP network.
Step 9	<p><code>access-code {{fxo e&m} dial-string {bri pri} dial-string} [direct-inward-dial]</code></p> <p>Example: Router(config-cm-fallback)# <code>access-code e&m 8</code></p>	<p>(Optional) Configures trunk access codes for each type of line—Basic Rate Interface (BRI), E&M, Foreign Exchange Office (FXO), and Primary Rate Interface (PRI)—so that the Cisco IP phones can access the trunk lines while in Cisco CallManager fallback mode.</p> <p>The <i>dial-string</i> argument is used to set up temporary dial peers for each specified line type. The direct-inward-dial keyword enables you to set Direct Inward Dialing (DID) access for PRI and BRI trunk lines.</p> <p>Note The access-code command creates temporary dial peers in Cisco CallManager fallback mode. In many cases, you may already have the local PSTN ports configured with appropriate access codes provided by dial peers (for example, dial 9 to select an FXO PSTN line), in which case this command is not needed.</p>
Step 10	<p><code>voicemail phone-number</code></p> <p>Example: Router(config-cm-fallback)# <code>voicemail 914085551000</code></p>	(Optional) Configures the telephone number that is dialed when the message button on a Cisco IP phone is pressed.

	Command	Purpose
Step 11	<pre>cor {incoming outgoing} cor-list-name {cor-list-number starting-number - ending-number default}</pre> <p>Example: Router(config-cm-fallback)# cor outgoing LockforPhoneC 1 5010 - 5020</p>	<p>(Optional) Configures a COR on the dial peers associated with directory numbers. COR is used to specify which incoming dial peer can use which outgoing dial peer to make a call. Each dial peer can be provisioned with an incoming and an outgoing COR list.</p> <p>The <i>cor-list-number</i> argument ranges from 1 to 6.</p> <p>The <i>starting-number - ending-number</i> argument provides a range of directory numbers.</p> <p>The default keyword instructs the router to use an existing default COR list.</p>
Step 12	<pre>call-forward busy directory-number</pre> <p>Example: Router(config-cm-fallback)# call-forward busy 50..</p>	<p>(Optional) Configures call forwarding to another number when the Cisco IP phone is busy.</p> <p>Note The E.164 number you enter can contain one or more "." wildcard characters. The wildcard characters correspond to the right-justified digits in the directory number extension.</p>
Step 13	<pre>call-forward noan directory-number timeout seconds</pre> <p>Example: Router(config-cm-fallback)# call-forward noan 5005 timeout 10</p>	<p>(Optional) Configures call forwarding to another number when no answer is received from the Cisco IP phone.</p> <p>The timeout keyword sets the waiting time, in seconds, before the call is forwarded to another phone. The <i>seconds</i> range is from 3 to 6000 seconds.</p> <p>Note The E.164 number you enter can contain one or more "." wildcard characters. The wildcard characters correspond to the right-justified digits in the directory number extension.</p>
Step 14	<pre>translate {called calling} translation-rule-tag</pre> <p>Example: Router(config-cm-fallback) translate called 20</p>	<p>(Optional) Applies a translation rule to numbers dialed by Cisco IP phone users. The called keyword applies the translation rule to the outbound called party number. The calling keyword applies the translation rule to the inbound called party number.</p> <p>The <i>translation-rule-tag</i> argument is the reference number of the translation rule. Valid entries are from 1 to 2147483647. For further details, refer to the "Configuration Dial Plans, Dial Peers, and Digit Manipulation" chapter of the <i>Cisco IOS Voice, Video, and Fax Configuration Guide</i>, Release 12.2.</p>

Command	Purpose
<p>Step 15 <code>timeouts interdigit seconds</code></p> <p>Example: <pre>Router(config-cm-fallback)# timeouts interdigit 5</pre></p>	<p>(Optional) Configures the interdigit timeout value for all Cisco IP phones attached to the router. The interdigit timeout specifies the number of seconds that the system waits after the caller has entered the initial digit or a subsequent digit of the dialed string. If the timeout ends before the destination is identified, a tone sounds and the call ends.</p> <p>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 “message URL Configuration Dial Plans, Dial Peers, and Digit Manipulation” chapter of the <i>Cisco IOS Voice, Video, and Fax Configuration Guide</i>, Release 12.2.</p> <p>The <i>seconds</i> argument is the interdigit timeout wait time in seconds. A valid entry is an integer from 2 to 120 seconds. The default is 10 seconds.</p>
<p>Step 16 <code>moh filename</code></p> <p>Example: <pre>Router(config-cm-fallback)# moh minuet.wav</pre></p>	<p>(Optional) Configures music on hold. This feature supports .au and .wav format music files. Music on hold works only for G.711 calls and on-net VoIP and PSTN calls. For all other calls, callers hear a tone. Internal calls between Cisco IP phones do not get music on hold, instead, callers hear a tone.</p> <p>Note The music on hold file can be in .wav or .au file format; however, the file format must contain 8-bit 8 kHz data, for example, Consultative Committee for International Telegraph and Telephone (CCITT) a-law or u-law data format.</p>
<p>Step 17 <code>user-locale country-code</code></p> <p>Example: <pre>Router(config-cm-fallback)# user-locale ES</pre></p>	<p>Selects a language for display on the Cisco IP Phone7940 and Cisco IP Phone 7960.</p> <ul style="list-style-type: none"> • If you have Cisco CallManager V3.2, enter one of the following ISO-3166 codes: <ul style="list-style-type: none"> – Denmark—DE – France—FR – Germany—DE – Italy—IT – The Netherlands—NL – Norway—NO – Portugal—PT – Spain—ES – Sweden—SE – United States—US (default) • If you have software prior to Cisco CallManager V3.2, the only <i>country-code</i> option is US (United States).

Troubleshooting Tips

- To set detail debugging for the Cisco IP phones, use the **debug ephone detail** command.
- To set error debugging for the Cisco IP phones, use the **debug ephone error** command.
- To set call statistics debugging for the Cisco IP phones, use the **debug ephone statistics** command.
- To provide voice packet level debugging and print 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*.

Configuring Cisco SRS Telephony for Unity Voice-Mail Integration

For dual tone multifrequency (DTMF) integrations, information on how to route incoming or forwarded calls is sent by the telephone system in the form of DTMF digits. The DTMF digits are in the form of a pattern and depend on the voice-mail system connected to the Cisco SRS Telephony router. These patterns are required for the DTMF integration with most voice-mail systems. The DTMF integration configuration on the Cisco SRS Telephony router works with any analog voice-mail system. Voice-mail systems are designed to respond to DTMF after the system has answered the incoming calls. The tasks described in the following sections are required:

- Configuring DTMF Patterns on the Router, page 2-9 (required)
- Configuring Integration Files on Legacy Voice-Mail Systems, page 2-11 (required)



Note

FXO hairpin forwarded calls to voice mail must have disconnect supervision from the central office.

Configuring DTMF Patterns on the Router

The Cisco SRS Telephony router provides flexibility for the integration with any legacy voice-mail system. You can configure multiple tags and tokens for each pattern, depending on the voice-mail system and type of access. The *tag* in the configuration pattern must match the number defined in the voice-mail system's integration file to identify the type of call. The keywords—**CGN** (calling number), **CDN** (called number), and **FDN** (forwarding number)—define the type of call information sent to the voice-mail system.

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	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 system.
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. <ul style="list-style-type: none"> The <i>tag</i> attribute is an 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. The Cisco SRS Telephony router supports a maximum of four tags. The keywords—CGN, CDN, and FDN—configure the type of call information sent to the voice-mail system, such as calling number (CGN), called number (CDN), or forwarding number (FDN).
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-integration) 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.
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-integration) 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.

	Command	Purpose
Step 5	<pre>pattern trunk-to-ext busy tag1 {CGN CDN FDN} [tag2 {CGN CDN FDN}] [tag3 {CGN CDN FDN}] [last-tag]</pre> <p>Example: Router(config-vm-integration) pattern trunk-to-ext busy 6 FDN * CGN *</p>	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.
Step 6	<pre>pattern trunk-to-ext no-answer tag1 {CGN CDN FDN} [tag2 {CGN CDN FDN}] [tag3 {CGN CDN FDN}] [last-tag]</pre> <p>Example: Router(config-vm-integration) pattern trunk-to-ext no-answer 4 FDN * CGN *</p>	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.

Although it is unlikely that you will use multiple instances of the **CGN**, **CDN**, or **FDN** keyword in a single command line, it is permissible to do so.

Configuring Integration Files on Legacy Voice-Mail Systems

To configure the integration files for a legacy voice-mail system, follow the instructions in the voice-mail system's analog voice mail integration configuration guide or recommended documents. You must design the DTMF integration patterns appropriately, so that the voice-mail system and the Cisco SRS Telephony router work with each other. For a configuration example, see the "Configuring Cisco SRS Telephony for Unity Voice-Mail Integration" section on page 2-9.

Monitoring and Maintaining SRS Telephony

To monitor and maintain the router with SRS Telephony feature, use the following commands in EXEC mode:

Command	Purpose
Router# show run	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 SRS Telephony router.
Router# show call-manager-fallback dial-peer	Displays the output of the dial peers of the SRS Telephony router.
Router# show call-manager-fallback ephone-dn	Displays Cisco IP phone destination number.
Router# show call-manager-fallback voice-port	Displays output for the voice ports.
Router# show ephone dn dn-tag	Displays Cisco IP phone destination number.
Router# show ephone 7910	Displays Cisco 7910 phone status.
Router# show ephone 7940	Displays Cisco 7940 phone status.
Router# show ephone 7960	Displays Cisco 7960 phone status.
Router# show ephone offhook	Displays Cisco IP phone status for all phones that are off hook.


```
ip dhcp pool 2600
  network 10.2.0.0 255.255.0.0
  option 150 ip 10.0.0.1
  default-router 10.0.0.1
!
no ip dhcp-client network-discovery
lcp max-session-starts 0
!
!
!
translation-rule 1
  Rule 0 85... 919785
!
translation-rule 2
  Rule 0 408734.... 4
!
!
!
interface FastEthernet0/0
  ip address 10.0.0.2 255.255.0.0
  duplex auto
  speed auto
!
interface FastEthernet0/1
  ip address 10.0.0.1 255.255.0.0
  duplex auto
  speed auto
!
router eigrp 100
  network 10.0.0.0
  auto-summary
  no eigrp log-neighbor-changes
!
ip classless
ip route 0.0.0.0 0.0.0.0 10.0.0.1
ip http server

snmp-server packetsize 4096
snmp-server manager
call rsvp-sync
!
voice-port 1/1/0
!
voice-port 1/1/1
!
mgcp modem passthrough voip mode ca
no mgcp timer receive-rtcp
!
mgcp profile default
!
dial-peer cor custom
  name call911
  name call1800
  name call1900
!
!
dial-peer cor list allowall
  member call911
  member call1800
  member call1900
!
!
!
```

```

dial-peer cor list allow1800
  member call1800
!
!
dial-peer cor list alloww1800and1900
  member call1800
  member call1900
!
dial-peer cor list allow1900
  member call1900

!
dial-peer voice 1 voip
  destination-pattern 919715....
  translate-outgoing called 2
  session target ipv4:10.0.0.5
!
dial-peer voice 2 voip
  shutdown
  destination-pattern 6....T
  session target ipv4:10.0.0.6
  codec g711ulaw
!
dial-peer voice 3 voip
  destination-pattern 65087.....
  session target ipv4:10.0.0.7
  codec g711ulaw
!
dial-peer voice 90 voip
  corlist outgoing allow1900
  destination-pattern 9000
  session target ipv4:10.0.0.8
!
dial-peer voice 45 pots
  destination-pattern 9
  port 1/1/0

!
call-manager-fallback
  ip source-address 10.0.0.1 port 2000
  max-ephones 10
  max-dn 10
  dialplan-pattern 1 408735.... extension-length 4 no-reg
  dialplan-pattern 2 919785.... extension-length 4 no-reg
  voicemail 4001
  no huntstop
  alias 2 3... to 5555
  translate called 1
  call-forward busy 5001
  call-forward noan 5001 timeout 8
  cor incoming allowall default
  cor incoming allowall 1 4000 - 4999
  cor incoming allowall 2 4000 - 5000
  moh minuet.au
  time-format 12
  date-format mm-dd-yy
  transfer-pattern 1...
  transfer-pattern 2...
  keepalive 30
  interdigit timeout 5
!

```

```

line con 0
line aux 0
line vty 0 4
  login
line vty 5 15
  login
!
!
end

```

Voice-Mail Integration Configuration Example

The examples in this section show how to configure analog voice-mail integration. They include configuration examples for local and central voice-mail systems.

Local Voice-Mail System Example

The “Dial-Peer Configuration for Integration for Voice-mail System” section of the example shows a legacy dial-peer configuration for a local voice-mail system. The “Cisco SRS Telephony Voice-mail Integration Pattern Configuration” part is a compatible Cisco SRS Telephony configuration.

```

! Dial-Peer Configuration for Integration for Voice-mail System
!
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 A14012
  port 3/1/1
!
! Cisco SRS Telephony configuration
!
call-manager-fallback
  max-ephones 24
  max-dn 144
  ip source-address 1.4.214.104 port 2000
  voicemail 14011
!
! Cisco SRS Telephony 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 *

```

Central Location Voice-Mail System Example

The “Dial-Peer Configuration for Integration with Voice-Mail System” section of the example shows a legacy dial-peer configuration for a central voice-mail system. The “Cisco SRS Telephony Voice-mail Integration Pattern Configuration” section is a compatible Cisco SRS Telephony configuration.


Note

MWI integration is not supported for PSTN access to voice-mail systems at a central locations.

```

! Dial-Peer Configuration for Integration with Voice-Mail System
! located in central location
!
dial-peer voice 101 pots
 destination-pattern 14011
 port 3/0/0
!
! Cisco SRS Telephony configuration
!
call-manager-fallback
 max-ephones 24
 max-dn 144
 ip source-address 1.4.214.104 port 2000
 voicemail 14011
!
! Cisco SRS Telephony 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 *

```



Command Reference

This section documents Cisco SRS Telephony commands. The following are new and modified commands in Cisco SRS Telephony for Cisco IOS Software Release 12.2(15)T:

- **dialplan-pattern (modified)**
- **user-locale (new)**



Note

For commands not specific to Cisco SRS Telephony, refer to the Cisco IOS Release 12.2 command reference publications.

The following is a list of all of the Cisco SRS Telephony commands.

- **access-code**
- **alias**
- **call-forward busy**
- **call-forward noan**
- **call-manager-fallback**
- **cor**
- **date-format**
- **debug ephone alarm**
- **debug ephone detail**
- **debug ephone error**
- **debug ephone keepalive**
- **debug ephone pak**
- **debug ephone raw**
- **debug ephone register**
- **debug ephone state**
- **debug ephone statistics**
- **default-destination**
- **huntstop**
- **ip source-address**
- **keepalive**

- **limit-dn**
- **max-dn**
- **max-ephones**
- **moh**
- **pattern direct**
- **pattern ext-to-ext busy**
- **pattern ext-to-ext no-answer**
- **pattern trunk-to-ext busy**
- **pattern trunk-to-ext no-answer**
- **pattern direct**
- **show call-manager-fallback all**
- **show call-manager-fallback dial-peer**
- **show call-manager-fallback ephone-dn**
- **show call-manager-fallback voice-port**
- **show ephone**
- **show ephone-dn**
- **show ephone summary**
- **time-format**
- **timeouts interdigit**
- **transfer-pattern**
- **translate**
- **user-locale**
- **vm-integration**
- **voicemail**

access-code

To configure trunk access codes for each type of line so that the Cisco IP phones can access the trunk lines only during Cisco CallManager fallback mode when the SRS Telephony feature is enabled, use the **access-code** command in CallManager fallback configuration mode. To remove the telephone access code configuration from the Cisco IP phones, use the **no** form of this command.

- **access-code** { **fxo** | **e&m** } *dial-string*
- **no access-code** { **fxo** | **e&m** } [*dial-string*]
- **access-code** { **bri** | **pri** } *dial-string* [**direct-inward-dial**]
- **no access-code** { **bri** | **pri** } [*dial-string*] [**direct-inward-dial**]

Syntax Description		
fxo		Enables a foreign exchange office (FXO) interface.
e&m		Enables an analog ear and mouth (E&M) interface.
<i>dial-string</i>		Sets up dial access codes for each specified line type by creating dial peers.
bri		Enables a BRI interface.
pri		Enables a PRI interface.
direct-inward-dial		(Optional) Enables direct-inward-dial on a POTS dial peer.

Defaults No default behavior or values.

Command Modes CallManager fallback configuration

Command History	Release	Modification
	12.1(5)YD	This command was introduced on the following platforms: Cisco 2600 series and Cisco 3600 series multiservice routers, and Cisco IAD2420 series IADs.
	12.2(2)XT	This command was implemented on Cisco 1750 and Cisco 1751 multiservice routers.
	12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers.
	12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
	12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers. The direct-inward-dial keyword was added.

Usage Guidelines

The **access-code** command configures trunk access codes for each type of line—BRI, E&M, FXO, and PRI—so that the Cisco IP phones can access the trunk lines in Cisco CallManager fallback mode when the SRS Telephony feature is enabled. This provides system-wide access.

**Note**

The **access-code** command creates temporary dial peers in Cisco CallManager fallback mode. In many cases, you may already have the local PSTN ports configured with appropriate access codes provided by dial peers (for example, dial 9 to select an FXO PSTN line), in which case this command is not needed.

The **access-code** command creates temporary POTS voice dial peers for all the selected types of voice ports, during Cisco CallManager fallback mode. Use this command only if your normal network dial-plan configuration prevents you from configuring permanent POTS voice dial peers to provide trunk access for use in the fallback mode. When the **access-code** command is used, it is important to ensure that all ports covered by the command have valid trunk connections. Selection between ports for outgoing calls is random.

The dial string is used to set up temporary dial peers for each specified line type. If there are multiple lines of the same type, a dial peer is set up for each line. The dial peers are active only during Cisco CallManager fallback mode when the SRS Telephony feature is enabled. The result of this configuration is that all PSTN interfaces of the same type, for example BRI, are treated as equivalent, and any port may be selected to place the outgoing PSTN call. The **direct-inward-dial** keyword enables you to set direct-inward-dial access for PRI and BRI trunk lines.

Examples

The following example sets the **access-code** command for BRI 8:

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# access-code bri 8 direct-inward-dial
```

The following example sets the **access-code** command for E&M 8:

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# access-code e&m 8
```

The following example sets the **access-code** command for FXO 9:

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# access-code fxo 9
```

The following example sets the **access-code** command for PRI 9:

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# access-code pri 9 direct-inward-dial
```

Related Commands

Command	Description
call-manager-fallback	Enables SRS Telephony feature support and enters CallManager fallback configuration mode.

alias

To provide a mechanism for rerouting calls to telephone numbers that are unavailable during Cisco CallManager fallback, use the **alias** command in call-manager-fallback configuration mode. To disable rerouting of unmatched call destination calls, use the **no** form of this command.

alias *tag number-pattern* **to** *alternate-number* [**preference** *preference-value*] [**huntstop**]

no alias *tag number-pattern* **to** *alternate-number* [**preference** *preference-value*] [**huntstop**]

Syntax Description

<i>tag</i>	Identifier for alias rule range. The range is from 1 to 10.
<i>number-pattern</i>	Pattern to match the incoming telephone number. This pattern may include wildcards.
to	Connects the tag number pattern to the alternate number.
<i>alternate-number</i>	Alternate telephone number to route incoming calls to match the number pattern. This has to be a specific extension that belongs to an IP phone active registered on the Cisco SRST router.
preference	(Optional) Assigns a dial-peer preference value to the alias.
<i>preference-value</i>	(Optional) Preference value of the associated dial peer from 0 to 10.
huntstop	(Optional) Stops call hunting after trying the <i>alternate-number</i> .

Defaults

No default behavior or values.

Command Modes

Call-manager-fallback configuration

Command History

Cisco IOS Release	Modification
12.2(2)XT	This command was introduced on the following platforms: Cisco 1750, Cisco 1751, Cisco 2600 series and Cisco 3600 series multiservice routers, and Cisco IAD2420 series IADs.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers.
12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers. The preference keyword was added.

Usage Guidelines

The **alias** command provides a mechanism for rerouting calls to telephone numbers that are unavailable during fallback. An alias is activated when extension numbers are registered that fall within the range of the command's *alternate-number* argument.

The **alias** command supports all port types.

You can use a **to** *alternate-number* value once only. You can not put the same **to** *alternate-number* value into multiple **alias** command statements.

Examples

In the following example, alias 1 is configured to route calls to extensions 6000 through 6099 to extension 5001 using a dial-peer with a preference value of 2. Extensions 6000 through 6099 are a subset of IP phones without fallback service. During fallback, calls to these extension are routed to 5001.

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# alias 1 60.. to 5001 preference 2
```

Related Commands

Command	Description
call-manager-fallback	Enables Cisco SRST feature support and enters call-manager-fallback configuration mode.
default-destination	Assigns a default destination number for incoming telephone calls on the Cisco SRST router.

call-forward busy

To configure call forwarding to another number when a Cisco IP phone is busy, use the **call-forward busy** command in CallManager fallback configuration mode. To disable call forwarding, use the **no** form of this command.

call-forward busy *directory-number*

no call-forward busy [*directory-number*]

Syntax Description	<i>directory-number</i>	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.
---------------------------	-------------------------	---

Defaults	No default behavior or values.
-----------------	--------------------------------

Command Modes	CallManager fallback configuration
----------------------	------------------------------------

Command History	Release	Modification
	12.2(2)XT	This command was introduced on the following platforms: Cisco 1750, Cisco 1751, Cisco 2600 series and Cisco 3600 series multiservice routers, and Cisco IAD2420 series IADs.
	12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers.
	12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
	12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.

Usage Guidelines	The call-forward busy command configures call forwarding to another number when a Cisco IP phone is busy. The call forwarding mechanism is applied globally to all phones that register during fallback.
-------------------------	---

Examples	The following example forwards calls to extension number 5005 when an incoming call reaches a busy IP phone extension number:
-----------------	---

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# call-forward busy 5005
```

The following example forwards calls to any available extension number in the 50xx bank of extensions when an incoming call reaches a busy IP phone extension number:

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# call-forward busy 50..
```

**Note**

You can forward an incoming VoIP call only to destination numbers local to the router. VoIP calls can not be forwarded to an alternate (on-net) VoIP destination.

Related Commands

Command	Description
call-forward noan	Configures call-forwarding to another number when no answer is received from the Cisco IP phone.
call-manager-fallback	Enables SRS Telephony feature support and enters CallManager fallback configuration mode.

call-forward noan

To configure call forwarding to another number when no answer is received from a Cisco IP phone, use the **call-forward noan** command in CallManager fallback configuration mode. To disable call forwarding, use the **no** form of this command.

call-forward noan *directory-number* **timeout** *seconds*

no call-forward noan [*directory-number*]

Syntax Description

<i>directory-number</i>	Selected directory number. Represents 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	Ring no answer timeout duration. It is the waiting time before the call is forwarded to another phone. The time is set in seconds.
<i>seconds</i>	Time (in seconds) before call forwarding starts. The range is from 3 to 60,000 seconds.

Defaults

No default behavior or values.

Command Modes

CallManager fallback configuration

Command History

Release	Modification
12.2(2)XT	This command was introduced on the following platforms: Cisco 1750, Cisco 1751, Cisco 2600 series and Cisco 3600 series multiservice routers, and Cisco IAD2420 series IADs.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers.
12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.

Usage Guidelines

The **call-forward noan** command configures call forwarding to another number when no answer is received from a Cisco IP phone. The call-forwarding mechanism is applied globally to all phones that register during fallback. The **timeout** keyword sets the waiting time before the call is forwarded to another phone. The time is set in seconds. The range is from 3 to 60,000 seconds.

Examples

The following example shows how to set call forwarding of incoming calls to directory number 5005 when line 1, directory number 5001, does not answer. The **timeout** before the call is forwarded to the directory number 5005 is set for 10 seconds:

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# call-forward noan 5005 timeout 10
```

The following example shows how to set call forwarding of incoming calls to an available extension in the 50xx bank of extensions when line 1, directory number 5001, does not answer. The timeout period before the call is forwarded to the directory number 5005 is set for 10 seconds:

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# call-forward noan 50.. timeout 10
```

**Note**

An incoming VoIP call can be forwarded only to destination numbers local to the router. VoIP calls cannot be forwarded to an alternate (on-net) VoIP destination.

Related Commands

Command	Description
call-forward busy	Configures call forwarding to another number when a Cisco IP phone is busy.
call-manager-fallback	Enables SRS Telephony feature support and enters CallManager fallback configuration mode.

call-manager-fallback

To enable SRS Telephony support and enter CallManager fallback configuration mode, use the **call-manager-fallback** command in global configuration mode. To disable SRS Telephony support, use the **no** form of this command.

call-manager-fallback

no call-manager-fallback

Syntax Description This command has no arguments or keywords.

Defaults No default behavior or values.

Command Modes Global configuration

Command History

Release	Modification
12.1(5)YD	This command was introduced on the following platforms: Cisco 2600 series and Cisco 3600 series multiservice routers, and Cisco IAD2420 series IADs.
12.2(2)XT	This command was implemented on Cisco 1750 and Cisco 1751 multiservice routers.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers.
12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.

Examples

The following example shows how to enter the CallManager fallback configuration mode:

```
Router(config)# call-manager-fallback
```

The resulting router prompt is `Router(config-cm-fallback)#`.

Related Commands

Command	Description
access-code	Configures trunk access codes for each type of line so that the Cisco IP phones can access the trunk lines.
alias	Provides a mechanism for servicing calls to telephone numbers that are unavailable during CallManager fallback.
call-forward busy	Configures call forwarding to another number when a Cisco IP phone is busy.
call-forward noan	Configures call forwarding to another number when no answer is received from a Cisco IP phone.
cor	Configures COR on the dial peers associated with directory numbers.
default-destination	Assigns a default destination number for incoming telephone calls.
dialplan-pattern	Creates a global prefix that can be used to expand the abbreviated extension numbers into fully qualified E.164 numbers.
huntstop	Sets huntstop for the dial peers associated with a Cisco IP phone lines.
ip source-address	Enables the router to receive messages from a Cisco IP phones through the specified IP addresses and ports.
keepalive	Configures the time interval between sending keepalive messages to the router used by a Cisco IP phones.
max-dn	Sets the maximum number of directory numbers or virtual voice ports that can be supported by the router.
max-ephone	Configures the maximum number of Cisco IP phones that can be supported by the router.
reset	Resets a Cisco IP phones.
timeouts interdigit	Configures the interdigit timeout value for all Cisco IP phones attached to the router.
transfer-pattern	Allows transfer of telephone calls by Cisco IP phones to other phone numbers.
translate	Applies a translation rule to modify the phone number dialed by any Cisco IP phone user during the CallManager fallback mode.
voicemail	Configures the telephone number that is speed-dialed when the message button on a Cisco IP phone is pressed.

COR

To configure a class of restriction (COR) on the dial peers associated with directory numbers, use the **cor** command in call-manager-fallback configuration mode. To disable COR associated with directory numbers, use the **no** form of this command.

```
cor {incoming | outgoing} cor-list-name {cor-list-number starting-number - ending-number |
default}
```

```
no cor cor-list-name cor-list-number
```

Syntax Description		
incoming		COR list to be used by incoming dial peers.
outgoing		COR list to be used by outgoing dial peers.
<i>cor-list-name</i>		COR list name.
<i>cor-list-number</i>		COR list identifier. The maximum number of COR lists that can be created is six, 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.
<i>starting-number - ending-number</i>		Directory number range; for example, 2000 - 2025.
default		Instructs the COR list to assume behavior according to a predefined default COR list.

Defaults No default behavior or values.

Command Modes Call-manager-fallback configuration

Command History	Cisco IOS Release	Modification
	12.2(2)XT	This command was introduced on the following platforms: Cisco 1750, Cisco 1751, Cisco 2600 series and Cisco 3600 series multiservice routers, and Cisco IAD2420 series IADs.
	12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers.
	12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
	12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers. The default keyword was added.

Usage Guidelines The **cor** command sets the dial-peer COR parameter for dial peers associated with the directory numbers created during Cisco CallManager fallback. A list-based mechanism is provided to assign COR to specific sets of directory numbers during Cisco CallManager fallback. The 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.

COR is used to specify which incoming dial peer can use which outgoing dial peer to make a call. Each dial peer can be provisioned with an incoming and an outgoing COR list.

A default COR is assigned to the directory numbers that do not match any COR list number or number range. The assigned COR is invoked for the dial peers automatically created for each directory number during Cisco CallManager fallback registration.

You can have up to 20 COR lists for each incoming and outgoing calls. 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 dia-peer (for incoming calls) and it is a super set or 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 to be incoming or outgoing. If you hook up a phone to an FXS port on an Cisco SRST router and try to make a call from that phone, the call will considered to be an incoming call to the router and voice port. If you make a call to the FXS phone, then the call will be considered to be outgoing.

By default, an incoming call leg has the highest COR priority. The outgoing COR list has the lowest. If there is no COR configuration for incoming calls on a dial peer, then 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 1 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, then 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 (super set 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 COR list for outgoing calls on the outgoing dial peer
COR list applied for incoming calls (sub set 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.

Examples

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

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# cor incoming LockforPhoneC 1 5002 - 5010
```

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

```
Router(config)# call-manager-fallback
Router(config-cm-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:

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# cor outgoing 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:

```
Router(config)# dial-peer cor custom
Router(config-dp-cor)# name 911
Router(config-dp-cor)# name 1800
Router(config-dp-cor)# name 1900
Router(config-dp-cor)# name local_call
```

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

```
Router(config)# dial-peer cor list call911
Router(config-dp-corlist)# member 911
```

```
Router(config)# dial-peer cor list call1800
Router(config-dp-corlist)# member 1800
```

```
Router(config)# dial-peer cor list call1900
Router(config-dp-corlist)# member 1900
```

```
Router(config)# dial-peer cor list calllocal
Router(config-dp-corlist)# member local_call
```

```
Router(config)# dial-peer cor list engineering
Router(config-dp-corlist)# member 911
Router(config-dp-corlist)# member local_call
```

```
Router(config)# dial-peer cor list manager
Router(config-dp-corlist)# member 911
Router(config-dp-corlist)# member 1800
Router(config-dp-corlist)# member 1900
Router(config-dp-corlist)# member local_call
```

```
Router(config)# dial-peer cor list hr
Router(config-dp-corlist)# member 911
Router(config-dp-corlist)# member 1800
Router(config-dp-corlist)# 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.

```
Router(config)# dial-peer voice 1 voip
Router(config-dial-peer)# destination pattern 734...
Router(config-dial-peer)# session target ipv4:1.1.1.1
Router(config-dial-peer)# cor outgoing calllocal
```

```

Router(config)# dial-peer voice 2 voip
Router(config-dial-peer)# destination pattern 1800.....
Router(config-dial-peer)# session target ipv4:1.1.1.1
Router(config-dial-peer)# cor outgoing call1800

Router(config)# dial-peer voice 3 pots
Router(config-dial-peer)# destination pattern 1900.....
Router(config-dial-peer)# port 1/0/0
Router(config-dial-peer)# cor outgoing call1900

Router(config)# dial-peer voice 4 pots
Router(config-dial-peer)# destination pattern 911
Router(config-dial-peer)# port 1/0/1
Router(config-dial-peer)# cor outgoing call911

Router(config)# dial-peer voice 5 pots
Router(config-dial-peer)# destination pattern 316...
Router(config-dial-peer)# port 1/1/0 ; No cor is applied.

```

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

```

Router(config)# call-manager-fallback
Router(config-cm-fallback)# max-conferences 8
Router(config-cm-fallback)# cor incoming engineering 1 1001 - 1001
Router(config-cm-fallback)# cor incoming hr 2 1002 - 1002
Router(config-cm-fallback)# cor incoming manager 3 1003 - 1008

```

The example configuration allows for the following:

- Extension 1001 to call 408... numbers, 911 and 316....
- Extension 1002 to call 408..., 1800 numbers, 911 and 316....
- Extension 1003 through 1008 to call all of the possible Cisco SRST router numbers
- All extensions to call 316....

Related Commands

Command	Description
call-manager-fallback	Enables Cisco SRST feature support and enters call-manager-fallback configuration mode.
corlist-incoming	Specifies the COR list to be used when a specified dial peer acts as the incoming dial peer.
corlist-outgoing	Specifies the COR list to be used by outgoing dial peers.
dial-peer cor list	Defines a COR list name.

date-format

To set the date display format on all the Cisco IP phones attached to the router, use the **date-format** command in CallManager fallback configuration mode. To display the date in the default format, use the **no** form of this command.

```
date-format { mm-dd-yy | dd-mm-yy }
```

```
no date-format { mm-dd-yy | dd-mm-yy }
```

Syntax Description	mm-dd-yy	dd-mm-yy
	Sets to month, day, and year. Each slot needs a two-digit number. This format is the default setting.	Sets to day, month, and year. Each slot needs a two-digit number.

Defaults The default is **mm-dd-yy**.

Command Modes CallManager fallback configuration

Command History	Release	Modification
	12.2(4)XT	This command was introduced on the following platforms: Cisco 1750, Cisco 1751, Cisco 2600 series and Cisco 3600 series multiservice routers, and Cisco IAD2420 series IADs.
	12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers.
	12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
	12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.

Usage Guidelines The **date-format** command sets the date display format on all the Cisco IP phones attached to the router.

Examples The following example sets the date format to date, month, and year for all affected Cisco IP phones:

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# date-format dd-mm-yy
```

Related Commands	Command	Description
	call-manager-fallback	Enables SRS Telephony feature support and enters CallManager fallback configuration mode.

debug ephone alarm

To set SkinnyStation alarm messages debugging for the Cisco IP phone, use the **debug ephone alarm** debug command in privileged EXEC mode. To disable debugging, use the **no** form of this command.

debug ephone alarm [**mac-address** *mac-address*]

no debug ephone alarm [**mac-address** *mac-address*]

Syntax Description	mac-address	(Optional) Defines the MAC address of the Cisco IP phone.
	<i>mac-address</i>	(Optional) Specifies the MAC address of the Cisco IP phone.

Command Modes	Privileged EXEC
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Command History	Release	Modification
		12.2(2)XT
	12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers.
	12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
	12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.

Usage Guidelines The **debug ephone alarm** command shows all the SkinnyStation alarm messages sent by the Cisco IP phone. Under normal circumstances, this message is sent by the Cisco IP phone just before it registers, and has the severity level for the alarm set to “Informational” and contains the reason for the phone reboot or re-register. This type of message is entirely benign and does not indicate an error condition.

If the **mac-address** keyword is not used, the **debug ephone alarm** debug command debugs all Cisco IP phones that are registered to the router. You can remove debugging for the Cisco IP phones that you do not want to debug by using the **mac-address** keyword with the **no** form of this command.

Debugging can be enabled or disabled on any number of Cisco IP phones. The Cisco IP phones that have debugging enabled can be seen by entering the **show ephone** command and looking at the debug field in the output. When debugging is enabled for a Cisco IP phone, the debug output is displayed for any Cisco IP phone directory numbers associated with the Cisco IP phone.

Examples The following example shows a SkinnyStation alarm message that is sent before the Cisco IP phone registers:

```
Router# debug ephone alarm
```

```

phone keypad reset
CM-closed-TCP
CM-bad-state

```

Related Commands

Command	Description
debug ephone detail	Sets detail debugging for the Cisco IP phone.
debug ephone error	Sets error debugging for the Cisco IP phone.
debug ephone keepalive	Sets keepalive debugging for the Cisco IP phone.
debug ephone pak	Provides voice packet level debugging and prints the contents of one voice packet in every 1024 voice packets.
debug ephone raw	Provides raw low-level protocol debugging display for all Skinny Client Control Protocol messages
debug ephone register	Sets registration debugging for the Cisco IP phone.
debug ephone state	Sets state debugging for the Cisco IP phone.
debug ephone statistics	Sets statistics debugging for the Cisco IP phone.
show debugging	Displays information about the types of debugging that are enabled for your router.

debug ephone detail

To set detail debugging for the Cisco IP phone, use the **debug ephone detail** command in privileged EXEC mode. To disable debugging, use the **no** form of this command.

debug ephone detail [**mac-address** *mac-address*]

no debug ephone detail [**mac-address** *mac-address*]

Syntax Description	mac-address	(Optional) Defines the MAC address of the Cisco IP phone.
	<i>mac-address</i>	(Optional) Specifies the MAC address of the Cisco IP phone.

Command Modes	Privileged EXEC
---------------	-----------------

Command History	Release	Modification
	12.1(5)YD	This command was introduced on the following platforms: Cisco 2600 series and Cisco 3600 series multiservice routers, and Cisco IAD2420 series IADs.
	12.2(2)XT	This command was implemented on Cisco 1750 and Cisco 1751 multiservice routers.
	12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers.
	12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
	12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.

Usage Guidelines	<p>The debug ephone detail command includes the error and state levels.</p> <p>If the mac-address keyword is not used, the debug ephone detail debug command debugs all Cisco IP phones that are registered to the router. You can remove debugging for the Cisco IP phones that you do not want to debug by using the mac-address keyword with the no form of this command.</p> <p>Debugging can be enabled or disabled on any number of Cisco IP phones. The Cisco IP phones that have debugging enabled can be seen by entering the show ephone command and looking at the debug field in the output. When debugging is enabled for a Cisco IP phone, the debug output is displayed for any Cisco IP phone directory numbers or virtual voice ports associated with the Cisco IP phone.</p>
------------------	--

Examples	<p>The following is sample output of detail debugging of the Cisco IP phone with MAC address 0030.94c3.8724. The sample is an excerpt of some of the activities that take place during call setup, connected state, active call, and the call getting disconnected:</p>
----------	---

```
Router# debug ephone detail mac-address 0030.94c3.8724
```



```
Ephone detail debugging is enabled

1d04h: ephone-1[1]:OFFHOOK
.
.
1d04h: Skinny Call State change for DN 1 SIEZE
.
.
1d04h: ephone-1[1]:SetCallState line 1 DN 1 TsOffHook
.
.
1d04h: ephone-1[1]:SetLineLamp 1 to ON
.
.
1d04h: ephone-1[1]:KeypadButtonMessage 5
.
.
1d04h: ephone-1[1]:KeypadButtonMessage 0
.
.
1d04h: ephone-1[1]:KeypadButtonMessage 0
.
.
1d04h: ephone-1[1]:KeypadButtonMessage 2
.
.
1d04h: ephone-1[1]:Store ReDial digit: 5002
.
SkinnyTryCall to 5002 instance 1
.
.
1d04h: ephone-1[1]:Store ReDial digit: 5002
1d04h: ephone-1[1]:
SkinnyTryCall to 5002 instance 1
.
.
1d04h: Skinny Call State change for DN 1 ALERTING
.
.
1d04h: ephone-1[1]:SetCallState line 1 DN 1 TsRingOut
.
.
1d04h: ephone-1[1]:SetLineLamp 1 to ON
1d04h: SetCallInfo calling dn 1 dn 1
calling [5001] called [5002]
.
.
1d04h: ephone-1[1]: party1 calling
1d04h: ephone-1[1]: party2
.
.
1d04h: SkinnyUpdateDnState by EFXS_RING_GENERATE
for DN 2 to state RINGING
.
.
1d04h: SkinnyGetCallState for DN 2 CONNECTED
.
.
1d04h: ephone-1[1]:SetLineLamp 3 to ON
1d04h: ephone-1[1]:UpdateCallState DN 1 state 4 calleddn 2
.
.
1d04h: Skinny Call State change for DN 1 CONNECTED
.
```

```

.
1d04h: ephone-1[1]:OpenReceive DN 1 codec 4:G711Ulaw64k duration 10 ms bytes 80
.
.
1d04h: ephone-1[1]:OpenReceiveChannelAck 1.2.172.21 port=20180
1d04h: ephone-1[1]:Outgoing calling DN 1 Far-ephone-2 called DN 2
1d04h: SkinnyGetCallState for DN 1 CONNECTED
.
.
1d04h: ephone-1[1]:SetCallState line 3 DN 2 TsOnHook
.
.
1d04h: ephone-1[1]:SetLineLamp 3 to OFF
.
.
1d04h: ephone-1[1]:SetCallState line 1 DN 1 TsOnHook
.
.
1d04h: ephone-1[1]:Clean Up Speakerphone state
1d04h: ephone-1[1]:SpeakerPhoneOnHook
1d04h: ephone-1[1]:Clean up activeline 1
1d04h: ephone-1[1]:StopTone sent to ephone
1d04h: ephone-1[1]:Clean Up phone offhook state
1d04h: SkinnyGetCallState for DN 1 IDLE
1d04h: called DN -1, calling DN -1 phone -1
1d04h: ephone-1[1]:SetLineLamp 1 to OFF
1d04h: UnBinding ephone-1 from DN 1
1d04h: UnBinding called DN 2 from DN 1
1d04h: ephone-1[1]:ONHOOK
1d04h: ephone-1[1]:SpeakerPhoneOnHook
1d04h: ephone-1[1]:ONHOOK NO activeline
.
.
.

```

Related Commands

Command	Description
debug ephone error	Sets error debugging for the Cisco IP phone.
debug ephone keepalive	Sets keepalive debugging for the Cisco IP phone.
debug ephone pak	Provides voice packet level debugging and prints the contents of one voice packet in every 1024 voice packets.
debug ephone raw	Provides raw low-level protocol debugging display for all Skinny Client Control Protocol messages
debug ephone register	Sets registration debugging for the Cisco IP phone.
debug ephone state	Sets state debugging for the Cisco IP phone.
debug ephone statistics	Sets statistics debugging for the Cisco IP phone.
show debugging	Displays information about the types of debugging that are enabled for your router.

debug ephone error

To set error debugging for the Cisco IP phone, use the **debug ephone error** command in privileged EXEC mode. To disable debugging, use the **no** form of this command.

debug ephone error [**mac-address** *mac-address*]

no debug ephone error [**mac-address** *mac-address*]

Syntax Description	Parameter	Description
	mac-address	(Optional) Defines the MAC address of the Cisco IP phone.
	<i>mac-address</i>	(Optional) Specifies the MAC address of the Cisco IP phone.

Command Modes Privileged EXEC

Command History	Release	Modification
	12.1(5)YD	This command was introduced on the following platforms: Cisco 2600 series and Cisco 3600 series multiservice routers, and Cisco IAD2420 series IADs.
	12.2(2)XT	This command was implemented on Cisco 1750 and Cisco 1751 multiservice routers.
	12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers.
	12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
	12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.

Usage Guidelines The **debug phone error** command cancels debugging at the detail and state level. This command is used to detect significant internal errors.

If the **mac-address** keyword is not used, the **debug ephone error** debug command debugs all Cisco IP phones that are registered to the router. You can remove debugging for the Cisco IP phones that you do not want to debug by using the **mac-address** keyword with the **no** form of this command.

Debugging can be enabled or disabled on any number of Cisco IP phones. The Cisco IP phones that have debugging enabled can be seen by entering the **show ephone** command and looking at the debug field in the output. When debugging is enabled for a Cisco IP phone, the debug output is displayed for any Cisco IP phone directory numbers or virtual voice ports associated with the Cisco IP phone.

Examples The following is sample output of error debugging for the Cisco IP phone with MAC address 0030.94c3.8724:

```
Router# debug ephone error mac-address 0030.94c3.8724
```

■ debug ephone error

```
EPHONE error debugging is enabled
```

```
socket [2] send ERROR 11
Skinny Socket [2] retry failure
```

Related Commands

Command	Description
debug ephone detail	Sets detail debugging for the Cisco IP phone.
debug ephone keepalive	Sets keepalive debugging for the Cisco IP phone.
debug ephone pak	Provides voice packet level debugging and prints the contents of one voice packet in every 1024 voice packets.
debug ephone raw	Provides raw low-level protocol debugging display for all Skinny Client Control Protocol messages
debug ephone register	Sets registration debugging for the Cisco IP phone.
debug ephone state	Sets state debugging for the Cisco IP phone.
debug ephone statistics	Sets statistics debugging for the Cisco IP phone.
show debugging	Displays information about the types of debugging that are enabled for your router.

debug ephone keepalive

To set keepalive debugging for the Cisco IP phone, use the **debug ephone keepalive** command in privileged EXEC mode. To disable debugging, use the **no** form of this command.

debug ephone keepalive [**mac-address** *mac-address*]

no debug ephone keepalive [**mac-address** *mac-address*]

Syntax Description	Parameter	Description
	mac-address	(Optional) Defines the MAC address of the Cisco IP phone.
	<i>mac-address</i>	(Optional) Specifies the MAC address of the Cisco IP phone.

Command Modes Privileged EXEC

Command History	Release	Modification
	12.1(5)YD	This command was introduced on the following platforms: Cisco 2600 series and Cisco 3600 series multiservice routers, and Cisco IAD2420 series IADs.
	12.2(2)XT	This command was implemented on Cisco 1750 and Cisco 1751 multiservice routers.
	12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers.
	12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
	12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.

Usage Guidelines The **debug ephone keepalive** command sets keepalive debugging.

If the **mac-address** keyword is not used, the **debug ephone keepalive** debug command debugs all Cisco IP phones that are registered to the router. You can remove debugging for the Cisco IP phones that you do not want to debug by using the **mac-address** keyword with the **no** form of this command.

Debugging can be enabled or disabled on any number of Cisco IP phones. The Cisco IP phones that have debugging enabled can be seen by entering the **show ephone** command and looking at the debug field in the output. When debugging is enabled for a Cisco IP phone, the debug output is displayed for any Cisco IP phone directory numbers or virtual voice ports associated with the Cisco IP phone.

Examples The following is sample output of the keepalive status for the Cisco IP phone with MAC address 0030.94C3.E1A8:

```
Router# debug ephone keepalive mac-address 0030.94c3.E1A8
```

```
EPHONE keepalive debugging is enabled for phone 0030.94C3.E1A8
```

■ debug ephone keepalive

```

1d05h: ephone-1 Set interface FastEthernet0/0 ETHERNET
1d05h: ephone-1[1]:Keepalive socket[1] SEP003094C3E1A8
1d05h: ephone-1 Set interface FastEthernet0/0 ETHERNET
1d05h: ephone-1[1]:Keepalive socket[1] SEP003094C3E1A8
1d05h: Skinny Checking for stale sockets
1d05h: ephone-1 Set interface FastEthernet0/0 ETHERNET
1d05h: ephone-1[1]:Keepalive socket[1] SEP003094C3E1A8
1d05h: ephone-1 Set interface FastEthernet0/0 ETHERNET
1d05h: ephone-1[1]:Keepalive socket[1] SEP003094C3E1A8
1d05h: Skinny active socket list (3/96): 1 2 4

```

Related Commands

Command	Description
debug ephone detail	Sets detail debugging for the Cisco IP phone.
debug ephone error	Sets error debugging for the Cisco IP phone.
debug ephone pak	Provides voice packet level debugging and prints the contents of one voice packet in every 1024 voice packets.
debug ephone raw	Provides raw low-level protocol debugging display for all Skinny Client Control Protocol messages
debug ephone register	Sets registration debugging for the Cisco IP phone.
debug ephone state	Sets state debugging for the Cisco IP phone.
debug ephone statistics	Sets statistics debugging for the Cisco IP phone.
show debugging	Displays information about the types of debugging that are enabled for your router.

debug ephone pak

To provide voice packet level debugging and to print the contents of one voice packet in every 1024 voice packets, use the **debug ephone pak** command in privileged EXEC mode. To disable debugging, use the **no** form of this command.

debug ephone pak [**mac-address** *mac-address*]

no debug ephone pak [**mac-address** *mac-address*]

Syntax Description

mac-address	(Optional) Defines the MAC address of the Cisco IP phone.
<i>mac-address</i>	(Optional) Specifies the MAC address of the Cisco IP phone.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.1(5)YD	This command was introduced on the following platforms: Cisco 2600 series and Cisco 3600 series multiservice routers, and Cisco IAD2420 series IADs.
12.2(2)XT	This command was implemented on Cisco 1750 and Cisco 1751 multiservice routers.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers.
12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.

Usage Guidelines

The **debug ephone pak** command provides voice packet level debugging and prints the contents of one voice packet in every 1024 voice packets.

If the **mac-address** keyword is not used, the **debug ephone pak** debug command debugs all Cisco IP phones that are registered to the router. You can remove debugging for the Cisco IP phones that you do not want to debug by using the **mac-address** keyword with the **no** form of this command.

Debugging can be enabled or disabled on any number of Cisco IP phones. The Cisco IP phones that have debugging enabled can be seen by entering the **show ephone** command and looking at the debug field in the output. When debugging is enabled for a Cisco IP phone, the debug output is displayed for any Cisco IP phone directory numbers or virtual voice ports associated with the Cisco IP phone.

Examples

The following is sample output of packet debugging for the Cisco IP phone with MAC address 0030.94c3.8724:

```
Router# debug ephone pak mac-address 0030.94c3.8724
```

```

EPHONE packet debugging is enabled

01:29:14: ***ph_xmit_ephone DN 3 tx_pkts 5770 dest=10.2.1.1 orig len=32
  pakcopy=0 discards 27 ip_enctype 0 0 last discard: unsupported payload type
01:29:14: to_skinny_duration 130210 offset -30 last -40 seq 0 adj 0
01:29:14: IP: 45B8 003C 0866 0000 3F11 3F90 2800 0001 0A02 0101
01:29:14: TTL 63 TOS B8 prec 5
01:29:14: UDP: 07D0 6266 0028 0000
01:29:14: sport 2000 dport 25190 length 40 checksum 0
01:29:14: RTP: 8012 16AF 9170 6409 0E9F 0001
01:29:14: is_rtp:1 is_frfl1:0 vlen:0 delta_t:160 vofr1:0 vofr2:0
scodec:11 rtp_bits:8012 rtp_codec:18 last_bad_payload 19
01:29:14: vencap FAILED
01:29:14: PROCESS SWITCH
01:29:15: %SYS-5-CONFIG_I: Configured from console by console
01:29:34: ***SkinnyPktIp DN 3 10.2.1.1 to 40.0.0.1 pkts 4880 FAST sw
01:29:34: from_skinny_duration 150910
01:29:34: nw 3BBC2A8 addr 3BBC2A4 mac 3BBC2A4 dg 3BBC2C4 dgs 2A
01:29:34: MAC: 1841 0800
01:29:34: IP: 45B8 0046 682E 0000 3E11 E0BD 0A02 0101 2800 0001
01:29:34: TTL 62 TOS B8 prec 5
01:29:34: UDP: 6266 07D0 0032 0000
01:29:34: sport 25190 dport 2000 length 50 checksum 0
01:29:34: RTP: 8012 55FF 0057 8870 3AF4 C394
01:29:34: RTP: rtp_bits 8012 seq 55FF ts 578870 ssrc 3AF4C394
01:29:34: PAYLOAD:
01:29:34: 1409 37C9 54DE 449C 3B42 0446 3AAB 182E
01:29:34: 56BC 5184 58E5 56D3 13BE 44A7 B8C4
01:29:34:
01:29:37: ***ph_xmit_ephone DN 3 tx_pkts 6790 dest=10.2.1.1 orig len=32
  pakcopy=0 discards 31 ip_enctype 0 0 last discard: unsupported payload type
01:29:37: to_skinny_duration 153870 offset -150 last -40 seq 0 adj 0
01:29:37: IP: 45B8 003C 0875 0000 3F11 3F81 2800 0001 0A02 0101
01:29:37: TTL 63 TOS B8 prec 5
01:29:37: UDP: 07D0 6266 0028 0000
01:29:37: sport 2000 dport 25190 length 40 checksum 0
01:29:37: RTP: 8012 1AAF 9173 4769 0E9F 0001
01:29:37: is_rtp:1 is_frfl1:0 vlen:0 delta_t:160 vofr1:0 vofr2:0

```

Related Commands

Command	Description
debug ephone detail	Sets detail debugging for the Cisco IP phone.
debug ephone error	Sets error debugging for the Cisco IP phone.
debug ephone keepalive	Sets keepalive debugging for the Cisco IP phone.
debug ephone raw	Provides raw low-level protocol debugging display for all Skinny Client Control Protocol messages
debug ephone register	Sets registration debugging for the Cisco IP phone.
debug ephone state	Sets state debugging for the Cisco IP phone.
debug ephone statistics	Sets statistics debugging for the Cisco IP phone.
show debugging	Displays information about the types of debugging that are enabled for your router.

debug ephone raw

To provide raw low-level protocol debugging display for all Skinny Client Control Protocol messages, use the **debug ephone raw** command in privileged EXEC mode. To disable debugging, use the **no** form of this command.

debug ephone raw [**mac-address** *mac-address*]

no debug ephone raw [**mac-address** *mac-address*]

Syntax Description

mac-address	(Optional) Defines the MAC address of the Cisco IP phone.
<i>mac-address</i>	(Optional) Specifies the MAC address of the Cisco IP phone.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.1(5)YD	This command was introduced on the following platforms: Cisco 2600 series and Cisco 3600 series multiservice routers, and Cisco IAD2420 series IADs.
12.2(2)XT	This command was implemented on Cisco 1750 and Cisco 1751 multiservice routers.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers.
12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.

Usage Guidelines

The **debug ephone raw** command provides raw low-level protocol debug display for all Skinny Client Control Protocol messages. The debug display provides byte level display of Skinny TCP socket messages.

If the **mac-address** keyword is not used, the **debug ephone raw** debug command debugs all Cisco IP phones that are registered to the router. You can remove debugging for the Cisco IP phones that you do not want to debug by using the **mac-address** keyword with the **no** form of this command.

Debugging can be enabled or disabled on any number of Cisco IP phones. The Cisco IP phones that have debugging enabled can be seen by entering the **show ephone** command and looking at the debug field in the output. When debugging is enabled for a Cisco IP phone, the debug output is displayed for any Cisco IP phone directory numbers or virtual voice ports associated with the Cisco IP phone.

Examples

The following is sample output of raw protocol debugging for the Cisco IP phone with MAC address 0030.94C3.E1A8:

```

Router# debug ephone raw mac-address 0030.94c3.E1A8

EPHONE raw protocol debugging is enabled for phone 0030.94C3.E1A8

1d05h: skinny socket received 4 bytes on socket [1]
0 0 0 0
1d05h:
1d05h: SkinnyMessageID = 0
1d05h: skinny send 4 bytes
4 0 0 0 0 0 0 0 0 0 1 0 0
1d05h: socket [1] sent 12 bytes OK (incl hdr) for ephone-(1)

1d06h: skinny socket received 4 bytes on socket [1]
0 0 0 0
1d06h:
1d06h: SkinnyMessageID = 0
1d06h: skinny send 4 bytes
4 0 0 0 0 0 0 0 0 0 1 0 0
1d06h: socket [1] sent 12 bytes OK (incl hdr) for ephone-(1)

```

Related Commands

Command	Description
debug ephone detail	Sets detail debugging for the Cisco IP phone.
debug ephone error	Sets error debugging for the Cisco IP phone.
debug ephone keepalive	Sets keepalive debugging for the Cisco IP phone.
debug ephone pak	Provides voice packet level debugging and prints the contents of one voice packet in every 1024 voice packets.
debug ephone register	Sets registration debugging for the Cisco IP phone.
debug ephone state	Sets state debugging for the Cisco IP phone.
debug ephone statistics	Sets statistics debugging for the Cisco IP phone.
show debugging	Displays information about the types of debugging that are enabled for your router.

debug ephone register

To set registration debugging for the Cisco IP phone, use the **debug ephone register** command in privileged EXEC mode. To disable debugging, use the **no** form of this command.

debug ephone register [**mac-address** *mac-address*]

no debug ephone register [**mac-address** *mac-address*]

Syntax Description	Parameter	Description
	mac-address	(Optional) Defines the MAC address of the Cisco IP phone.
	<i>mac-address</i>	(Optional) Specifies the MAC address of the Cisco IP phone.

Command Modes Privileged EXEC

Command History	Release	Modification
	12.1(5)YD	This command was introduced on the following platforms: Cisco 2600 series and Cisco 3600 series multiservice routers, and Cisco IAD2420 series IADs.
	12.2(2)XT	This command was implemented on Cisco 1750 and Cisco 1751 multiservice routers.
	12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers.
	12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
	12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.

Usage Guidelines The **debug ephone register** command sets registration debugging for the Cisco IP phones. If the **mac-address** keyword is not used, the **debug ephone register** debug command debugs all Cisco IP phones that are registered to the router. You can remove debugging for the Cisco IP phones that you do not want to debug by using the **mac-address** keyword with the **no** form of this command.

Debugging can be enabled or disabled on any number of Cisco IP phones. The Cisco IP phones that have debugging enabled can be seen by entering the **show ephone** command and looking at the debug field in the output. When debugging is enabled for a Cisco IP phone, the debug output is displayed for any Cisco IP phone directory numbers or virtual voice ports associated with the Cisco IP phone.

Syntax Description The following is sample output of registration debugging for the Cisco IP phone with MAC address 0030.94c3.8724:

```
Router# debug ephone register mac-address 0030.94c3.8724

Ephone registration debugging is enabled
```

```

1d06h: New Skinny socket accepted [1] (2 active)
1d06h: sin_family 2, sin_port 50778, in_addr 10.1.0.21
1d06h: skinny_add_socket 1 10.1.0.21 50778
1d06h: ephone-(1)[1] StationRegisterMessage (2/3/12) from 10.1.0.21
1d06h: ephone-(1)[1] Register StationIdentifier DeviceName SEP003094C3E1A8
1d06h: ephone-(1)[1] StationIdentifier Instance 1 deviceType 7
1d06h: ephone-1[-1]:stationIpAddr 10.1.0.21
1d06h: ephone-1[-1]:maxStreams 0
1d06h: ephone-(1) Allow any Skinny Server IP address 10.1.0.6
.
.
1d06h: ephone-1[1]:RegisterAck sent to ephone 1: keepalive period 30
.
.

```

Related Commands

Command	Description
debug ephone detail	Sets detail debugging for the Cisco IP phone.
debug ephone error	Sets error debugging for the Cisco IP phone.
debug ephone keepalive	Sets keepalive debugging for the Cisco IP phone.
debug ephone pak	Provides voice packet level debugging and prints the contents of one voice packet in every 1024 voice packets.
debug ephone raw	Provides raw low-level protocol debugging display for all Skinny Client Control Protocol messages
debug ephone state	Sets state debugging for the Cisco IP phone.
debug ephone statistics	Sets statistics debugging for the Cisco IP phone.
show debugging	Displays information about the types of debugging that are enabled for your router.

debug ephone state

To set state debugging for the Cisco IP phone, use the **debug ephone state** command in privileged EXEC mode. To disable debugging, use the **no** form of this command.

debug ephone state [**mac-address** *mac-address*]

no debug ephone state [**mac-address** *mac-address*]

Syntax Description	Parameter	Description
	mac-address	(Optional) Defines the MAC address of the Cisco IP phone.
	<i>mac-address</i>	(Optional) Specifies the MAC address of the Cisco IP phone.

Command Modes Privileged EXEC

Command History	Release	Modification
	12.1(5)YD	This command was introduced on the following platforms: Cisco 2600 series and Cisco 3600 series multiservice routers, and Cisco IAD2420 series IADs.
	12.2(2)XT	This command was implemented on Cisco 1750 and Cisco 1751 multiservice routers.
	12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers.
	12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
	12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.

Usage Guidelines The **debug ephone state** command sets state debugging for the Cisco IP phones.

If the **mac-address** keyword is not used, the **debug ephone state** debug command debugs all Cisco IP phones that are registered to the router. You can remove debugging for the Cisco IP phones that you do not want to debug by using the **mac-address** keyword with the **no** form of this command.

Debugging can be enabled or disabled on any number of Cisco IP phones. The Cisco IP phones that have debugging enabled can be seen by entering the **show ephone** command and looking at the debug field in the output. When debugging is enabled for a Cisco IP phone, the debug output is displayed for any Cisco IP phone directory numbers or virtual voice ports associated with the Cisco IP phone.

Examples The following is sample output of state debugging for the Cisco IP phone with MAC address 0030.94c3.E1A8:

```
Router# debug ephone state mac-address 0030.94c3.E1A8

EPHONE state debugging is enabled for phone 0030.94C3.E1A8
```

debug ephone state

```

1d06h: ephone-1[1]:OFFHOOK
1d06h: ephone-1[1]:SIEZE on activeline 0
1d06h: ephone-1[1]:SetCallState line 1 DN 1 TsOffHook
1d06h: ephone-1[1]:Skinny-to-Skinny call DN 1 to DN 2 instance 1
1d06h: ephone-1[1]:SetCallState line 1 DN 1 TsRingOut
1d06h: ephone-1[1]:Call Info DN 1 line 1 ref 158 called 5002 calling 5001
1d06h: ephone-1[1]: Jane calling
1d06h: ephone-1[1]: Jill
1d06h: ephone-1[1]:SetCallState line 3 DN 2 TsRingIn
1d06h: ephone-1[1]:Call Info DN 2 line 3 ref 159 called 5002 calling 5001
1d06h: ephone-1[1]: Jane calling
1d06h: ephone-1[1]: Jill
1d06h: ephone-1[1]:SetCallState line 3 DN 2 TsCallRemoteMultiline
1d06h: ephone-1[1]:SetCallState line 1 DN 1 TsConnected
1d06h: ephone-1[1]:OpenReceive DN 1 codec 4:G711Ulaw64k duration 10 ms bytes 80
1d06h: ephone-1[1]:OpenReceiveChannelAck 1.2.172.21 port=24010
1d06h: ephone-1[1]:StartMedia 1.2.172.22 port=24612
1d06h: DN 1 codec 4:G711Ulaw64k duration 10 ms bytes 80
1d06h: ephone-1[1]:CloseReceive
1d06h: ephone-1[1]:StopMedia
1d06h: ephone-1[1]:SetCallState line 3 DN 2 TsOnHook
1d06h: ephone-1[1]:SetCallState line 1 DN 1 TsOnHook
1d06h: ephone-1[1]:SpeakerPhoneOnHook
1d06h: ephone-1[1]:ONHOOK
1d06h: ephone-1[1]:SpeakerPhoneOnHook
1d06h: SkinnyReportDnState DN 1 ONHOOK

```

Related Commands

Command	Description
debug ephone detail	Sets detail debugging for the Cisco IP phone.
debug ephone error	Sets ephone debugging for the Cisco IP phone.
debug ephone keepalive	Sets keepalive debugging for the Cisco IP phone.
debug ephone pak	Provides voice packet level debugging and prints the contents of one voice packet in every 1024 voice packets.
debug ephone raw	Provides raw low-level protocol debugging display for all Skinny Client Control Protocol messages.
debug ephone register	Sets registration debugging for the Cisco IP phone.
debug ephone statistics	Sets statistics debugging for the Cisco IP phone.
show debugging	Displays information about the types of debugging that are enabled for your router.

debug ephone statistics

To set call statistics debugging for the Cisco IP phone, use the **debug ephone statistics** command in privileged EXEC mode. To disable debugging, use the **no** form of this command.

debug ephone statistics [**mac-address** *mac-address*]

no debug ephone statistics [**mac-address** *mac-address*]

Syntax Description	Parameter	Description
	mac-address	(Optional) Defines the MAC address of the Cisco IP phone.
	<i>mac-address</i>	(Optional) Specifies the MAC address of the Cisco IP phone.

Command Modes Privileged EXEC

Command History	Release	Modification
	12.1(5)YD	This command was introduced on the following platforms: Cisco 2600 series and Cisco 3600 series multiservice routers, and Cisco IAD2420 series IADs.
	12.2(2)XT	This command was implemented on Cisco 1750 and Cisco 1751 multiservice routers.
	12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers.
	12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
	12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.

Usage Guidelines The **debug ephone statistics** command provides a debug monitor display of the periodic messages from the Cisco IP phone to the router. These include transmit-and-receive packet counts and an estimate of drop packets. The call statistics can also be displayed for live calls using the **show ephone** command.

If the **mac-address** keyword is not used, the **debug ephone statistics** debug command debugs all Cisco IP phones that are registered to the router. You can remove debugging for the Cisco IP phones that you do not want to debug by using the **mac-address** keyword with the **no** form of this command.

Debugging can be enabled or disabled on any number of Cisco IP phones. The Cisco IP phones that have debugging enabled can be seen by entering the **show ephone** command and looking at the debug field in the output. When debugging is enabled for a Cisco IP phone, the debug output is displayed for any Cisco IP phone directory numbers or virtual voice ports associated with the Cisco IP phone.

Examples The following is sample output of statistics debugging for the Cisco IP phone with MAC address 0030.94C3.E1A8:

```
Router# debug ephone statistics mac-address 0030.94C3.E1A8
```

```

EPHONE statistics debugging is enabled for phone 0030.94C3.E1A8

1d06h: Clear Call Stats for DN 1 call ref 162
1d06h: Clear Call Stats for DN 1 call ref 162
1d06h: Clear Call Stats for DN 1 call ref 162
1d06h: Clear Call Stats for DN 2 call ref 163
1d06h: ephone-1[1]:GetCallStats line 1 ref 162 DN 1: 5001
1d06h: ephone-1[1]:Call Stats for line 1 DN 1 5001 ref 162
1d06h: ephone-1[1]:TX Pkts 0 bytes 0 RX Pkts 0 bytes 0
1d06h: ephone-1[1]:Pkts lost 4504384 jitter 0 latency 0
1d06h: ephone-1[1]:Src 0.0.0.0 0 Dst 0.0.0.0 0 bytes 80 vad 0 G711Ulaw64k
1d06h: ephone-1[1]:GetCallStats line 1 ref 162 DN 1: 5001
1d06h: STATS: DN 1 Packets Sent 0
1d06h: STATS: DN 2 Packets Sent 0
1d06h: ephone-1[1]:Call Stats found DN -1 from Call Ref 162
1d06h: ephone-1[1]:Call Stats for line 0 DN -1 5001 ref 162
1d06h: ephone-1[1]:TX Pkts 275 bytes 25300 RX Pkts 275 bytes 25300
1d06h: ephone-1[1]:Pkts lost 0 jitter 0 latency 0

```

Related Commands

Command	Description
debug ephone detail	Sets detail debugging for the Cisco IP phone.
debug ephone error	Sets error debugging for the Cisco IP phone.
debug ephone keepalive	Sets keepalive debugging for the Cisco IP phone.
debug ephone pak	Provides voice packet level debugging and prints the contents of one voice packet in every 1024 voice packets.
debug ephone raw	Provides raw low-level protocol debugging display for all Skinny Client Control Protocol messages.
debug ephone register	Sets registration debugging for the Cisco IP phone.
debug ephone state	Sets state debugging for the Cisco IP phone.
show debugging	Displays information about the types of debugging that are enabled for your router.

default-destination

To assign a default destination number for routing incoming telephone calls to IP phones on the SRS Telephony router that can not go to fallback mode, use the **default-destination** command in CallManager fallback configuration mode. To delete the default destination number on the SRS Telephony router, use the **no** form of this command.

default-destination *phone-number*

no default-destination *phone-number*

Syntax Description	<i>phone-number</i>	Telephone number of the default destination.
---------------------------	---------------------	--

Defaults	No default behavior or values.
-----------------	--------------------------------

Command Modes	CallManager fallback configuration
----------------------	------------------------------------

Command History	Release	Modification
	12.1(5)YD	This command was introduced on the following platforms: Cisco 2600 series and Cisco 3600 series multiservice routers, and Cisco IAD2420 series IADs.
	12.2(2)XT	This command was implemented on Cisco 1750 and Cisco 1751 multiservice routers.
	12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers.
	12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
	12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.

Usage Guidelines	The default-destination command assigns default destination number for calls that cannot be reached in the fallback mode. For example, if 100 phones are connected to a Cisco router configured with Cisco SRT Telephony and the WAN link fails, then only the maximum allowed number of the phones for each platform can fall back to the router. The remaining phones are not reachable during the fallback mode. All calls to these phones will be directed to the number configured by the default-destination command.
-------------------------	---

Examples	The following example sets the default destination to 40802:
-----------------	--

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# default-destination 40802
```

default-destination**Related Commands**

Command	Description
alias	Provides a mechanism for servicing calls to telephone numbers that are unavailable during CallManager fallback.
call-manager-fallback	Enables SRS Telephony feature support and enters CallManager fallback configuration mode.

dialplan-pattern

To create a global prefix that can be used to expand the extension numbers of inbound and outbound calls into fully qualified E.164 numbers, use the **dialplan-pattern** command in call-manager-fallback configuration mode. To disable the **dialplan-pattern** command settings, use the **no** form of this command.

```
dialplan-pattern tag pattern extension-length extension-length [extension-pattern
extension-pattern] [no-reg]
```

```
no dialplan-pattern tag [pattern extension-length extension-length extension-pattern
extension-pattern] [no-reg]
```

Syntax Description	
<i>tag</i>	Dial-plan string tag used before a ten-digit telephone number. The tag number is from 1 to 5.
<i>pattern</i>	Dial-plan pattern, such as the area code, the prefix, and the first one or two digits of the extension number, plus wild card markers or dots (.) for the remainder of the extension number digits.
extension-length	Sets the number of extension digits that will appear as a caller ID.
<i>extension-length</i>	The number of extension digits. The extension length must match the setting for IP phones in Cisco CallManager mode. 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 <i>pattern</i> variable.
<i>extension-pattern</i>	(Optional) The extension number's leading digit pattern. Consists of one or more digits and wild card markers or dots (.). For example, 5.. would include extension 500 to 599, 5... would include 5000 to 5999. The extension pattern must match the setting for IP phones in Cisco CallManager mode.
no-reg	(Optional) Prevents the E.164 numbers in the dial peer from registering with the gatekeeper.

Defaults No default behavior or values

Command Modes Call-manager-fallback configuration

Command History	Cisco IOS Release	Modification
	12.1(5)YD	This command was introduced on the Cisco 2600 series and Cisco 3600 series multiservice routers, and on the Cisco IAD2420 series.
	12.2(2)XT	This command was implemented on the Cisco 1750 and Cisco 1751 multiservice routers.
	12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745 routers.

Cisco IOS Release	Modification
12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.
12.2(11)YT	The extension-pattern keyword was added.

Usage Guidelines

The **dialplan-pattern** command builds additional dial peers. For example, if a hidden Plain Old Telephone System (POTS) dial peer is created, such as the following:

```
Router(config)# dial-peer voice 20001 pots
Router(config-dial-peer)# destination-pattern 1001
Router(config-dial-peer)# voice-port 50/0/2
```

and a dial-plan pattern is created, such as 40855510.., then an additional dial peer will be created that allows calls to both the 1001 and 4085551001 numbers. For example:

```
Router(config)# dial-peer voice 20002 pots
Router(config-dial-peer)# destination-pattern 4085551001
Router(config-dial-peer)# voice-port 50/0/2
```

Both dial peers can be seen with the **show dial-peer** command.

The **dialplan-pattern** command also creates a global prefix that can be used by inbound calls (calls to an IP phone in a Cisco SRST system) and outbound calls (calls made from an IP phone in a Cisco SRST system) to expand their extension numbers to fully qualified E.164 numbers.

For inbound calls (calls to an IP phone in a Cisco SRST system) where the calling party number matches the dial-plan pattern, the call is considered a local call and has a distinctive ring that identifies the call as internal. Any calling party number that does not match the dial-plan pattern is considered an external call and has a distinctive ring that is different from the internal ringing.

For outbound calls, the **dialplan-pattern** command converts the calling party's extension number to an E.164 calling party number. Outbound calls that do not use an E.164 number and go through a Primary Rate Interface (PRI) connection to the Public Switched Telephony Network (PSTN), may be rejected by the PRI link as the calling party identifier.

A dial-plan pattern is required to register the Cisco IP phone lines with a gatekeeper. Directory numbers for the Cisco IP phones must match the number the *extension-length* argument. For example, if the extension length is 3, then all extension must be three numbers in length. Otherwise, the extension number cannot be converted to a qualified E.164 number.

If there are multiple patterns, called-party numbers are checked in numeric order, starting with pattern 1, until a match is found or the until last pattern has been checked. The valid dial-plan pattern with the lowest tag is used as a prefix to all local Cisco IP phones.

When **extension-pattern** *extension-length* keyword and argument are used, the leading digits of an extension pattern are stripped and replaced with the corresponding leading digits of the dial-plan. For example, the following command maps all extension numbers 4xx to the PSTN number 40855501xx, so that extension 412 corresponds to 4085550112.

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# dialplan-pattern 1 4085550100 extension-length 3
extension-pattern 4..
```

The number of *extension-pattern* argument characters must match the *extension-length* argument. For example, if the *extension-length* is three, then the *extension-pattern* can be 8.., 1.., 51., and so forth.

The **no-reg** keyword provides the option of not registering specific numbers to the gatekeeper so that those numbers can be used for other telephony services.

Examples

The following example shows how to create dial-plan pattern 1 for extension numbers 5000 to 5099 with a prefix of 408555. If an inbound calling party number (4085555044) matches dial-plan pattern 1, then the recipient phone will display an extension (5044) as the caller ID and use an internal ringing tone. If an outbound calling party extension number (5044) matches dial-plan pattern 1, then the calling party extension will be converted to an E.164 number (4085555044). The E.164 calling party number that will appear as the caller ID.

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# dialplan-pattern 1 40855550.. extension-length 4
extension-pattern 50..
```

In the following example the **dialplan-pattern** command creates dial-plan pattern 1 for extensions 800 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 4085579001 (an external number).

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# dialplan-pattern 1 40855590.. extension-length 3
extension-pattern 8..
```

In the following example shows a configuration for two Cisco SRST systems. Each is configured with the same **dialplan pattern** commands, but one system uses 50.. and the other uses 60.. for extension numbers. Calls from the “50..” system to the “60..” system, and vice versa, are treated as internal calls. Calls that go across a H.323 network and calls that go to a PSTN through an Integrated Service Digital Network (ISDN) interface on one of the configured Cisco SRST routers are represented as E.164.

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# dialplan-pattern 1 40855550.. extension-length 4
extension-pattern 50..
Router(config-cm-fallback)# dialplan-pattern 2 51055560.. extension-length 4
extension-pattern 60..
```

Related Commands

Command	Description
call-manager-fallback	Enables Cisco SRST support and enters call-manager-fallback configuration mode.

huntstop

To set the huntstop attribute for the dial peers associated with the Cisco IP phone dial peers created during CallManager fallback, use the **huntstop** command in CallManager fallback configuration mode. To disable huntstop, use the **no** form of this command.

huntstop

no huntstop

Syntax Description This command has no arguments or keywords.

Defaults Huntstop is enabled by default.

Command Modes CallManager fallback configuration

Command History

Release	Modification
12.1(5)YD	This command was introduced on the following platforms: Cisco 2600 series and Cisco 3600 series multiservice routers, and Cisco IAD2420 series IADs.
12.2(2)XT	This command was implemented on Cisco 1750 and Cisco 1751 multiservice routers.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers.
12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.

Usage Guidelines

In CallManager fallback configuration mode, the huntstop attribute by default is set uniformly to all Cisco IP phone lines (for example, to all or to none).



Note

Use the **no huntstop** command only if you want to disable huntstop.

Examples

The following example shows how to disable huntstop for all the Cisco IP phones:

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# no huntstop
```

Related Commands	Command	Description
	call-manager-fallback	Enables SRS Telephony feature support and enters CallManager fallback configuration mode.
	huntstop (dial-peer)	Disables all further dial-peer hunting if a call fails using hunt groups.

ip source-address

To enable a router to receive messages from Cisco IP phones through the specified IP addresses and ports, use the **ip source-address** command in CallManager fallback configuration mode. To disable the router from receiving messages from Cisco IP phones, use the **no** form of this command.

ip source-address *ip-address* [**port** *port*] [**any-match** | **strict-match**]

no ip source-address [*ip-address* **port** *port*] [**any-match** | **strict-match**]

Syntax Description		
	<i>ip-address</i>	The IP address is the preexisting router IP address, typically one of the addresses of the Ethernet port of the router.
	port	(Optional) The port to which the gateway router connects to receive messages from the Cisco IP phones.
	<i>port</i>	(Optional) The port number. The default port number is 2000.
	any-match	(Optional) Disable strict IP address checking for registration.
	strict-match	(Optional) Require strict IP address checking for registration.

Defaults

Default port number: 2000

Default server address match: **any-match**

Command Modes

CallManager fallback configuration

Command History

Release	Modification
12.1(5)YD	This command was introduced on the following platforms: Cisco 2600 series and Cisco 3600 series multiservice routers, and Cisco IAD2420 series IADs.
12.2(2)XT	This command was implemented on Cisco 1750 and Cisco 1751 multiservice routers.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers.
12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.

Usage Guidelines

The **ip source-address** command is a mandatory command, and the fallback subsystem does not start if the IP address is not provided. If the port number is not provided, the default value (2000) is used. The IP address is usually the IP address of the Ethernet port to which the phones are connected.

Use the **any-match** keyword to instruct the router to permit Cisco IP phone registration even when the IP server address used by the phone does not match the IP source address. This option can be used to allow registration of Cisco IP phones on different subnets that with different default Dynamic Host Configuration Protocol (DHCP) routers or different TFTP server addresses.

Use the **strict-match** keyword to instruct the router to reject Cisco IP phone registration attempts if the IP server address used by the phone does not exactly match the source address. By dividing the Cisco IP phones into groups on different subnets and giving each group different DHCP default-router or TFTP server addresses, this option can be used to restrict the number of Cisco IP phones allowed to register.

The **ip source-address** command enables a router to receive messages from Cisco IP phones through the specified IP addresses and port. If the router receives a registration request from a Cisco IP phone, the router in return requests the phone configuration and dial-plan information from the Cisco IP phone. This data is stored locally in the memory of the router and is used to create voice port and dial-plan information. The voice port and dial-plan information is used to handle telephony calls to and from the Cisco IP phone if the Cisco CallManager is unreachable.

Examples

The following example shows how to set the IP source address and port:

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# ip source-address 10.6.21.4 port 2002 strict-match
```

Related Commands

Command	Description
call-manager-fallback	Enables SRS Telephony feature support and enters CallManager fallback configuration mode.

keepalive

To configure the time interval between successive keepalive messages from Cisco IP phones, use the **keepalive** command in CallManager fallback configuration mode. To restore to the default interval, use the **no** form of this command.

keepalive *seconds*

no keepalive *seconds*

Syntax Description	<i>seconds</i>	Keepalive message transmission interval, in seconds. The valid range is from 10 to 65535 seconds. The default timeout value is 30.
---------------------------	----------------	--

Defaults	30 seconds
-----------------	------------

Command Modes	CallManager fallback configuration
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Command History	Release	Modification
	12.1(5)YD	This command was introduced on the following platforms: Cisco 2600 series and Cisco 3600 series multiservice routers; and Cisco IAD2420 series IADs.
	12.2(2)XT	This command was implemented on Cisco 1750 and Cisco 1751 multiservice routers.
	12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers.
	12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
	12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.

Usage Guidelines	The keepalive command configures the time interval between successive keepalive messages from Cisco IP phones to the SRS Telephony router. If the router fails to receive three successive keepalive messages, it considers the Cisco IP phone to be out of service until the phone reregisters.
-------------------------	---



Note

The **keepalive** command is applicable only after a Cisco IP phone has registered with the SRS Telephony-enabled router.

Examples	The following example sets the keepalive timeout value to 60 seconds:
-----------------	---

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# keepalive 60
```

Related Commands

Command	Description
call-manager-fallback	Enables SRS Telephony feature support and enters CallManager fallback configuration mode.

limit-dn

To specify the maximum number of lines available on each Cisco IP phone, use the **limit-dn** command in CallManager fallback configuration mode. To return to the default setting, use the **no** form of this command.

limit-dn {7910 | 7940 | 7960} *max-lines*

no limit-dn [7910 | 7940 | 7960]

Syntax Description		
	7910	Cisco IP Phone 7910.
	7940	Cisco IP Phone 7940.
	7960	Cisco IP Phone 7960.
	<i>max-lines</i>	Maximum lines setting is from 1 to 6. The default is 6.

Defaults Maximum number of lines: 6

Command Modes CallManager fallback configuration

Command History	Release	Modification
	12.2(4)XT	This command was introduced on the following platforms: Cisco 1750, Cisco 1751, Cisco 2600 series, and Cisco 3600 series multiservice routers; and Cisco IAD2420 series IADs.
	12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers.
	12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
	12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.

Usage Guidelines The **limit-dn** command specifies the maximum number of lines (directory numbers) available for each Cisco IP phone type.



Note

You must specify this value during initial SRS Telephony router configuration, before any Cisco IP phone actually registers with the SRS Telephony router. You can modify the number of lines at a later time.

The range of the maximum number of lines is from 1 to 6. The default number of lines is 6. If there is any active phone with a number greater than the specified limit, warning information is displayed for phone reset.

Examples

The following example shows how to set a directory number limit of 2 for the Cisco IP Phone 7910:

```
Router(config)# call-manager-fallback  
Router(config-cm-fallback)# limit-dn 7910 2
```

Related Commands

Command	Description
call-manager-fallback	Enables SRS Telephony feature support and enters CallManager fallback configuration mode.

max-dn

To set the maximum number of directory numbers or virtual voice ports that can be supported by the router, use the **max-dn** command in CallManager fallback configuration mode. To return to the default number of directories or virtual voice ports, use the **no** form of this command.

max-dn *max-directory-numbers*

no max-dn

Syntax Description	<i>max-directory-numbers</i> Maximum number of directory numbers or virtual voice ports supported by the router. The maximum number is platform-dependent. See the “Specifications” section on page 1-7. The default is 0.
---------------------------	--

Defaults	0 directory numbers
-----------------	---------------------

Command Modes	CallManager fallback configuration
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Command History	Release	Modification
	12.1(5)YD	This command was introduced on the following platforms: Cisco 2600 series and Cisco 3600 series multiservice routers; and Cisco IAD2420 series IADs.
	12.2(2)XT	This command was implemented on Cisco 1750 and Cisco 1751 multiservice routers.
	12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers.
	12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
	12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.

Usage Guidelines	The max-dn command limits the number of Cisco IP phone directory numbers or virtual voice ports available on the router.
-------------------------	---



Note

Once you have specified the maximum number of directory numbers available, you cannot reduce that number of directory numbers or virtual voice ports without rebooting the router.

Examples	The following example sets the maximum number of directory numbers or virtual voice ports to 12:
-----------------	--

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# max-dn 12
```

Related Commands

Command	Description
call-manager-fallback	Enables SRS Telephony feature support and enters CallManager fallback configuration mode.

max-ephones

To configure the maximum number of Cisco IP phones that can be supported by a router, use the **max-ephones** command in CallManager fallback configuration mode. To return to the default number of Cisco IP phones, use the **no** form of this command.

max-ephones *max-phones*

no max-ephones

Syntax Description	<i>max-phones</i>	Maximum number of Cisco IP phones supported by the router. The maximum number is platform-dependent. See the “Specifications” section on page 1-7. The default is 0.
---------------------------	-------------------	--

Defaults 0 Cisco IP phones.

Command Modes CallManager fallback configuration

Command History	Release	Modification
	12.1(5)YD	This command was introduced on the following platforms: Cisco 2600 series and Cisco 3600 series multiservice routers, and Cisco IAD2420 series IADs.
	12.2(2)XT	This command was implemented on Cisco 1750 and Cisco 1751 multiservice routers.
	12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers.
	12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
	12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.

Usage Guidelines The **max-ephones** command limits the number of Cisco IP phones supported on the router.



Note

Once you have specified the maximum number of Cisco IP phones, you cannot reduce that number without rebooting the router.

Examples The following example sets the maximum number of Cisco IP phones for a Cisco router to 24:

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# max-ephones 24
```

Related Commands

Command	Description
call-manager-fallback	Enables SRS Telephony feature support and enters CallManager fallback configuration mode.

moh

To configure music on hold (MOH), use the **moh** command in CallManager fallback configuration mode. To disable music on hold, use the **no** form of this command.

moh *filename*

no moh *filename*

Syntax Description	<i>filename</i>	Filename of the music file. The music file must be in the system Flash.
---------------------------	-----------------	---

Defaults	MOH is enabled by default.
-----------------	----------------------------

Command Modes	CallManager fallback configuration
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Command History	Release	Modification
	12.2(2)XT	This command was introduced on the following platforms: Cisco 1750, Cisco 1751, Cisco 2600 series and Cisco 3600 series multiservice routers; Cisco IAD2420 series IADs.
	12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers.
	12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
	12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.

Usage Guidelines	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. MOH works only for G.711 calls and on-net VoIP and PSTN calls. For all other calls, callers hear a periodic tone. For example, internal calls between Cisco IP phones do not get MOH, instead callers hear a tone.
-------------------------	---



Note

Music-on-hold files can be .wav or .au file format; however, the file format must contain 8-bit 8 kHz data; for example, CCITT a-law or u-law data format.
--

Examples	The following example enables MOH and specifies the music files:
-----------------	--

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# moh minuet.wav
Router(config-cm-fallback)# moh minuet.au
```

Related Commands

Command	Description
call-manager-fallback	Enables SRS Telephony feature support and enters CallManager fallback configuration mode.

pattern direct

To configure the dual-tone multifrequency (DTMF) digit pattern forwarding necessary to activate the voice-mail system when the user presses the messages button on the phone, use the **pattern direct** command in voicemail integration configuration mode. To disable DTMF digit pattern forwarding when the user presses the messages button on the phone, use the **no** form of this command.

```
pattern direct tag1 {CGN | CDN | FDN} [tag2 {CGN | CDN | FDN}]
[tag3 {CGN | CDN | FDN}] [tag4]
```

```
no pattern direct tag1 {CGN | CDN | FDN} [tag2 {CGN | CDN | FDN}]
[tag3 {CGN | CDN | FDN}] [tag4]
```

Syntax Description		
<i>tag</i>		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. The Cisco IOS Telephony Service router supports a maximum of four tags.
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.



Note

Although it is unlikely that you will use multiple instances of the **CGN**, **CDN**, or **FDN** keyword in a single command line, it is permissible to do so.

Defaults This feature is disabled by default.

Command Modes Voice-mail integration configuration

Command History	Release	Modification
	12.2(2)XT	This command was introduced on the following platforms: Cisco 1750, Cisco 1751, Cisco 2600 series and Cisco 3600 series multiservice routers; and Cisco IAD2420 series IADs.
	12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745 routers.
	12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
	12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.
	12.2(13)T	This command was introduced to the SRS Telephony Version 2.02

Usage Guidelines

The **pattern direct** command is used to configure the sequence of DTMF digits passed to a voice-mail system attached to the Cisco IOS Telephony Service router through one or more voice ports. When a call is placed directly from a Cisco IP phone attached to the Cisco IOS Telephony Service router, the voice-mail system expects to receive a sequence of DTMF digits at the beginning of the call that identify the mailbox of the user calling the voice-mail system accompanied by a string of digits indicating that the caller is attempting to access the designated mailbox in order to retrieve messages.

Examples

The following example sets the DTMF pattern for a calling number (\$CGN) for a direct call to the voice-mail system:

```
Router(config) vm-integration
Router(config-vm-integration) pattern direct 2 CGN
```

Related Commands

Command	Description
pattern ext-to-ext busy	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 voicemail.
pattern ext-to-ext no-answer	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 voicemail.
pattern trunk-to-ext busy	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 voicemail.
pattern trunk-to-ext no-answer	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 voicemail.
vm-integration	Enters voice-mail integration mode and enables voice-mail integration with DTMF and analog voice-mail system.

pattern ext-to-ext busy

To configure the dual-tone multifrequency (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 voicemail, use the **pattern ext-to-ext busy** command in voicemail integration configuration mode. To disable DTMF digit pattern forwarding when an internal extension calls a busy extension and the call is forwarded to a voice-mail system, use the **no** form of this command.

```
pattern ext-to-ext busy tag1 {CGN | CDN | FDN} [tag2 {CGN | CDN | FDN}]
[tag3 {CGN | CDN | FDN}] [tag4]
```

```
no pattern ext-to-ext busy tag1 {CGN | CDN | FDN} [tag2 {CGN | CDN | FDN}]
[tag3 {CGN | CDN | FDN}] [tag4]
```

Syntax Description	
<i>tag</i>	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. The Cisco IOS Telephony Service router supports a maximum of four tags.
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.



Note

Although it is unlikely that you will use multiple instances of the **CGN**, **CDN**, or **FDN** keyword in a single command line, it is permissible to do so.

Defaults This feature is disabled by default.

Command Modes Voice-mail integration configuration

Command History	Release	Modification
	12.2(2)XT	This command was introduced on the following platforms: Cisco 1750, Cisco 1751, Cisco 2600 series and Cisco 3600 series multiservice routers; and Cisco IAD2420 series IADs.
	12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745 routers.
	12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.

Release	Modification
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.
12.2(13)T	This command was introduced to Cisco SRS Telephony Version 2.02

Usage Guidelines

The **pattern ext-to-ext busy** command is used to configure the sequence of DTMF digits passed to a voice-mail system attached to the Cisco IOS Telephony Service router through one or more voice ports. When a call is routed to the voice-mail system by call forward on busy from a Cisco IP phone attached to the Cisco IOS Telephony Service router, the voice-mail system expects to receive a sequence of digits identifying the mailbox associated with the forwarding phone together with digits that identify the extension number of the calling IP phone.

Examples

The following example sets the DTMF pattern for a local call forwarded on busy to the voice-mail system:

```
Router(config) vm-integration
Router(config-vm-integration) pattern ext-to-ext busy 7 FDN * CGN *
```

Related Commands

Command	Description
pattern direct	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when the user presses the messages button on the phone.
pattern ext-to-ext no-answer	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 voicemail.
pattern trunk-to-ext busy	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 voicemail.
pattern trunk-to-ext no-answer	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 voicemail.
vm-integration	Enters voice-mail integration mode and enables voice-mail integration with DTMF and analog voice-mail system.

pattern ext-to-ext no-answer

To configure the dual-tone multifrequency (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 voicemail, use the **pattern ext-to-ext no-answer** command in voicemail integration configuration mode. To disable DTMF digit pattern forwarding, use the **no** form of this command.

```
pattern ext-to-ext no-answer tag1 {CGN | CDN | FDN} [tag2 {CGN | CDN | FDN}]
[tag3 {CGN | CDN | FDN}] [tag4]
```

```
no pattern ext-to-ext no-answer tag1 {CGN | CDN | FDN} [tag2 {CGN | CDN | FDN}]
[tag3 {CGN | CDN | FDN}] [tag4]
```

Syntax Description	
<i>tag</i>	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. The Cisco IOS Telephony Service router supports a maximum of four tags.
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.



Note

Although it is unlikely that you will use multiple instances of the **CGN**, **CDN**, or **FDN** keyword in a single command line, it is permissible to do so.

Defaults This feature is disabled by default.

Command Modes Voice-mail integration configuration

Command History	Release	Modification
	12.2(2)XT	This command was introduced on the following platforms: Cisco 1750, Cisco 1751, Cisco 2600 series and Cisco 3600 series multiservice routers; and Cisco IAD2420 series IADs.
	12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745 routers.
	12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
	12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.
	12.2(13)T	This command was introduced to Cisco SRS Telephony Version 2.02

Usage Guidelines

The **pattern ext-to-ext no-answer** command is used to configure the sequence of DTMF digits passed to a voice-mail system attached to the Cisco IOS Telephony Service router through one or more voice ports. When a call is routed to the voice-mail system by call forward on no-answer from an IP phone attached to the Cisco IOS Telephony Service router, the voice-mail system expects to receive a sequence of digits identifying the mailbox associated with the forwarding phone together with digits that identify the extension number of the calling IP phone.

Examples

The following example sets the DTMF pattern for a local call forwarded on **no-answer** to the voice-mail system:

```
Router(config) vm-integration
Router(config-vm-integration) pattern ext-to-ext no-answer 5 FDN * CGN *
```

Related Commands

Command	Description
pattern direct	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when the user presses the messages button on the phone.
pattern ext-to-ext busy	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 voicemail.
pattern trunk-to-ext busy	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 voicemail.
pattern trunk-to-ext no-answer	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 voicemail.
vm-integration	Enters voice-mail integration mode and enables voice-mail integration with DTMF and analog voice-mail system.

pattern trunk-to-ext busy

To configure the dual-tone multifrequency (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 voicemail, use the **pattern trunk-to-ext busy** command in voicemail integration configuration mode. To disable DTMF digit pattern forwarding when an external trunk call reaches a busy extension and the call is forwarded to a voice-mail system, use the **no** form of this command.

```
pattern trunk-to-ext busy tag1 {CGN | CDN | FDN} [tag2 {CGN | CDN | FDN}]
[tag3 {CGN | CDN | FDN}] [tag4]
```

```
no pattern trunk-to-ext busy tag1 {CGN | CDN | FDN} [tag2 {CGN | CDN | FDN}]
[tag3 {CGN | CDN | FDN}] [tag4]
```

Syntax Description

<i>tag</i>	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. The Cisco IOS Telephony Service router supports a maximum of four tags.
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.



Note

Although it is unlikely that you will use multiple instances of the **CGN**, **CDN**, or **FDN** keyword in a single command line, it is permissible to do so.

Defaults

This feature is disabled by default.

Command Modes

Voice-mail integration configuration

Command History

Release	Modification
12.2(2)XT	This command was introduced on the following platforms: Cisco 1750, Cisco 1751, Cisco 2600 series and Cisco 3600 series multiservice routers; and Cisco IAD2420 series IADs.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745 routers.
12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.

Release	Modification
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.
12.2(13)T	This command was introduced to Cisco SRS Telephony Version 2.02

Usage Guidelines

The **pattern trunk-to-ext busy** command is used to configure the sequence of DTMF digits passed to a voice-mail system attached to the Cisco IOS Telephony Service router through one or more voice ports. When a call is routed to the voice-mail system by call forward on busy from an IP phone attached to the Cisco IOS Telephony Service router, the voice-mail system expects to receive a sequence of digits identifying the mailbox associated with the forwarding phone together with digits indicating that the call originated from a PSTN or VoIP caller.

Examples

The following example sets the DTMF pattern for call forwarding when an external trunk call reaches a busy extension and the call is forwarded to the voice-mail system:

```
Router(config) vm-integration
Router(config-vm-integration) pattern trunk-to-ext busy 6 FDN * CGN *
```

Related Commands

Command	Description
pattern direct	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when the user presses the messages button on the phone.
pattern ext-to-ext busy	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 voicemail.
pattern ext-to-ext no-answer	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 voicemail.
pattern trunk-to-ext no-answer	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 voicemail.
vm-integration	Enters voice-mail integration mode and enables voice-mail integration with DTMF and analog voice-mail system.

pattern trunk-to-ext no-answer

To configure the dual-tone multifrequency (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 voicemail, use the **pattern trunk-to-ext no-answer** command in voicemail integration configuration mode. To disable DTMF digit pattern forwarding when an external trunk call reaches another extension where the called party does not answer and the call is forwarded to a voice-mail system, use the **no** form of this command.

```
pattern trunk-to-ext no-answer tag1 {CGN | CDN | FDN} [tag2 {CGN | CDN | FDN}]
[tag3 {CGN | CDN | FDN}] [tag4]
```

```
no pattern trunk-to-ext no-answer tag1 {CGN | CDN | FDN} [tag2 {CGN | CDN | FDN}]
[tag3 {CGN | CDN | FDN}] [tag4]
```

Syntax Description		
<i>tag</i>		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. The Cisco IOS Telephony Service router supports a maximum of four tags.
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.



Note

Although it is unlikely that you will use multiple instances of the **CGN**, **CDN**, or **FDN** keyword in a single command line, it is permissible to do so.

Defaults This feature is disabled by default.

Command Modes Voice-mail integration configuration

Command History	Release	Modification
	12.2(2)XT	This command was introduced on the following platforms: Cisco 1750, Cisco 1751, Cisco 2600 series and Cisco 3600 series multiservice routers; and Cisco IAD2420 series IADs.
	12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745 routers.
	12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.

Release	Modification
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.
12.2(13)T	This command was introduced to Cisco SRS Telephony Version 2.02

Usage Guidelines

The **pattern trunk-to-ext no-answer** command is used to configure the sequence of DTMF digits passed to a voice-mail system attached to the Cisco IOS Telephony Service router through one or more voice ports. When a call is routed to the voice-mail system by call forward on no-answer from an IP phone attached to the Cisco IOS Telephony Service router, the voice-mail system expects to receive a sequence of digits identifying the mailbox associated with the forwarding phone together with digits indicating that the call originated from a PSTN or VoIP caller.

Examples

The following example sets the DTMF pattern for call forwarding when an external trunk call reaches an unanswered extension and the call is forwarded (FDN) to a voice-mail system:

```
Router(config) vm-integration
Router(config-vm-integration) pattern trunk-to-ext no-answer 4 FDN * CGN *
```

Related Commands

Command	Description
pattern direct	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when the user presses the messages button on the phone.
pattern ext-to-ext busy	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 voicemail.
pattern ext-to-ext no-answer	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 voicemail.
pattern trunk-to-ext busy	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 voicemail.
vm-integration	Enters voice-mail integration mode and enables voice-mail integration with DTMF and analog voice-mail system.

reset

To reset Cisco IP phones, use the **reset** command in CallManager fallback configuration mode.

```
reset { all seconds | mac-address mac-address }
```

Syntax Description	all	All Cisco IP phones.
	<i>seconds</i>	Time interval, in seconds, that passes between each Cisco IP phone resetting. The range is from 0 to 15 seconds.
	mac-address <i>mac-address</i>	MAC address of a particular Cisco IP phone.

Defaults No default behavior or values.

Command Modes CallManager fallback configuration

Command History	Release	Modification
	12.1(5)YD	This command was introduced on the following platforms: Cisco 2600 series and Cisco 3600 series multiservice routers; Cisco IAD2420 series IADs.
	12.2(2)XT	This command was implemented on Cisco 1750 and Cisco 1751 multiservice routers.
	12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers.
	12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
	12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.

Usage Guidelines This command does not have a **no** form.

Examples The following example resets all Cisco IP phones in eight-second intervals:

```
Router(config)# call-manager-fallback  
Router(config-cm-fallback)# reset all 8
```

The following example resets the Cisco IP phone with MAC address CFBA.321B.96FA:

```
Router(config)# call-manager-fallback  
Router(config-cm-fallback)# reset mac-address CFBA.321B.96FA
```

Related Commands

Command	Description
call-manager-fallback	Enables SRS Telephony feature support and enters CallManager fallback configuration mode.

show call-manager-fallback all

To display the detailed configuration of all Cisco IP phones, voice ports, and dial peers in your network while the Cisco CallManager fallback feature is active, use the **show call-manager-fallback all** command in EXEC mode.

show call-manager-fallback all

Syntax Description This command has no arguments or keywords.

Command Modes EXEC

Command History	Release	Modification
	12.1(5)YD	This command was introduced on the following platforms: Cisco 2600 series and Cisco 3600 series multiservice routers; Cisco IAD2420 series IADs.
	12.2(2)XT	This command was implemented on Cisco 1750 and Cisco 1751 multiservice routers.
	12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers.
	12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
	12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.

Examples The following is sample output from the **show call-manager-fallback all** command:

```
Router# show call-manager-fallback all

CONFIG
=====
ip source-address 10.0.0.1 port 2000
max-ephones 10
max-dn 10
no huntstop
translate called 1
call-forward busy 5001
call-forward noan 5001 timeout 8
cor incoming allowall 1 5001-5010
cor outgoing allow1800 2 5010-5020
alias 2 3... to 5555
keepalive 30

ephone-dn 1
number 4444
no huntstop
translate called 1
```



```
ephone-dn 2
number 3333
no huntstop
translate called 1

ephone-dn 3
number 5555 secondary 3...
no huntstop
translate called 1

ephone-dn 4
no huntstop
translate called 1
.
.
.
voice-port 50/0/1
station-id number 4444
timeout ringing 8
translate called 1
!
voice-port 50/0/2
station-id number 3333
timeout ringing 8
translate called 1
!
voice-port 50/0/3
station-id number 5555
timeout ringing 8
translate called 1
!
voice-port 50/0/4
timeout ringing 8
translate called 1
.
.
.
dial-peer voice 20046 pots
destination-pattern 4444
call-forward busy 5001
call-forward noan 5001
port 50/0/1

dial-peer voice 20047 pots
destination-pattern 3333
call-forward busy 5001
call-forward noan 5001
port 50/0/2

dial-peer voice 20048 pots
destination-pattern 5555
call-forward busy 5001
call-forward noan 5001
port 50/0/3

dial-peer voice 20049 pots
preference 9
destination-pattern 3...
call-forward busy 5001
call-forward noan 5001
port 50/0/3
```

Table 2 provides an alphabetical listing of the command fields in the sample output.

Table 2 *show call-manager-fallback all Field Descriptions*

Field	Description
destination-pattern	Destination pattern (telephone number) configured for this dial peer.
dial-peer voice	Voice dial peer.
ephone-dn	Cisco IP phone directory number.
huntstop	Huntstop is set.
ip source address	IP address used by the Cisco IP phones to register with the router for service.
keepalive	Cisco IP phone keepalive period in seconds.
max-ephones	Maximum number of Cisco IP phones.
max-dn	Maximum directory numbers or virtual voice ports.
port	TCP port number used by the Cisco IP phones to communicate with the router.
station-id number	Number used for caller-ID purposes when calls are made using the line.
voicemail	A voice-mail (speed-dial) number is set.
voice-port	(Virtual) voice port designator.

Related Commands

Command	Description
show call-manager fallback dial-peer	Displays output for the dial peers in Cisco CallManager fallback mode when the SRS Telephony feature is enabled.
show call-manager fallback ephone-dn	Displays output for the Cisco IP phone directory numbers or virtual voice ports in Cisco CallManager fallback mode when the SRS Telephony feature is enabled.
show call-manager fallback voice-port	Displays output for the voice ports in Cisco CallManager fallback mode when the SRS Telephony feature is enabled.

show call-manager-fallback dial-peer

To display output for the dial peers while Cisco CallManager fallback mode is active, use the **show call-manager-fallback dial-peer** command in EXEC mode.

show call-manager-fallback dial-peer

Syntax Description This command has no arguments or keywords.

Command Modes EXEC

Command History	Release	Modification
	12.1(5)YD	This command was introduced on the following platforms: Cisco 2600 series and Cisco 3600 series multiservice routers; Cisco IAD2420 series IADs.
	12.2(2)XT	This command was implemented on Cisco 1750 and Cisco 1751 multiservice routers.
	12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers.
	12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
	12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.

Examples The following is sample output from the **show call-manager-fallback dial-peer** command:

```
Router# show call-manager-fallback dial-peer
```

```
dial-peer voice 20046 pots
destination-pattern 4444
call-forward busy 5001
call-forward noan 5001
port 50/0/1
```

```
dial-peer voice 20047 pots
destination-pattern 3333
call-forward busy 5001
call-forward noan 5001
port 50/0/2
```

```
dial-peer voice 20048 pots
destination-pattern 5555
call-forward busy 5001
call-forward noan 5001
port 50/0/3
```

```
dial-peer voice 20049 pots
preference 9
```

show call-manager-fallback dial-peer

```
destination-pattern 3...
call-forward busy 5001
call-forward noan 5001
port 50/0/3
```

Table 3 provides an alphabetical listing of the command fields in the sample output.

Table 3 *show call-manager-fallback dial-peer Field Descriptions*

Field	Description
destination pattern	Destination pattern (telephone number) configured for this dial peer.
dial-peer voice	Voice dial peer.
huntstop	Huntstop is set.
port	(Virtual) voice port designator.

Related Commands

Command	Description
show call-manager fallback all	Displays the detailed configuration of all the Cisco IP phones, voice ports, and dial peers in a network in Cisco CallManager fallback mode when the SRS Telephony feature is enabled.
show call-manager fallback ephone-dn	Displays output for the Cisco IP phone directory numbers or virtual voice ports in Cisco CallManager fallback mode when the SRS Telephony feature is enabled.
show call-manager fallback voice-port	Displays output for the voice ports in Cisco CallManager fallback mode when the SRS Telephony feature is enabled.

show call-manager-fallback ephone-dn

To display output for the Cisco IP phone directory numbers or virtual voice ports while Cisco CallManager fallback is active, use the **show call-manager-fallback ephone-dn** command in EXEC mode.

show call-manager-fallback ephone-dn

Syntax Description This command has no arguments or keywords.

Command Modes EXEC

Command History	Release	Modification
	12.1(5)YD	This command was introduced on the following platforms: Cisco 2600 series and Cisco 3600 series multiservice routers; Cisco IAD2420 series IADs.
	12.2(2)XT	This command was implemented on Cisco 1750 and Cisco 1751 multiservice routers.
	12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers.
	12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
	12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.

Examples The following is sample output from the **show call-manager-fallback ephone-dn** command:

```
Router# show call-manager-fallback ephone-dn
```

```
ephone-dn 1
number 4444
no huntstop
translate called 1
```

```
ephone-dn 2
number 3333
no huntstop
translate called 1
```

```
ephone-dn 3
number 5555 secondary 3...
no huntstop
translate called 1
```

```
ephone-dn 4
no huntstop
translate called 1
```

Table 4 provides an alphabetical listing of the command fields in the sample output.

Table 4 *show call-manager-fallback ephone-dn Field Descriptions*

Field	Description
ephone-dn	Cisco IP phone directory number.
huntstop	Huntstop is set.
number	Cisco IP phone number.

Related Commands

Command	Description
show call-manager fallback all	Displays the detailed configuration of all the Cisco IP phones, voice ports, and dial peers in your network in Cisco CallManager fallback mode when the SRS Telephony feature is enabled.
show call-manager fallback dial-peer	Displays output for the dial peers in Cisco CallManager fallback mode when the SRS Telephony feature is enabled.
show call-manager fallback voice-port	Displays output for the voice ports in Cisco CallManager fallback mode when the SRS Telephony feature is enabled.

show call-manager-fallback voice-port

To display output for the voice ports while Cisco CallManager is active, use the **show call-manager-fallback voice-port** command in EXEC mode.

show call-manager-fallback voice-port

Syntax Description This command has no arguments or keywords.

Command Modes EXEC

Command History	Release	Modification
	12.1(5)YD	This command was introduced on the following platforms: Cisco 2600 series and Cisco 3600 series multiservice routers, and Cisco IAD2420 series IADs.
	12.2(2)XT	This command was implemented on Cisco 1750 and Cisco 1751 multiservice routers.
	12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers.
	12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
	12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.

Examples The following is sample output from the **show call-manager-fallback voice-port** command:

```
Router# show call-manager-fallback voice-port

voice-port 50/0/1
  station-id number 4444
  timeout ringing 8
  translate called 1
!
voice-port 50/0/2
  station-id number 3333
  timeout ringing 8
  translate called 1
!
voice-port 50/0/3
  station-id number 5555
  timeout ringing 8
  translate called 1
!
voice-port 50/0/4
  timeout ringing 8
  translate called 1
!
```

Table 5 describes the significant fields shown in the display.

Table 5 *show call-manager-fallback voice-port Field Descriptions*

Field	Description
voice-port	(Virtual) voice port.
station-id number	The phone number used for caller-ID purposes for calls made from this voice port.

Related Commands

Command	Description
show call-manager fallback all	Displays the detailed configuration of all the Cisco IP phones, voice ports, and dial peers in your network in Cisco CallManager fallback mode when the SRS Telephony feature is enabled.
show call-manager fallback dial-peer	Displays fallback output for the dial peers in Cisco CallManager fallback mode when the SRS Telephony feature is enabled.
show call-manager fallback ephone-dn	Displays fallback output for the Cisco IP phone directory numbers or virtual voice ports in Cisco CallManager fallback mode when the SRS Telephony feature is enabled.

show ephone

To display Cisco IP phone output, use the **show ephone** command in EXEC mode.

```
show ephone [mac-address | summary | registered | unregistered | 7910 | 7940 | 7960 | ringing |
offhook | dn dn-tag | remote | telephone-number phone-number]
```

Syntax	Description
<i>mac-address</i>	(Optional) Specifies the MAC address of the Cisco IP phone.
summary	(Optional) Displays summary information for all Cisco IP phones registered with the SRST router.
registered	(Optional) Displays status information for any Cisco IP phones that are currently registered with the SRST router.
unregistered	(Optional) Displays status information for all unregistered Cisco IP phones.
7910	(Optional) Displays information about any Cisco IP Phone 7910s registered with the SRST router.
7940	(Optional) Displays information about any Cisco IP Phone 7940s registered with the SRST router.
7960	(Optional) Displays information about any Cisco IP Phone 7960s registered with the SRST router.
ringing	(Optional) Displays status information for all Cisco IP phones registered with the SRST router that are currently in ringing mode.
offhook	(Optional) Displays a list of Cisco IP phones that are currently engaged or off hook.
dn <i>dn-tag</i>	(Optional) Cisco IP phone associated with a directory number tag.
remote	(Optional) Displays status information about all nonlocal Cisco IP phones (that is, phones that have no ARP entry).
telephone-number <i>phone-number</i>	(Optional) Displays status information for a Cisco IP phone featuring the specified telephone number.

Command Modes
EXEC

Command History	Release	Modification
	12.1(5)YD	This command was introduced on the following platforms: Cisco 2600 series and Cisco 3600 series multiservice routers, and Cisco IAD2420 series IADs.
	12.2(2)XT	This command was implemented on the Cisco 1750 and Cisco 1751 multiservice routers.
	12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745 routers.
	12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
	12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.

Usage Guidelines

The **show ephone** command displays statistical information for registered and unregistered Cisco IP phones. If a MAC address is not specified, all phones that can be identified by the SRST router are displayed. This command includes several keywords to enable you to get different types of output specific to your needs.

Examples

The following is sample output from the **show ephone** command:

```
Router# show ephone

ephone-1 Mac:0003.E3E7.F627 TCP socket:[1] activeLine:1 REGISTERED
mediaActive:1 offhook:1 ringing:0 reset:0 reset_sent:0 debug:0
IP:10.0.0.51 50570 Telecaster 7940 keepalive 49
button 1: dn 1 number 3001 CONNECTED
Active Call on DN 1:3001 10.0.0.51 31808 to 1.2.159.100 22708
Tx Pkts 452 bytes 41584 Rx Pkts 452 bytes 41584 Lost 0
Jitter 0 Latency 0

ephone-2 Mac:0030.94C3.E1A8 TCP socket:[2] activeLine:1 REGISTERED
mediaActive:1 offhook:1 ringing:0 reset:0 reset_sent:0 debug:0
IP:1.2.159.100 50942 Telecaster 7960 keepalive 78
button 1: dn 2 number 3002 CONNECTED
Active Call on DN 2:3002 1.2.159.100 22708 to 10.0.0.51 31808
Tx Pkts 452 bytes 41584 Rx Pkts 452 bytes 41584 Lost 0
Jitter 0 Latency 0

ephone-3 Mac:0030.94C3.F946 TCP socket:[-1] activeLine:0 UNREGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 debug:0
IP:10.2.1.2 52163 Telecaster 7960 keepalive 59

ephone-4 Mac:0030.94C3.F43A TCP socket:[-1] activeLine:0 UNREGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 debug:0
IP:10.2.1.1 51768 Telecaster 7960 keepalive 59
```

The following is sample output from the **show ephone mac-address** command for the Cisco IP phone with the MAC address 0030.94c3.f43a:

```
Router# show ephone 0030.94c3.f43a

ephone-3 Mac:0030.94C3.F43A TCP socket:[3] activeLine:0 REGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 debug:0
IP:1.5.81.13 Telecaster 7960 keepalive 28
button 1: dn 3 number 5003 IDLE
button 2: dn 5 number 5005 IDLE
button 3: dn 6 number 5006 IDLE
speed dial 1:3005
speed dial 2:3006
```

The following is sample output from the **show ephone dn dn-tag** command:

```
Router# show ephone dn 5

Tag 5, Normal or Intercom dn
ephone 1, mac-address 0030.94C3.CAA2, line 2
ephone 2, mac-address 0030.94c2.9919, line 3
```

The following is sample output from the **show ephone 7910** command:

```
Router# show ephone 7910

ephone-5 Mac:0004.DD1E.56ED TCP socket:[1] activeLine:0 REGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
```

```
IP:10.30.0.83 52234 Telecaster 7910  keepalive 270 max_line 2 dual-line
button 1: dn 15 number 5000 IDLE
button 2: dn 16 number 5500 IDLE
```

The following is sample output from the **show ephone 7960** command:

```
Router# show ephone 7960
```

```
ephone-11 Mac:0007.0EA6.2AA9 TCP socket:[13] activeLine:0 REGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
IP:10.30.0.53 52548 Telecaster 7960  keepalive 272 max_line 6
button 1: dn 16 number 9001 IDLE
button 2: dn 17 number 9002 IDLE
button 3: dn 18 number 9003 IDLE
button 4: dn 26 number A5001 auto dial A5002 IDLE
button 5: dn 31 number 1111 IDLE
speed dial 1:8001
```

```
ephone-12 Mac:0002.B9EB.0CB1 TCP socket:[14] activeLine:0 REGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
IP:10.30.0.54 50387 Telecaster 7960  keepalive 272 max_line 6
button 1: dn 20 number 9005 IDLE
button 2: dn 21 number 9006 IDLE
button 3: dn 22 number 9007 IDLE
button 4: dn 23 number 9008 IDLE
button 5: dn 30 number 1110 IDLE
```

The following is sample output from the **show ephone registered** command:

```
Router# show ephone registered
```

```
ephone-1 Mac:0653.D6B4.C601 TCP socket:[4] activeLine:0 REGISTERED
ephone-2 Mac:0653.D6B4.C602 TCP socket:[6] activeLine:0 REGISTERED
ephone-3 Mac:0653.D6B4.C603 TCP socket:[2] activeLine:0 REGISTERED
ephone-4 Mac:0653.D6B4.C604 TCP socket:[7] activeLine:0 REGISTERED
ephone-5 Mac:0653.D6B4.C605 TCP socket:[9] activeLine:0 REGISTERED
ephone-6 Mac:0653.D6B4.C608 TCP socket:[1] activeLine:0 REGISTERED
ephone-7 Mac:0653.D6B4.C609 TCP socket:[8] activeLine:0 REGISTERED
ephone-8 Mac:0653.D6B4.C614 TCP socket:[5] activeLine:0 REGISTERED
ephone-9 Mac:0653.D6B4.C622 TCP socket:[11] activeLine:0 REGISTERED
ephone-10 Mac:0653.D6B4.C623 TCP socket:[3] activeLine:0 REGISTERED
ephone-11 Mac:0007.0EA6.2AA9 TCP socket:[13] activeLine:0 REGISTERED
ephone-12 Mac:0002.B9EB.0CB1 TCP socket:[14] activeLine:0 REGISTERED
ephone-13 Mac:0653.D6B4.C625 TCP socket:[10] activeLine:0 REGISTERED
ephone-14 Mac:0653.D6B4.C624 TCP socket:[12] activeLine:0 REGISTERED
```

The following is sample output from the **show ephone telephone-number *phone-number*** command:

```
Router# show ephone telephone-number 4085998001
```

```
DP tag: 1, primary
Tag 1, Normal or Intercom dn
ephone 1, mac-address 0030.94c3.CAA2, line 1
```

Table 6 provides an alphabetic listing of the command fields in the sample output.

Table 6 *show ephone Field Descriptions*

Field	Description
Active Call	An active call is in progress.
activeLine	Indicates the line (button) on the phone that is in use. Zero indicates that no line is in use.

Table 6 *show ephone Field Descriptions (continued)*

Field	Description
button 1 : dn 1	Shows the directory number (DN) tag number associated with the phone button.
bytes	Total number of voice data bytes sent or received by the Cisco IP phone.
debug	If set to 1, indicates that debug for the phone is enabled; otherwise, it is set to 0.
DP tag	Dial-peer tag for the destination-pattern matching the specified telephone number.
dual_line	Indicates that the dual telephone line feature available on single-line Cisco IP phones (the Cisco 7905 or Cisco 7910, for example) has been activated.
ephone-1	Cisco IP phone tag number.
IDLE	The state of the DN is idle.
IP	Assigned IP address of the Cisco IP phone.
Jitter	The amount of variation (in milliseconds) of the time interval between voice packets received by the Cisco IP phone.
keepalive	Number of keepalive messages received from the Cisco IP phone by the router.
Latency	The estimated playout delay for voice packets received by the Cisco IP phone.
line 1	Indicates which line on the IP phone is associated with the specified telephone number.
Lost	Number of voice packets lost, as calculated by the Cisco IP phone, based on examining voice packet timestamp and sequence numbers during playout.
Mac	MAC address.
max_line	Indicates the maximum number of lines that may be assigned to the Cisco IP phone in question.
mediaActive	If set to 1, indicates that an active conversation is going on; otherwise, it is set to 0.
number	The telephone number associated with the Cisco IP phone button and its DN tag.
offhook	If set to 1, indicates that the Cisco IP phone is off the hook.
paging	Indicates the paging group to which the Cisco IP phone has been assigned.
REGISTERED	Indicates that the Cisco IP phone is active and registered. Alternatives states are UNREGISTERED (indicating that the connection to the Cisco IP phone was closed in a normal manner) and DECEASED (indicating that the connection to the Cisco IP phone was closed because of a keepalive timeout).
reset	Pending reset.
reset_sent	Request for reset sent to the Cisco IP phone.

Table 6 *show ephone Field Descriptions (continued)*

Field	Description
ringing	If set to 1, indicates that the IP phone's ringer is turned on and the phone is ringing; otherwise, it is set to 0.
Rx Pkts	Number of received voice packets.
speed dial	Speed-dial is set to a specific directory number.
Tag 1	Specifies the tag number identified in the ephone-dn command output.
TCP socket	Indicates the TCP socket number used to communicate with the IP phone. This can be correlated with the output of various other debug and show commands.
Telecaster 7960	Indicates the type and model of the Cisco IP phone. This information is received from the phone during its registration with the router.
Tx Pkts	Number of transmitted voice packets.

The following is a sample output from the **show ephone summary** command:

```
Router# show ephone summary
```

```
ephone-1 Mac:0030.94C3.37CB TCP socket:[-1] activeLine:0 REGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 debug:0
IP:10.1.1.1 Telecaster 7910 keepalive 45 1:1
sp1:5002 sp2:5003
```

```
ephone-2 Mac:0030.94C3.F96A TCP socket:[-1] activeLine:0 REGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 debug:0
IP:10.1.1.2 Telecaster 7960 keepalive 45 1:2 2:3 3:4
sp1:5004 sp2:5001
```

```
ephone-3 Mac:0030.94C3.F946 TCP socket:[-1] activeLine:0 REGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 debug:0
IP:10.2.1.2 Telecaster 7960 keepalive 59
```

```
ephone-4 Mac:0030.94C3.F43A TCP socket:[-1] activeLine:0 REGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 debug:0
IP:10.2.1.1 Telecaster 7960 keepalive 59
```

Table 3-9 provides an alphabetic listing of the command fields shown in the sample output.

Table 7 *show ephone summary Field Descriptions*

Field	Description
activeLine	Indicates the line (button) on the phone that is in use. Zero indicates that no line is in use
debug	If set to 1, indicates that debug for the phone is enabled; otherwise, it is set to 0.
ephone-1	Cisco IP phone tag number.
IP	Assigned IP address of the Cisco IP phone.

Table 7 show ephone summary Field Descriptions (continued)

Field	Description
keepalive	Number of keepalive messages received from the IP phone by the router.
Mac	MAC address.
mediaActive	If set to 1, indicates that an active conversation is going on; otherwise, it is set to 0.
offhook	If set to 1, indicates that the phone is off the hook.
REGISTERED	Indicates that the phone is active and registered. Alternatives states are UNREGISTERED (indicating that the connection to the Cisco IP phone was closed in a normal manner) and DECEASED (indicating that the connection to the Cisco IP phone was closed because of a keepalive timeout).
reset	Pending reset.
reset_sent	Request for reset sent to the Cisco IP phone.
ringing	If set to 1, indicates that the Cisco IP phone's ringer is turned on and the phone is ringing; otherwise, it is set to 0.
sp1	Speed-dial 1 set to a directory number.
sp2	Speed-dial 2 set to a directory number.
TCP socket	Indicates the TCP socket number used to communicate with the Cisco IP phone. This can be correlated with the output of various other debug and show commands.
Telecaster 7960	Indicates the type and model of the IP phone. This information is received from the phone during its registration with the router.

Related Commands

Command	Description
show ephone-dn	Displays the Cisco IP phone destination number.
show ephone-dn summary	Displays summary of the Cisco IP phone destination numbers.
show telephony-service	Displays the detailed configuration of all Cisco IP phones.

show ephone-dn

To display a Cisco IP phone destination number, use the **show ephone-dn** command in EXEC mode.

show ephone-dn [*tag* | **summary**]

Syntax Description	
<i>tag</i>	(Optional) Destination number tag. The destination number can be from 1 to 24.
loopback	(Optional) Cisco IP phone destination numbers in loopback mode.
summary	(Optional) Summary of all Cisco IP phone destination number.

Command Modes	
	EXEC

Command History	Release	Modification
	12.1(5)YD	This command was introduced on the following platforms: Cisco 2600 series and Cisco 3600 series multiservice routers, and Cisco IAD2420 series IADs.
	12.2(2)XT	This command was implemented on Cisco 1750 and Cisco 1751 multiservice routers.
	12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers.
	12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
	12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.

Examples

The following is sample output from the **show ephone-dn** command:

```
Router# show ephone-dn 7

50/0/7 INVALID

EFXS 50/0/7 Slot is 50, Sub-unit is 0, Port is 7
Type of VoicePort is EFXS
Operation State is UP
Administrative State is UP
No Interface Down Failure
Description is not set
Noise Regeneration is enabled
Non Linear Processing is enabled
Non Linear Mute is disabled
Non Linear Threshold is -21 dB
Music On Hold Threshold is Set to -38 dBm
In Gain is Set to 0 dB
Out Attenuation is Set to 0 dB
Echo Cancellation is enabled
Echo Cancellation NLP mute is disabled
```

```

Echo Cancellation NLP threshold is -21 dB
Echo Cancel Coverage is set to 8 ms
Playout-delay Mode is set to default
Playout-delay Nominal is set to 60 ms
Playout-delay Maximum is set to 200 ms
Playout-delay Minimum mode is set to default, value 4 ms
Playout-delay Fax is set to 300 ms
Connection Mode is normal
Connection Number is not set
Initial Time Out is set to 10 s
Interdigit Time Out is set to 10 s
Call Disconnect Time Out is set to 60 s
Ringing Time Out is set to 8 s
Wait Release Time Out is set to 30 s
Companding Type is u-law
Region Tone is set for US
Station name None, Station number None

```

```

Caller ID Info Follows:
Standard BELLCORE

```

```

Voice card specific Info Follows:
Digit Duration Timing is set to 100 ms

```

The following is sample output from the **show ephone-dn summary** command:

```
Router# show ephone-dn summary
```

```

PORT      DN STATE   CODEC   VAD VTSP STATE      VPM STATE
=====  =====  =====  === =====
50/0/1    DOWN      -       - -              EFXS_ONHOOK
50/0/2    DOWN      -       - -              EFXS_ONHOOK
50/0/3    DOWN      -       - -              EFXS_ONHOOK
50/0/4    INVALID   -       - -              EFXS_INIT
50/0/5    INVALID   -       - -              EFXS_INIT
50/0/6    INVALID   -       - -              EFXS_INIT

```

Table 8 provides an alphabetical listing of the command fields shown in the sample output.

Table 8 *show ephone-dn Field Descriptions*

Field	Description
Administrative State	Administrative (configured) state of the voice port.
Caller ID Info	Information about the caller ID.
CODEC	Codec type.
Companding Type	Not applicable to the Cisco IP phone.
Connection Mode	Not applicable to the Cisco IP phone.
Connection Number	Not applicable to the Cisco IP phone.
Description	Not applicable to the Cisco IP phone.
DN STATE	State of the Cisco IP phone line associated with directory numbers (DN).

Table 8 *show ephone-dn Field Descriptions (continued)*

Field	Description
Echo Cancellation	Not applicable to the Cisco IP phone.
Echo Cancel Coverage	Not applicable to the Cisco IP phone.
EFXS	The voice port type.
In Gain	Not applicable to the Cisco IP phone.
Initial Time Out	Amount of time the system waits for an initial input digit from the caller.
Interdigit Time Out	Amount of time the system waits for a subsequent input digit from the caller.
Music-On-Hold Threshold	Not applicable to the Cisco IP phone.
No Interface Down Failure	State of the interface.
Noise Regeneration	Not applicable to the Cisco IP phone.
Non-Linear Processing	Not applicable to the Cisco IP phone.
Operational State	Operational state of the voice port.
Out Attenuation	Not applicable to the Cisco IP phone.
Playout-delay Maximum	Not applicable to the Cisco IP phone.
Playout-delay Mode	Not applicable to the Cisco IP phone.
Playout-delay Nominal	Not applicable to the Cisco IP phone.
Port	Port number for this interface associated with the voice interface card.
Region Tone	Not applicable to the Cisco IP phone.
Ringling Time Out	Ringling time out duration.
Station name	Station name.
Station number	Station number.
Slot	Slot used in the voice interface card for this port.
Sub-unit	Subunit used in the voice interface card for this port.
Type of VoicePort	Indicates the voice port type.
VAD	Voice activity detection.
Voice card specific Info	Information specific to the voice card.
VPM STATE	State indication for the VPM software component.
VTSP STATE	State indication for the VTSP software component.
Wait Release Time Out	The time that a voice port stays in the call-failure state while the router sends a busy tone, reorder tone, or an out-of-service tone to the port.

Related Commands

Command	Description
show ephone	Displays Cisco IP phone output.

show ephone summary

To display a summary of all Cisco IP phones, use the **show ephone summary** EXEC command.

show ephone summary

Syntax Description This command has no arguments or keywords.

Command Modes EXEC

Command History	Release	Modification
	12.1(5)YD	This command was introduced on the Cisco 2600 series and Cisco 3600 series multiservice routers, and Cisco IAD2420 series integrated access devices (IADs).
	12.2(2)XG	This command was implemented on the Cisco MC3810-V3 concentrators.
	12.2(2)XB	This command was integrated into Cisco IOS Release 12.2(2)XB and was implemented on the Cisco 7200 series routers.
	12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T.

Usage Guidelines The **show ephone summary** command is similar to the **show ephone** command. However, the **show ephone summary** command does not display the destination numbers listed. If you do not specify a MAC address, you get the status information about all the Cisco IP phones.

Examples The following is sample output from the **show ephone summary** command:

```
Router# show ephone summary

ephone-1 Mac:0030.94C3.37CB TCP socket:[1] activeLine:0 REGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 debug:1
IP:10.1.1.1 Telecaster 7910 keepalive 75

ephone-2 Mac:0030.94C3.F96A TCP socket:[2] activeLine:0 REGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 debug:1
IP:10.1.1.2 Telecaster 7960 keepalive 90

ephone-3 Mac:0030.94C3.F43A TCP socket:[-1] activeLine:0 DECEASED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 debug:1
IP:10.2.1.1 Telecaster 7960 keepalive 2258

ephone-4 Mac:0030.94C3.F946 TCP socket:[-1] activeLine:0 REGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 debug:1
IP:0.0.0.0 Unknown 0 keepalive 0
```

Table 3-9 provides an alphabetical listing of the command fields in the sample output.

Table 3-9 *show ephone summary Field Descriptions*

Field	Description
activeLine	Indicates the line (button) on the phone that is in use. Zero indicates that no line is in use
debug	If set to 1, indicates debug for the phone is enabled; otherwise, it is set to 0.
ephone-1	Cisco IP phone tag number.
IP	Assigned IP address of the Cisco IP phone.
keepalive	Number of keepalive messages received from the Cisco IP phone by the router.
Mac	MAC address.
mediaActive	If set to 1, indicates that an active conversation is going on; otherwise it is set to 0.
offhook	If set to 1, indicates that the phone is off the hook.
REGISTERED	Indicates that the Cisco IP phone is active and registered. Alternatives states are UNREGISTERED (indicating that the connection to the Cisco IP phone was closed in a normal manner) and DECEASED (indicating that the connection to the Cisco IP phone was closed because of a keepalive timeout).
reset	Pending reset.
reset_sent	Request for reset sent to the Cisco IP phone.
ringing	If set to 1, indicates that the Cisco IP phone's ringer is turned on and the phone is ringing, otherwise set to 0.
TCP socket	Indicates the TCP socket number used to communicate with the Cisco IP phone. This can be correlated with the output of various other debug and show commands.
Telecaster <i>model number</i>	Indicates the type and model of the Cisco IP phone. This information is received from the phone during its registration with the router.

Related Commands

Command	Description
show ephone	Displays Cisco IP phone output.

time-format

To set the time display format on all Cisco IP phones attached to a router, use the **time-format** command in CallManager fallback configuration mode. To disable the time display format, use the **no** form of this command.

time-format { 12 | 24 }

no time-format { 12 | 24 }

Syntax Description	12	Sets format to 12-hour increments.
	24	Sets format to 24-hour increments.

Defaults 12-hour format.

Command Modes CallManager fallback configuration

Command History	Release	Modification
	12.2(2)XT	This command was introduced on the following platforms: Cisco 1750, Cisco 1751, Cisco 2600 and Cisco 3600 series multiservice routers; Cisco IAD2420 series IADs.
	12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers.
	12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
	12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.

Usage Guidelines The **time-format** command sets the time display format on all the Cisco IP phones attached to the router.

Examples The following example shows the time format on the Cisco IP phones being set to the 24-hour format:

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# time-format 24
```

Related Commands	Command	Description
	call-manager-fallback	Enables SRS Telephony feature support and enters CallManager fallback configuration mode.

timeouts interdigit

To configure the interdigit timeout value for all Cisco IP phones attached to a router, use the **timeouts interdigit** command in CallManager fallback configuration mode. To return the interdigit timeout value to its default, use the **no** form of this command.

timeouts interdigit *seconds*

no timeouts interdigit *seconds*

Syntax Description	<i>seconds</i>	Interdigit timeout duration, in seconds, for all Cisco IP phones. Valid entries are integers from 2 to 120. The default is 10.
---------------------------	----------------	--

Defaults	10 seconds
-----------------	------------

Command Modes	CallManager fallback configuration
----------------------	------------------------------------

Command History	Release	Modification
	12.2(2)XB	This command was introduced on the following platforms: Cisco 2600 series and Cisco 3600 series multiservice routers; Cisco IAD2420 series IADs; Cisco 7200 series routers.
	12.2(2)XT	This command was implemented on Cisco 1750 and Cisco 1751 multiservice routers.
	12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers.
	12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
	12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.

Usage Guidelines The **timeouts interdigit** command specifies how long, in seconds, the system waits after a caller enters the initial digit or a subsequent digit of the dialed string. The timeouts interdigit timer is activated when the 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. The default is 10 seconds.

To disable the timeouts interdigit timer, set the *seconds* value to 0.

Examples The following example sets the interdigit timeout value to 5 seconds for all Cisco IP phones:

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# timeouts interdigit 5
```

In this example, the 5 seconds refers to the elapsed time after which an incompletely dialed number times out. For example, if you dial nine digits (408555898) instead of the required ten digits (4085558984) you hear a busy tone after the 5 “timeout” seconds have elapsed.

Related Commands

Command	Description
call-manager-fallback	Enables SRS Telephony feature support and enters CallManager fallback configuration mode.
timeouts interdigit	Configures the interdigit timeout value for a specified voice port.

transfer-pattern

To allow Cisco IP phones to transfer telephone calls from callers outside the local IP network to another Cisco IP phone, use the **transfer-pattern** command in CallManager fallback configuration mode. To disable transfer of calls to other numbers, use the **no** form of this command.

transfer-pattern *transfer-pattern*

no transfer-pattern

Syntax Description	<i>transfer-pattern</i>	String of digits for permitted call transfers.
Defaults	This feature is enabled by default.	
Command Modes	CallManager fallback configuration	
Command History	Release	Modification
	12.1(5)YD	This command was introduced on the following platforms: Cisco 2600 series and Cisco 3600 series multiservice routers, and Cisco IAD2420 series IADs.
	12.2(2)XT	This command was implemented on Cisco 1750 and Cisco 1751 multiservice routers.
	12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers.
	12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
	12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.

Usage Guidelines The **transfer-pattern** command allows you to transfer a call from a non-IP phone number to another Cisco IP phone on the same IP network using the specified transfer pattern. By default, all Cisco IP phone directory numbers or virtual voice ports are allowed as transfer targets.

Examples The following example sets a transfer pattern:

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# transfer-pattern 52540..
```

A maximum of 32 transfer patterns can be entered. In this example, 55540.. (the two decimal points are used here as wildcards) permits transfers to any numbers in the range from 555-4000 to 555-4099.

transfer-pattern

Related Commands

Command	Description
call-manager-fallback	Enables SRS Telephony feature support and enters CallManager fallback configuration mode.

translate

To apply a translation rule to modify the phone number dialed by any Cisco IP phone user while CallManager fallback is active, use the **translate** command in CallManager fallback configuration mode. To disable this feature, use the **no** form of this command.

```
translate { called | calling } translation-rule-tag
```

```
no translate { called | calling } translation-rule-tag
```

Syntax Description

called	Translation rule to apply to the number called by a Cisco IP phone.
calling	Translation rule to apply to the calling party number sent in the call setup message for calls originated from a Cisco IP phone.
<i>translation-rule-tag</i>	Tag number by which the rule set is referenced. This is an arbitrarily chosen number. The range is from 1 to 2147483647.

Defaults

No default behavior or values.

Command Modes

CallManager fallback configuration

Command History

Release	Modification
12.2(2)XT	This command was introduced on the following platforms: Cisco 1750, Cisco 1751, Cisco 2600 series and Cisco 3600 series multiservice routers; Cisco IAD2420 series IADs.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers.
12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.

Usage Guidelines

The **translate** command allows you to apply a previously configured number-translation rule to modify the number dialed by a specific extension. Translation rules are a powerful general-purpose number-manipulation mechanism that performs operations such as automatically adding telephone area and prefix codes to dialed numbers.

Examples

The following example applies translation rule 20 to the inbound called number:

```
Router(config)# translation-rule 20
Router(config-translate)# rule 0 1234 2345 abbreviated
Router(config-translate)# exit
Router(config)# call-manager-fallback
```

```
Router(config-cm-fallback) translate called 20
```

Related Commands

Command	Description
call-manager-fallback	Enables SRS Telephony feature support and enters CallManager fallback configuration mode.
translation-rule	Creates a translation name and enters translation-rule configuration mode.

user-locale

To set language for displays on the Cisco IP Phone 7940 and Cisco IP Phone 7960 by country, use the **user-locale** command in call-manager-fallback configuration mode. To disable the country selection and use the default country (United State), use the **no** form of this command.

user-locale *country-code*

no user locale *country-code*

Syntax Description	<i>country-code</i>	For systems running Cisco CallManager V3.2 use, one of the following ISO-3166 codes: <ul style="list-style-type: none"> • France—FR • Germany—DE • Italy—IT • Portugal—PT • Spain—ES • United States—US (default)
---------------------------	---------------------	---

Defaults The default country code is US (United States).

Command Modes Call-manager-fallback configuration

Command History	Release	Modification
	12.2(11)YT	This command was introduced for use on the Cisco IP Phone 7940 and Cisco IP Phone 7960 for Cisco SRS Telephony systems running Cisco CallManager V3.2. Support includes country codes for Austria, Canada, Denmark, France, Germany, Italy, Netherlands, Norway, Portugal, Spain, Sweden, Switzerland, United Kingdom, United States (default).

Usage Guidelines All of the *country-code* options are available to Cisco Survivable Remote Site Telephony systems running under Cisco CallManager V3.2. Systems running Cisco CallManager prior to V3.2 can use only the default country, United States (US).

To deactivate one country code and activate another, use the **no network-locale** command.

Examples The following example shows how to set the user locale the ISO-3166 code for Spain:

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# user-locale ES
```

Related Commands

Command	Description
call-manager-fallback	Enables SRS Telephony feature support and enters CallManager fallback configuration mode.

vm-integration

To enter voice-mail integration configuration mode and enable voice-mail integration with dual tone multifrequency (DTMF) and analog voice-mail on systems connected to the Public Switched Telephone Network (PSTN) through the Foreign Exchange Office (FXO) or Foreign Exchange Service (FXS) ports, use the **vm-integration** command in global configuration mode. To disable voice-mail integration, use the **no** form of this command.

vm-integration

no vm-integration

Syntax Description This command has no arguments or keywords.

Defaults No default behavior or values

Command Modes Global configuration

Command History	Release	Modification
	12.2(2)XT	This command was introduced on the following platforms: Cisco 1750, Cisco 1751, Cisco 2600 series, and Cisco 3600 series multiservice routers; and Cisco IAD2420 series IADs.
	12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745 routers.
	12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
	12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.
	12.2(13)T	This command was introduced to Cisco SRS Telephony Version 2.02

Usage Guidelines The vm-integration command allows integration with DTMF and analog voice-mail systems. Typically, calls are forwarded to voice mail when the called number is busy or does not answer. To play personal greetings, a voice-mail system requires the number of the phone that does not answer, known as the redirected dialed number identification service (RDNIS). During fallback, a RDNIS must to be passed to voice-mail systems through the PSTN. If the trunks are FXS or FXO, the vm-integration command can facilitate the in-band passing of RDNIS information to a voice-mail system.

The vm-integration command does not support Primary Rate Interface (PRI) or Basic Rate Interface (BRI). If trunks are an Integrated Services Digital Network (ISDN), then it may be possible to pass the RDNIS as part of Q.931 signalling. Cisco SRST includes RDNIS in the Q.931 setup signalling by default. Note, however, that some carriers drop RDNIS, thus nullifying this solution

Examples

The following example shows how to enter the voice-mail integration configuration mode:

```
Router(config) vm-integration
Router(config-vm-integration)
```

Related Commands

Command	Description
pattern direct	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when the user presses the messages button on the phone.
pattern ext-to-ext busy	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 voicemail.
pattern ext-to-ext no-answer	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 voicemail.
pattern trunk-to-ext busy	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 voicemail.
pattern trunk-to-ext no-answer	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 voicemail.

voicemail

To configure the telephone number that is speed-dialed when the messages button on a Cisco IP phone is pressed, use the **voicemail** command in CallManager fallback configuration mode. To disable the messages button, use the **no** form of this command.

voicemail *phone-number*

no voicemail

Syntax Description	<i>phone-number</i>	Phone number configured as a speed-dial number for retrieving messages.
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Defaults	No phone number is configured, and the messages button is ineffective.
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Command Modes	CallManager fallback configuration
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Command History	Release	Modification
	12.1(5)YD	This command was introduced on the following platforms: Cisco 2600 series and Cisco 3600 series multiservice routers, and Cisco IAD2420 series IADs.
	12.2(2)XT	This command was implemented on Cisco 1750 and Cisco 1751 multiservice routers.
	12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers.
	12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
	12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.

Usage Guidelines	The voicemail command configures the telephone number that is speed-dialed when the messages button on a Cisco IP phone is pressed. The same voicemail telephone number is configured for all Cisco IP phones connected to the router.
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Examples	The following example specifies 4085551000 as the speed-dial number that is dialed to retrieve messages when the messages button is pressed:
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```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# voicemail 914085551000
```

The number 914085551000 is called when the Cisco IP phone messages button is pressed to retrieve messages.

Related Commands

Command	Description
call-manager-fallback	Enables SRS Telephony feature support and enters CallManager fallback configuration mode.



Glossary



Note

Refer to the *Internetworking Terms and Acronyms* for terms not included in this glossary.

AA—auto-attendant. A software application that provides automated operator assistance, provides callanswering with a menu of selection for the caller, such as pressing “0” for the operator, pressing “1” to enter an extension number and pressing “2” to enter a person's name. When the application receives a valid extension, it transfers the call.

ADSL—asymmetric digital subscriber line. One of four DSL technologies. ADSL is designed to deliver more bandwidth downstream (from the central office to the customer site) than upstream. Downstream rates range from 1.5 to 9 Mbps, whereas upstream bandwidth ranges from 16 to 640 kbps. ADSL transmissions work at distances up to 18,000 feet (5,488 meters) over a single copper twisted pair.

ANI—automatic number identification. SS7 (signaling system 7) feature in which a series of digits, either analog or digital, are included in the call, identifying the telephone number of the calling device. In other words, ANI identifies the number of the calling party.

AVVID—Cisco Architecture for Voice, Video and Integrated Data. Cisco AVVID provides the framework for today's Internet business solutions. As the industry's only enterprise-wide, standards-based network architecture, Cisco AVVID provides the road map for combining your business and technology strategies into one cohesive model.

Cisco AVVID provides the baseline infrastructure that enables enterprises to design networks that scale to meet Internet business demands. Cisco AVVID delivers the e-business infrastructure and intelligent network services that are essential for rapid deployment of emerging technologies and new Internet business solutions.

BRI—Basic Rate Interface. ISDN interface composed of two B channels and one D channel for circuit-switched communication of voice, video, and data.

CCM—Cisco CallManager. Cisco CallManager is the software-based call-processing component of the Cisco IP telephony solution, part of Cisco AVVID. The software extends enterprise telephony features and functions to packet telephony network devices such as IP phones, media processing devices, voice-over-IP (VoIP) gateways, and multimedia applications.

COR—Class of restriction. Functionality that provides the capability to deny certain call attempts based on 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, to 900 numbers), and applies different restrictions to call attempts from different originators. COR specifies which incoming dial peer can use which outgoing dial peer to make a call.

DHCP—Dynamic Host Configuration Protocol. Provides a mechanism for allocating IP addresses dynamically so that addresses can be reused when hosts no longer need them.

DID—direct inward dial. Allows a user outside a company to dial an internal extension number without needing to pass through an operator or an attendant. The dialed digits are passed to the PBX, which then completes the call.

DID/DNIS—Direct Inward Dialing/Dialed Number Identification Service. When a call arrives at an ACD or PBX, the carrier sends a digital code on the trunk line. The switch can read this code to determine how it should dispatch the call. Typically, this value is the specific number dialed by the user. By mapping each possible code with an internal extension, the switch can provide direct inward dialing (DID).

DNs—directory numbers.

DOD—Direct Outward Dialing.

FXO—Foreign Exchange Office. An FXO interface connects to the Public Switched Telephone Network (PSTN) central office and is the interface offered on a standard telephone. Cisco's FXO interface is an RJ-11 connector that allows an analog connection at the PSTN's central office or to a station interface on a PBX.

FXS—Foreign Exchange Station. An FXS interface connects directly to a standard telephone and supplies ring, voltage, and dial tone. Cisco's FXS interface is an RJ-11 connector that allows connections to basic telephone service equipment, keysets, and PBXs.

ISDN—Integrated Services Digital Network. Communication protocol offered by telephone companies that permits telephone networks to carry data, voice, and other source traffic.

NPE—

PRI—Primary Rate Interface. ISDN interface to primary rate access. Primary rate access consists of a single 64-kbps D channel plus 23 (T1) or 30 (E1) B channels for voice or data.

PSTN—public switched telephone network. General term referring to the variety of telephone networks and services in place worldwide. Sometimes called POTS.

QoS—quality of service. Measure of performance for a transmission system that reflects its transmission quality and service availability.

SCCP—Signaling Connection Control Part. Trillium software that supports routing and translation and management functions and data transfer without logical signaling connections.

SNAP—Subnetwork Access Protocol. Internet protocol that operates between a network entity in the subnetwork and a network entity in the end system. SNAP specifies a standard method of encapsulating IP datagrams and ARP messages on IEEE networks. The SNAP entity in the end system makes use of the services of the subnetwork and performs three key functions: data transfer, connection management, and QoS selection.

SRS—Survivable Remote Site. As in Cisco SRS Telephony.

TCP—Transmission Control Protocol. Connection-oriented transport layer protocol that provides reliable full-duplex data transmission. TCP is part of the TCP/IP protocol stack.