



Configuring Call Handling

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

This chapter also describes support for standardized RFC 3261 features for SIP phones. Features include call blocking and call forwarding.



Note Configuring Call Handling for SIP phones applies to versions 4.0 and 3.4 only.

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Prerequisites for Configuring SIP SRST Features Using Back-to-Back User Agent Mode

- Complete the prerequisites documented in the [“Prerequisites for Configuring Cisco Unified SIP SRST” section on page 9](#) section in the [“Cisco Unified SRST Feature Overview” section on page 1](#).
- Configure the SIP registrar. The SIP registrar gives users control of accepting or rejecting registrations. To configure acceptance of incoming SIP Register messages, see the [“” section on page 165](#).

Restrictions for Configuring SIP SRST Features Using Back-to-Back User Agent Mode

- See the restrictions documented in the [“Restrictions for Configuring Cisco Unified SIP SRST” section on page 10](#) section in the [“Cisco Unified SRST Feature Overview” section on page 1](#).

Information About Configuring SCCP SRST Call Handling

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

- [H.323 VoIP Call Preservation Enhancements for WAN Link Failures, page 184](#)
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H.323 VoIP Call Preservation Enhancements for WAN Link Failures

H.323 VoIP call preservation enhancements for WAN link failures sustains connectivity for H.323 topologies where signaling is handled by an entity, such as Cisco Unified Communications Manager, that is different from the other endpoint and brokers signaling between the two connected parties.

Call preservation is useful when a gateway and the other endpoint (typically a Cisco Unified IP phone) are collocated at the same site and call agent is remote and therefore more likely to experience connectivity failures.

For configuration information see “Configuring H.323 Gateways” chapter in [Cisco IOS H.323 Configuration Guide](#), Release 12.4T.

Toll Fraud Prevention

When a Cisco router platform is installed with a voice-capable Cisco IOS software image, appropriate features must be enabled on the platform to prevent potential toll fraud exploitation by unauthorized users. Deploy these features on all Cisco router Unified Communications applications that process voice calls, such as Cisco Unified Communications Manager Express (CME), Cisco Survivable Remote Site Telephony (SRST), Cisco Unified Border Element (UBE), Cisco IOS-based router and standalone analog and digital PBX and public-switched telephone network (PSTN) gateways, and Cisco contact-center VoiceXML gateways. For more information about Toll Fraud Prevention, see *Toll Fraud Prevention* in *Cisco Unified Communications Manager Express System Administration Guide*.

Information About Configuring SIP SRST Features Using Back-to-Back User Agent Mode

A Cisco Unified SRST system can now support SIP phones with standard-based RFC 3261 feature support locally and across SIP WAN networks. With Cisco Unified SIP SRST, SIP phones can place calls across SIP networks with similar features, as SCCP phones do. For example, most SCCP phone features such as caller ID, speed dial, and redial are supported now on SIP networks, which gives users the opportunity to choose SCCP or SIP.

Cisco Unified SIP SRST also uses a back-to-back user agent (B2BUA), which is a separate call agent that has more features than Cisco SIP SRST 3.0, which used a redirect server that only accepted and forwarded calls. The main advantage of a B2BUA call agent is in call forwarding, because it forwards calls on behalf of the phone. In addition, it maintains a presence as call middleman in the call path.

Cisco SIP SRST 3.4 supports the following call combinations:

- SIP phone to SIP phone
- SIP phone to PSTN / router voice port
- SIP phone to SCCP phone

Cisco Unified SIP SRST and Cisco SIP Communications Manager Express Feature Crossover

The **voice register dn**, **voice register global** and **voice register pool** configuration mode commands are accessible in both Cisco Unified SIP CME and Cisco Unified SIP SRST modes of operation. However, not all of the commands within these modes are intended for use in SIP SRST mode. [Table 8-1](#) provides a summary guide to which commands are relevant to the CME or SRST modes of operation.

For more detailed information, refer to the command reference pages for each of the individual commands.



Note

[Table 8-1](#) is not all-inclusive; additional commands may exist.

Table 8-1 *Version 3.4 New or Enhanced Commands for Cisco Unified SRST and Cisco Unified CME (Sorted by Configuration Mode)*

Command	Dial Peer	Voice Register Mode	Configurable for Cisco Unified (SIP) CME and Cisco Unified SIP SRST	Applicable to Cisco Unified (SIP) CME Only
after-hour exempt	X	dn	X	—
auto-answer	—	dn	—	X
call forward	X	dn	X	—
huntstop	X	dn	X	—
label	—	dn	—	X
name	—	dn	—	X
number	X	dn	X	—
preference	X	dn	X	—
application	X	global	X	—
authenticate	—	global	—	X
create	—	global	—	X
date-format	—	global	—	X
dst	—	global	—	X
external ring	—	global	X	—
file	—	global	—	X
hold-alert	—	global	—	X
load	—	global	—	X
logo	—	global	—	X
max-dn	—	global	X	—
max-pool	—	global	X	—
max-redirect	—	global	—	X
mode	—	global	X	—
mwi	—	global	—	X
reset	—	global	—	X
tftp-path	—	global	—	X
timezone	—	global	—	X
upgrade	—	global	—	X
url	—	global	—	X
voicemail	—	global	—	X
after-hour exempt	X	pool	X	—
application	X	pool	X	—
call-forward	—	pool	X	—
call-waiting	—	pool	—	X

Table 8-1 Version 3.4 New or Enhanced Commands for Cisco Unified SRST and Cisco Unified CME (Sorted by Configuration Mode) (continued)

Command	Dial Peer	Voice Register Mode	Configurable for Cisco Unified (SIP) CME and Cisco Unified SIP SRST	Applicable to Cisco Unified (SIP) CME Only
codec	X	pool	X	—
description	—	pool	—	X
dnd-control	—	pool	—	X
dtmf-relay	—	pool	X	—
id	—	pool	X	—
keep-conference	—	pool	—	X
max-pool	—	pool	X	—
number	X	pool	X	—
preference	X	pool	X	—
proxy	X	pool	X	—
reset	—	pool	—	X
speed-dial	—	pool	—	X
template	—	pool	—	X
translation-profile	X	pool	X	—
type	—	pool	—	X
username	—	pool	—	X
vad	X	pool	X	—
anonymous	—	template	—	X
caller-id	—	template	—	X
conference	—	template	—	X
dnd-control	—	template	—	X
forward	—	template	—	X
transfer	—	template	—	X

How to Configure Cisco Unified SCCP SRST

Setting up call handling involves the following set of tasks:

- [Configuring Incoming Calls, page 188](#)
- [Configuring Outgoing Calls, page 207](#)
- [Configuring Call Blocking Based on Time of Day, Day of Week, or Date, page 228](#)

Configuring Incoming Calls

Incoming call configuration can include the following tasks:

- Call Forwarding and Rerouting
 - [Configuring Call Forwarding During a Busy Signal or No Answer, page 188](#) (Optional)
 - [Configuring Call Rerouting, page 190](#) (Optional)
 - [Configuring Call Pickup, page 193](#) (Optional)
 - [Configuring Transfer Digit Collection Method, page 196](#)
- Phone Number Conversion and Translation
 - [Configuring Global Prefixes, page 197](#) (Optional)
 - [Enabling Digit Translation Rules, page 199](#) (Optional)
 - [Enabling Translation Profiles, page 200](#) (Optional)
 - [Verifying Translation Profiles, page 203](#) (Optional)
- Hunting and Ringing Timeout Behavior
 - [Configuring Dial-Peer and Channel Hunting, page 204](#) (Optional)
 - [Configuring Busy Timeout, page 205](#) (Optional)
 - [Configuring the Ringing Timeout Default, page 206](#) (Optional)

Configuring Call Forwarding During a Busy Signal or No Answer

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

SUMMARY STEPS

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

DETAILED STEPS

	Command or Action	Purpose
Step 1	<code>call-manager-fallback</code> Example: Router(config)# call-manager-fallback	Enters call-manager-fallback configuration mode.
Step 2	<code>call-forward busy directory-number</code> Example: Router(config-cm-fallback)# call-forward busy 50..	Configures call forwarding to another number when the Cisco IP phone is busy. <ul style="list-style-type: none"> <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.
Step 3	<code>call-forward noan directory-number timeout seconds</code> Example: Router(config-cm-fallback)# call-forward noan 5005 timeout 10	Configures call forwarding to another number when no answer is received from the Cisco IP phone. <ul style="list-style-type: none"> <i>directory-number</i>: Selected directory number representing a fully qualified E.164 number or a local extension number. This number can contain "." wildcard characters that correspond to the right-justified digits in the directory number extension. timeout seconds: Sets the waiting time, in seconds, before the call is forwarded to another phone. The <i>seconds</i> range is from 3 to 60000.
Step 4	<code>exit</code> Example: Router(config-cm-fallback)# exit	Exits call-manager-fallback configuration mode.

Examples

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

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

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

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

Configuring Call Rerouting



Note

We recommend the **alias** command, which obsoletes the **default-destination** command, instead of the **default-destination** command.

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

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

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

Call Forward Destination

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



Note

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

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

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

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

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

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

Huntstop on an Individual Alias

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

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

SUMMARY STEPS

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

DETAILED STEPS

	Command or Action	Purpose
Step 1	<p><code>call-manager-fallback</code></p> <p>Example: Router(config)# <code>call-manager-fallback</code></p>	Enters call-manager-fallback configuration mode.
Step 2	<p><code>alias tag number-pattern to alternate-number</code> [<code>preference preference-value</code>] [<code>cfw number</code> <code>timeout timeout-value</code>] [<code>huntstop</code>]</p> <p>Example: Router(config-cm-fallback)# <code>alias 1 60.. to</code> <code>5001 preference 1 cfw 2000 timeout 10</code></p>	<p>Creates a set rules for rerouting calls to sets of phones that are unavailable during Cisco Unified CM fallback.</p> <ul style="list-style-type: none"> • <i>tag</i>: Identifier for alias rule range. The range is from 1 to 50. • <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. The alternate number has to be a specific extension that belongs to an IP phone that is actively registered on the Cisco Unified SRST router. The alternate telephone number can be used in multiple alias commands. • preference preference-value (Optional). Assigns a dial-peer preference value to the alias. The preference value of the associated dial peer is from 0 to 10. Use with the max-dn command. • cfw number (Optional). The cfw keyword allows users to set call forward busy and call forward no answer to a set string and override globally configured call forward settings. • timeout timeout-value (Optional). Sets the ring no-answer timeout duration for call forwarding, in seconds. Range is from 3 to 60000. • huntstop (Optional). Stops call hunting after trying the alternate number.
Step 3	<p><code>max-dn max-directory-numbers [dual-line]</code> [<code>preference preference-order</code>]</p> <p>Example: Router(config-cm-fallback)# <code>max-dn 10</code> <code>preference 2</code></p>	<p>Sets the maximum possible number of directory numbers or virtual voice ports that can be supported by a router and sets the global preference for creating the VoIP dial peers for all directory numbers that are associated with the primary number.</p> <ul style="list-style-type: none"> • Using the max-dn command sets the preference for the default dial peers created with the alias command. • When configuring call rerouting, set the max-dn preference to a higher numeric preference than the preference that was set with the alias command.

	Command or Action	Purpose
Step 4	<code>end</code> Example: Router(config-cm-fallback)# end	Returns to privileged EXEC mode.
Step 5	<code>show dial-peer voice summary</code> Example: Router# show dial-peer voice summary	Displays information for voice dial peers. <ul style="list-style-type: none"> If you suspect a problem with the dial peers, use this command to display the dial peers created by the alias command.

Examples

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

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

Configuring Call Pickup

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

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

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



Note

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

SUMMARY STEPS

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

DETAILED STEPS

	Command or Action	Purpose
Step 1	<code>call-manager-fallback</code> Example: Router(config)# <code>call-manager-fallback</code>	Enters call-manager-fallback configuration mode.
Step 2	<code>no huntstop</code> Example: Router(config-cm-fallback)# <code>no huntstop</code>	Disables huntstop.
Step 3	<code>alias tag number-pattern to alternate-number</code> Example: Router(config-cm-fallback)# <code>alias 1 8005550100 to 5001</code>	Creates a set rules for rerouting calls to sets of phones that are unavailable during Cisco Unified CM fallback. <ul style="list-style-type: none"> • <i>tag</i>: Identifier for alias rule range. The range is from 1 to 50. • <i>number-pattern</i>: Pattern to match the incoming telephone number. This pattern may include wildcards. • <i>to</i>: 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. The alternate number has to be a specific extension that belongs to an IP phone that is actively registered on the Cisco Unified SRST router. The alternate telephone number can be used in multiple alias commands.
Step 4	<code>pickup telephone-number</code> Example: Router(config-cm-fallback)# <code>pickup 8005550100</code>	Enables the PickUp soft key on all Cisco Unified IP Phones, allowing an external Direct Inward Dialing (DID) call coming into one extension to be picked up from another extension during SRST. The <i>telephone-number</i> argument is the telephone number to match an incoming called number.
Step 5	<code>end</code> Example: Router(config-cm-fallback)# <code>end</code>	Returns to privileged EXEC mode.

Examples

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

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

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

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

Configuring Consultative Transfer

Before Cisco Unified SRST 4.3, the consultative transfer feature played a dial tone and collected dialed digits until the digits matched the pattern for consultative transfer, blind transfer, or PSTN transfer blocking. The after-hours blocking criteria was applied after the consultative transfer digit collection and pattern matching.

The new feature modifies the transfer digit-collection process to make it consistent with Cisco Unified Communications Manager. This feature is supported only if the **transfer-system full-consult** command (default) is specified in call-manager-fallback configuration mode and an idle line or channel is available for seizing, digit collection, and dialing.

Two lines are required for a consultative transfer. When the transferor party is an octo-line directory number, Cisco Unified SRST selects the next available idle channel on that directory number. If the maximum number of channels of the directory number are in use, another idle line on the transferor phone is considered. If the **auto-line** command is configured on the phone, the specified autoline (if idle) takes precedence over other nonauto lines. If no idle line is available on the transferor phone, a blind transfer is initiated instead of the consultative transfer.

During the consultative transfer, the transferor line to the transferee party is locked on the transferor phone to prevent it from being stolen by other phones sharing the same directory number. When the user presses the Transfer soft key for a consultative transfer, the Transfer soft key does not display while digits are being dialed and collected on this seized consultative transfer call leg. The method for consultative transfer pattern matching, blind transfer, PSTN transfer blocking, or after-hour blocking criteria remain the same although the manipulation after the matching is different. When the criteria for the blind transfer is met, Cisco Unified SMST terminates the consultative transfer call leg, informs the Cisco IOS software to transfer the call, and then terminates the original call bubble. The PARK FAC code is handled in the same way as by a new call which requires that a ten-second timer is applied by the Cisco IOS software.



Note

The enhancement, by default, collects the transfer digits from the new call leg. If required, you can configure the system to collect the transfer digits from the original call leg. See the [“Configuring Transfer Digit Collection Method”](#) section on page 196.

The error handling for transfer failure because of transfer blocking or interdigit timer expiration remains. It includes displaying an error message on the prompt line and logging it if “debug ephone error” is enabled, playing a fast-busy or busy tone, and terminating the consultative transfer call leg.

No new configuration is required to support these enhancements.

Conference Calls

No configuration steps are required for these conference call enhancements.

Single-Line Directory Number

If the initiating party for the conference call is a single-line directory number and the phone has multiple directory numbers configured, the system selects another directory number's idle channel for creating the conference. If there are multiple directory numbers (dual-line or single-line directory numbers) on the phone and each has calls on hold, the system prompts the user to select a line for the conference call.

Dual-Line Directory Number

If the initiating party for the conference call is a dual-line directory number, the system selects another idle channel from the dual-line directory number. If the selected channel has a call on hold, the conference operation will automatically select the hold channel and create the conference.

Octo-Line Directory Number

If the initiating party for the conference call is an octo-line directory number, the system selects an idle channel from the initiating party directory number and the user must establish a new call to complete the conference. If there is no idle channel on the same directory number, other idle directory numbers or channels on the same phone are not selected. If there are existing calls on hold on the other channels of the same directory number or other directory numbers, the user will not have the option to select them to join the conference. If there is no idle channel on the same directory number, the conference will abort with a No Line Available message.

Configuring Transfer Digit Collection Method

By default, transfer digits are collected from the new call leg. To change the transfer digit collection method, perform the following steps.

Prerequisites for Cisco Unified SRST 4.3

- Cisco Unified SRST 4.3
- Cisco Unified CM 6.0
- Cisco IOS Release 12.4(15)XZ

Restrictions for Cisco Unified SRST 4.3

- The Cisco 3200 Series Mobile Access Router does not support SRST.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **call-manager-fallback**
4. **transfer-digit-collect {new-call | orig-call}**
5. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	<code>enable</code> Example: <code>Router> enable</code>	Enables privileged EXEC mode. <ul style="list-style-type: none">• Enter your password if prompted.
Step 2	<code>configure terminal</code> Example: <code>Router# configure terminal</code>	Enters global configuration mode.
Step 3	<code>call-manager-fallback</code> Example: <code>Router(config)# call-manager-fallback</code>	Enters call-manager-fallback configuration mode.
Step 4	<code>transfer-digit-collect {new-call orig-call}</code> Example: <code>Router(config-cm-fallback)# transfer-digit-collect orig-call</code>	Selects the digit-collection method used for consultative call transfers. <ul style="list-style-type: none">• new-call: Digits are collected from the new call leg.• orig-call: Digits are collected from the original call-leg. This was the default behavior in versions before Cisco Unified SRST 4.3.
Step 5	<code>end</code> Example: <code>Router(config)# end</code>	Returns to privileged EXEC mode.

Examples

The following example shows the **transfer-digit-collect** method set to the legacy value of orig-call:

```
!
call-manager-fallback
  transfer-digit collect orig-call
!
```

Configuring Global Prefixes

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

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

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

SUMMARY STEPS

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

DETAILED STEPS

	Command or Action	Purpose
Step 1	call-manager-fallback	Enters call-manager-fallback configuration mode.
	Example: Router(config)# call-manager-fallback	
Step 2	dialplan-pattern <i>tag pattern extension-length length</i> [extension-pattern <i>extension-pattern</i>] [no-reg]	Creates a global prefix that can be used to expand the abbreviated extension numbers into fully qualified E.164 numbers
	Example: Router(config-cm-fallback)# dialplan-pattern 1 4085550100 extension-length 3 extension-pattern 4..	<ul style="list-style-type: none"> • <i>tag</i>: Dial-plan string tag used before a 10-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 wildcard markers or dots (.) for the remainder of the extension number digits. • extension-length: Sets the number of extension digits. • <i>length</i>: The number of extension digits. The range is from 1 to 32. • extension-pattern: (Optional) Sets an extension number's leading digit pattern when it is different from the E.164 telephone number's leading digits defined in the <i>pattern</i> argument. • <i>extension-pattern</i>: (Optional) The extension number's leading digit pattern. Consists of one or more digits and wildcard markers or dots (.). For example, 5.. would include extension 500 to 599; 5... would include 5000 to 5999. • no-reg: (Optional) Prevents the E.164 numbers in the dial peer from registering with the gatekeeper.
	Note This example maps all extension numbers 4xx to the PSTN number 40855501xx, so that extension 412 corresponds to 4085550112.	
Step 3	exit	Exits call-manager-fallback configuration mode.
	Example: Router(config-cm-fallback)# exit	

Examples

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

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

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

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

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

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

Enabling Digit Translation Rules

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



Note

Digit translation rules have many applications and variations. For further information about them, see [Cisco IOS Voice Configuration Library](#).

If you are running Cisco SRST 3.2 and later or Cisco Unified SRST 4.0 and later, use the configuration described in the [“Enabling Translation Profiles” section on page 200](#) instead of using the **translate** command as described below. Translation Profiles are new to Cisco SRST 3.2 and provide added capabilities.

Translation rules can be used as follows:

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

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

SUMMARY STEPS

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

DETAILED STEPS

	Command or Action	Purpose
Step 1	<code>call-manager-fallback</code> Example: Router(config)# call-manager-fallback	Enters call-manager-fallback configuration mode.
Step 2	<code>translate {called calling}</code> <code>translation-rule-tag</code> Example: Router(config-cm-fallback)# translate called 20	Applies a translation rule to modify the phone number dialed or received by any Cisco Unified IP Phone user while Cisco Unified CM fallback is active. <ul style="list-style-type: none"> • called: Applies the translation rule to an outbound call number. • calling: Applies the translation rule to an inbound call number. • <i>translation-rule-tag:</i> The reference number of the translation rule from 1 to 2147483647.
Step 3	<code>exit</code> Example: Router(config-cm-fallback)# exit	Exits call-manager-fallback configuration mode.

Examples

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

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

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

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

Enabling Translation Profiles

Cisco SRST 3.2 and later and Cisco Unified SRST 4.0 and later support translation profiles. Translation profiles are the suggested way to allow you to group translation rules and provide instructions on how to apply the translation rules to the following:

- Called numbers
- Calling numbers
- Redirected called numbers

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

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

**Note**

For Cisco SRST 3.2 and later versions and Cisco Unified SRST 4.0 and later versions, use the **voice translation-rule** and **translation-profile** commands shown below instead of the translation rule configuration described in the “[Enabling Digit Translation Rules](#)” section on page 199. Voice translation rules are a separate feature from translation rules. See the **voice translation-rule** command in *Cisco IOS Voice Command Reference* for more information and the *VoIP Gateway Trunk and Carrier Based Routing Enhancements* documentation for more general information on translation rules and profiles.

SUMMARY STEPS

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

DETAILED STEPS

	Command or Action	Purpose
Step 1	<pre>voice translation-rule <i>number</i></pre> <p>Example: Router(config)# voice translation-rule 1</p>	<p>Defines a translation rule for voice calls and enters voice translation-rule configuration mode.</p> <ul style="list-style-type: none"> <i>number</i>: Number that identifies the translation rule. Range is from 1 to 2147483647.
Step 2	<pre>rule <i>precedence</i>/<i>match-pattern</i>/ /<i>replace-pattern</i>/</pre> <p>Example: Router(cfg-translation-rule)# rule 1/^9/ //</p>	<p>Defines a translation rule.</p> <ul style="list-style-type: none"> <i>precedence</i>: Priority of the translation rule. Range is from 1 to 15. <i>match-pattern</i>: Stream editor (SED) expression used to match incoming call information. The slash (/) is a delimiter in the pattern. <i>replace-pattern</i>: SED expression used to replace the match pattern in the call information. The slash (/) is a delimiter in the pattern.
Step 3	<pre>exit</pre> <p>Example: Router(cfg-translation-rule)# exit</p>	<p>Exits voice translation-rule configuration mode.</p>
Step 4	<pre>voice translation-profile <i>name</i></pre> <p>Example: Router(config)# voice translation-profile name1</p>	<p>Defines a translation profile for voice calls.</p> <ul style="list-style-type: none"> <i>name</i>: Name of the translation profile. Maximum length of the voice translation profile name is 31 alphanumeric characters.
Step 5	<pre>translate {called calling redirect-called} <i>translation-rule-number</i></pre> <p>Example: Router(cfg-translation-profile)# translate called 1</p>	<p>Associates a voice translation rule with a voice translation profile.</p> <ul style="list-style-type: none"> called: Associates the translation rule with called numbers. calling: Associates the translation rule with calling numbers. redirect-called: Associates the translation rule with redirected called numbers. <i>translation-rule-number</i>: The reference number of the translation rule from 1 to 2147483647.
Step 6	<pre>exit</pre> <p>Example: Router(cfg-translation-profile)# exit</p>	<p>Exits translation-profile configuration mode.</p>
Step 7	<pre>call-manager-fallback</pre> <p>Example: Router(config)# call-manager-fallback</p>	<p>Enters call-manager-fallback configuration mode.</p>

	Command or Action	Purpose
Step 8	<pre>translation-profile {incoming outgoing} name</pre> <p>Example: Router(config-cm-fallback)# translation-profile outgoing name1</p>	<p>Assigns a translation profile for incoming or outgoing call legs on a Cisco IP phone.</p> <ul style="list-style-type: none"> • incoming: Applies the translation profile to incoming calls. • outgoing: Applies the translation profile to outgoing calls. • <i>name:</i> The name of the translation profile.
Step 9	<pre>exit</pre> <p>Example: Router(config-cm-fallback)# exit</p>	<p>Exits call-manager-fallback configuration mode.</p>

Examples

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

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

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

Verifying Translation Profiles

To verify translation profiles, perform the following steps.

SUMMARY STEPS

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

DETAILED STEPS

	Command or Action	Purpose
Step 1	<pre>show voice translation-rule <i>number</i></pre> <p>Example: Router# show voice translation-rule 6 Translation-rule tag: 6 Rule 1: Match pattern: 65088801.. Replace pattern: 6508880101 Match type: none Replace type: none Match plan: none Replace plan: none </p>	Use this command to verify the translation rules that you have defined for your translation profiles.
Step 2	<pre>test voice translation-rule <i>number input-test-string</i> [<i>type match-type</i> [<i>plan match-type</i>]]</pre> <p>Example: Router(config)# voice translation-rule 5 Router(cfg-translation-rule)# rule 1 /201/ /102/ Router(cfg-translation-rule)# end Router# test voice translation-rule 5 2015550101 Matched with rule 5 Original number:2015550101 Translated number:1025550101 Original number type: none Translated number type: none Original number plan: none Translated number plan: none </p>	Use this command to test your translation profiles. See the test voice translation-rule command in Cisco IOS Voice Command Reference for more information.

Configuring Dial-Peer and Channel Hunting

Dial-peer hunting, the search through a group of dial peers for an available phone line, is disabled during Cisco Unified CM fallback by default. To enable dial-peer hunting, use the **no huntstop** command. For more information about dial-peer hunting, see [Cisco IOS Voice Configuration Library](#).

If you have a dual-line phone configuration, see the “[Configuring Dual-Line Phones](#)” section on [page 156](#). Keep incoming calls from hunting to the second channel if the first channel is busy or does not answer by using the **channel** keyword in the **huntstop** command.

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

SUMMARY STEPS

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

DETAILED STEPS

	Command or Action	Purpose
Step 1	<code>call-manager-fallback</code> Example: <code>Router(config)# call-manager-fallback</code>	Enters call-manager-fallback configuration mode.
Step 2	<code>huntstop [channel]</code> Example: <code>Router(config-cm-fallback)# huntstop channel</code>	Sets the huntstop attribute for the dial peers associated with the Cisco Unified IP Phone dial peers created during Communications Manager fallback. <ul style="list-style-type: none"> For dual-line configurations, the channel keyword keeps incoming calls from hunting to the second channel if the first channel is busy or does not answer.
Step 3	<code>exit</code> Example: <code>Router(config-cm-fallback)# exit</code>	Exits call-manager-fallback configuration mode.

Examples

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

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

Configuring Busy Timeout

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

SUMMARY STEPS

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

DETAILED STEPS

	Command or Action	Purpose
Step 1	<code>call-manager-fallback</code> Example: Router(config)# call-manager-fallback	Enters call-manager-fallback configuration mode.
Step 2	<code>timeouts busy seconds</code> Example: Router(config-cm-fallback)# timeouts busy 20	Sets the amount of time after which calls are disconnected when they are transferred to busy destinations. <ul style="list-style-type: none"> <code>seconds</code>: Number of seconds. Range is from 0 to 30. Default is 10. Note This command sets the busy timeout only for calls that are transferred to busy destinations and does not affect the timeout for calls that directly dial busy destinations.
Step 3	<code>exit</code> Example: Router(config-cm-fallback)# exit	Exits call-manager-fallback configuration mode.

Examples

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

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

Configuring the Ringing Timeout Default

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

SUMMARY STEPS

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

DETAILED STEPS

	Command or Action	Purpose
Step 1	<code>call-manager-fallback</code> Example: Router(config)# call-manager-fallback	Enters call-manager-fallback configuration mode.
Step 2	<code>timeouts ringing seconds</code> Example: Router(config-cm-fallback)# timeouts ringing 30	Sets the ringing timeout default, in seconds. The range is from 5 to 60000. There is no default value.
Step 3	<code>exit</code> Example: Router(config-cm-fallback)# exit	Exits call-manager-fallback configuration mode.

Examples

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

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

Configuring Outgoing Calls

Outgoing call configuration can include the following tasks:

- [Configuring Call Transfer](#)
 - [Configuring Local and Remote Call Transfer, page 207](#) (Optional)
 - [Enabling Consultative Call Transfer and Forward Using H.450.2 and H.450.3 with Cisco SRST 3.0, page 208](#) (Optional)
 - [Enabling Analog Transfer Using Hookflash and the H.450.2 Standard with Cisco SRST 3.0 or Earlier, page 212](#) (Optional)
- [Configuring Trunk Access Codes, page 216](#) (Required Under Certain Conditions)
- [Configuring Interdigit Timeout Values, page 217](#) (Optional)
- [Configuring Class of Restriction, page 218](#) (Optional)
- [Call Blocking \(Toll Bar\) Based on Time of Day and Day of Week or Date, page 222](#) (Optional)

Configuring Local and Remote Call Transfer

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

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

SUMMARY STEPS

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

DETAILED STEPS

	Command or Action	Purpose
Step 1	call-manager-fallback Example: Router(config)# call-manager-fallback	Enters call-manager-fallback configuration mode.
Step 2	transfer-pattern <i>transfer-pattern</i> Example: Router(config-cm-fallback)# transfer-pattern 52540..	Enables the transfer of a call from a non-IP phone number to another Cisco Unified IP Phone on the same IP network using the specified transfer pattern. <ul style="list-style-type: none"> • <i>transfer-pattern</i>: String of digits for permitted call transfers. Wildcards are permitted.
Step 3	exit Example: Router(config-cm-fallback)# exit	Exits call-manager-fallback configuration mode.

Examples

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

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

Enabling Consultative Call Transfer and Forward Using H.450.2 and H.450.3 with Cisco SRST 3.0

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

**Note**

For Cisco SRST 3.1 and later versions and Cisco Unified SRST 4.0 and later versions, call transfer and call forward using H.450.2 is supported automatically with the default session application.

Prerequisites

- Call transfer with consultation is available only when a second line or call instance is supported by the IP phone. Please see the **dual-line** keyword in the **max-dn** command.
- All voice gateway routers in the VoIP network must support the H.450 standard.
- All voice gateway routers in the VoIP network must be running the following software:
 - Cisco IOS Release 12.3(2)T or a later release
 - Cisco SRST 3.0

Restrictions

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

SUMMARY STEPS

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

DETAILED STEPS

	Command or Action	Purpose
Step 1	<code>call-manager-fallback</code> Example: Router(config)# call-manager-fallback	Enters call-manager-fallback configuration mode.
Step 2	<code>call-forward pattern pattern</code> Example: Router(config-cm-fallback)# call-forward pattern 4...	Specifies the H.450.3 standard for call forwarding. <ul style="list-style-type: none"> <i>pattern</i>: Digits to match for call forwarding using the H.450.3 standard. If an incoming calling-party number matches the pattern, it can be forwarded using the H.450.3 standard. A pattern of .T forwards all calling parties using the H.450.3 standard.
Step 3	<code>transfer-system {blind full-blind full-consult local-consult}</code> Example: Router(config-cm-fallback)# transfer-system full-consult	Not supported if the transfer-to destination is on the Cisco ATA, Cisco VG224, or an SCCP-controlled FXS port. Defines the call-transfer method for all lines served by the Cisco Unified SRST router. <ul style="list-style-type: none"> blind: Calls are transferred without consultation with a single phone line using the Cisco proprietary method. <p> Note We do not recommend the blind keyword. Use either the full-blind or full-consult keyword instead.</p> <ul style="list-style-type: none"> full-blind: Calls are transferred without consultation using H.450.2 standard methods. full-consult: Calls are transferred with consultation using a second phone line if available. The calls fall back to full-blind if the second line is unavailable. local-consult: Calls are transferred with local consultation using a second phone line if available. The calls fall back to blind for nonlocal consultation or nonlocal transfer target.
Step 4	<code>transfer-pattern transfer-pattern</code> Example: Router(config-cm-fallback)# transfer-pattern 52540..	Allows transfer of telephone calls by Cisco Unified IP Phones to specified phone number patterns. <ul style="list-style-type: none"> <i>transfer-pattern</i>: String of digits for permitted call transfers. Wildcards are allowed.
Step 5	<code>exit</code> Example: Router(config-cm-fallback)# exit	Exits call-manager-fallback configuration mode. Timesaver Before exiting call-manager-fallback configuration mode, configure any other parameters that you need to set for the entire Cisco Unified SRST phone network.

	Command or Action	Purpose
Step 6	<code>voice service voip</code> Example: Router(config)# voice service voip	(Optional) Enters voice service configuration mode.
Step 7	<code>h323</code> Example: Router(conf-voi-serv)# h323	(Optional) Enters H.323 voice service configuration mode.
Step 8	<code>h450 h450-2 timeout {T1 T2 T3 T4}</code> <i>milliseconds</i> Example: Router(conf-serv-h323)# h450 h450-2 timeout T1 750	(Optional) Sets timeouts for supplementary service timers, in milliseconds. This command is used primarily when the default settings for these timers do not match your network delay parameters. See the ITU-T H.450.2 specification for more information on these timers. <ul style="list-style-type: none"> • T1: Timeout value to wait to identify a response. Default is 2000. • T2: Timeout value to wait for call setup. Default is 5000. • T3: Timeout value to wait to initiate a response. Default is 5000. • T4: Timeout value to wait for setup of a response. Default is 5000. • <i>milliseconds:</i> Number of milliseconds. Range is from 500 to 60000.
Step 9	<code>end</code> Example: Router(conf-serv-h323)# end	(Optional) Returns to privileged EXEC mode.

Examples

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

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

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

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

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

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

```

!
dial-peer voice 4000 voip
 destination-pattern 4
 session-target ipv4:10.1.1.1
!
call-manager-fallback
 call-forward pattern 4

```

Enabling Analog Transfer Using Hookflash and the H.450.2 Standard with Cisco SRST 3.0 or Earlier

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

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

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

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



Note

For Cisco SRST 3.1 and later versions and Cisco Unified SRST 4.0 and later versions, call transfer using H.450.2 is supported automatically with the default session application.

Prerequisites

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

- Tcl IVR 2.0
- H.450 Tcl script (app-h450-transfer.tcl)

**Note**

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

Restrictions

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

SUMMARY STEPS

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

DETAILED STEPS

	Command or Action	Purpose
Step 1	<p><code>call application voice application-name location</code></p> <p>Example: Router(config)# call application voice transfer_app flash:app-h450-transfer.tcl</p>	<p>Loads the Tcl script and specifies its application name.</p> <ul style="list-style-type: none"> <i>application-name</i>: User-defined name for the IVR application. This name does not have to match the script filename. <i>location</i>: Script directory and filename in URL format. For example, flash memory (<code>flash:filename</code>), a TFTP (<code>tftp://..filename</code>), or an HTTP server (<code>http://..filename</code>) are valid locations.
Step 2	<p><code>call application voice application-name language number language</code></p> <p>Example: Router(config)# call application voice transfer_app language 1 en</p>	<p>(Optional) Sets the language for dynamic prompts used by the application.</p> <ul style="list-style-type: none"> <i>application-name</i>: IVR application name that was assigned in Step 1. <i>number</i>: Number that identifies the language used by the audio files for the IVR application. <i>language</i>: Two-character code that specifies the language of the prompts. Valid entries are en (English:default), sp (Spanish), ch (Chinese), or aa (all).
Step 3	<p><code>call application voice application-name set-location language category location</code></p> <p>Example: Router(config)# call application voice transfer_app set-location en 0 flash:/prompts</p>	<p>Defines the location and category of the audio files that are used by the application for dynamic prompts.</p> <ul style="list-style-type: none"> <i>application-name</i>: Name of the Tcl IVR application. <i>language</i>: Two-character code to specify the language of the prompts. Valid entries are en (English: default), sp (Spanish), ch (Chinese), or aa (all). <i>category</i>: Category group (0 to 4) for the audio files from this location. The value 0 means all categories. <i>location</i>: URL of the directory that contains the language audio files used by the application, without filenames. Flash memory (<code>flash</code>) or a directory on a server (TFTP, HTTP, or RTSP) are all valid. <p>Prompts are required for call transfer from analog FXS phones. No prompts are needed for call transfer from IP phones.</p>

	Command or Action	Purpose
Step 4	<pre>call application voice application-name delay-time seconds</pre> <p>Example: Router(config)# call application voice transfer_app delay-time 1</p>	<p>(Optional) Sets the delay time for consultation call setup for an analog phone that is making a call transfer using the H.450 application. This command passes a value to the Tcl script by using an attribute-value (AV) pair.</p> <ul style="list-style-type: none"> <i>seconds</i>: Number of seconds to delay call setup. Range is from 1 to 10. Default is 2. <p>A delay of more than 2 seconds is generally noticeable to users.</p> <p>For more information about AV pairs and the Tcl script for H.450 call transfer and forwarding, see the ReadMe file that accompanies the script.</p>
Step 5	<pre>dial-peer voice number pots</pre> <p>Example: Router(config)# dial-peer voice 25 pots</p>	Enters dial-peer configuration mode to configure a POTS dial peer.
Step 6	<pre>application application-name</pre> <p>Example: Router(config-dial-peer)# application transfer_app</p>	Loads the application named in Step 1 onto the dial peer.
Step 7	<pre>exit</pre> <p>Example: Router(config-dial-peer)# exit</p>	<p>Exits dial-peer configuration mode.</p> <p>Timesaver Before exiting dial-peer configuration mode, configure any other dial-peer parameters that you need to set for this dial peer.</p>
Step 8	<pre>dial-peer voice number voip</pre> <p>Example: Router(config)# dial-peer voice 29 voip</p>	Enters dial-peer configuration mode to configure a VoIP dial peer.
Step 9	<pre>application application-name</pre> <p>Example: Router(config-dial-peer)# application transfer_app</p>	Loads the application named in Step 1 onto the dial peer.
Step 10	<pre>exit</pre> <p>Example: Router(config-dial-peer)# exit</p>	<p>Exits dial-peer configuration mode.</p> <p>Timesaver Before exiting dial-peer configuration mode, configure any other dial-peer parameters that you need to set for this dial peer.</p>

Examples

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

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

Configuring Trunk Access Codes



Note

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

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

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

SUMMARY STEPS

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

DETAILED STEPS

	Command or Action	Purpose
Step 1	<code>call-manager-fallback</code> Example: Router(config)# call-manager-fallback	Enters call-manager-fallback configuration mode.
Step 2	<code>access-code {{fxo e&m} dial-string {bri pri} dial-string [direct-inward-dial]}</code> Example: Router(config-cm-fallback)# access-code e&m 8	Configures trunk access codes for each type of line so that the Cisco Unified IP Phones can access the trunk lines only in Cisco Unified Communications Manager fallback mode when the Cisco Unified SRST is enabled. <ul style="list-style-type: none"> • fxo: Enables a Foreign Exchange Office (FXO) interface. • e&m: Enables an analog Ear and Mouth (E&M) interface. • dial-string: String of characters that sets up dial access codes for each specified line type by creating dial peers. The <i>dial-string</i> argument is used to set up temporary dial peers for each specified line type. • bri: Enables a BRI interface. • pri: Enables a PRI interface. • direct-inward-dial: (Optional) Enables Direct Inward Dialing (DID) on the POTS dial peer.
Step 3	<code>exit</code> Example: Router(config-cm-fallback)# exit	Exits call-manager-fallback configuration mode.

Examples

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

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

Configuring Interdigit Timeout Values

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

SUMMARY STEPS

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

DETAILED STEPS

	Command or Action	Purpose
Step 1	call-manager-fallback Example: Router(config)# call-manager-fallback	Enters call-manager-fallback configuration mode.
Step 2	timeouts interdigit <i>seconds</i> Example: Router(config-cm-fallback)# timeouts interdigit 5	(Optional) Configures the interdigit timeout value for all Cisco IP phones that are attached to the router. <ul style="list-style-type: none"> • <i>seconds</i>: Interdigit timeout duration, in seconds, for all Cisco Unified IP Phones. Valid entries are integers from 2 to 120.
Step 3	exit Example: Router(config-cm-fallback)# exit	Exits call-manager-fallback configuration mode.

Examples

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

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

Configuring Class of Restriction

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

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

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

a Cisco Unified SRST router and try to make a call from that phone, the call will be considered an incoming call to the router and voice port. If you make a call to the FXS phone, the call will be considered outgoing.

By default, an incoming call leg has the highest COR priority; the outgoing call leg has the lowest priority. If there is no COR configuration for incoming calls on a dial peer, you can make a call from a phone attached to the dial peer, so that the call will go out of any dial peer regardless of the COR configuration on that dial peer. Table 8-2 describes call functionality based on how your COR lists are configured.

Table 8-2 Combinations of COR List and Results

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

SUMMARY STEPS

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

DETAILED STEPS

	Command or Action	Purpose
Step 1	<code>call-manager-fallback</code> Example: Router(config)# call-manager-fallback	Enters call-manager-fallback configuration mode.
Step 2	<code>cor {incoming outgoing} cor-list-name</code> [<i>cor-list-number starting-number - ending-number</i> default] Example: Router(config-cm-fallback)# cor outgoing LockforPhoneC 1 5010 - 5020	Configures a COR on dial peers associated with directory numbers. <ul style="list-style-type: none"> 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 20, comprised of incoming or outgoing dial peers. The first six COR lists are applied to a range of directory numbers. The directory numbers that do not have a COR configuration are assigned to the default COR list. <i>starting-number - ending-number:</i> Directory number range; for example, 2000–2025. default: Instructs the router to use an existing default COR list.
Step 3	<code>exit</code> Example: Router(config-cm-fallback)# exit	Exits call-manager-fallback configuration mode.

Examples

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

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

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

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

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

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

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

```
dial-peer cor list call911
  member 911

dial-peer cor list call1800
  member 1800

dial-peer cor list call1900
  member 1900

dial-peer cor list calllocal
  member local_call

dial-peer cor list engineering
  member 911
  member local_call

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

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

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

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

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

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

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

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

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

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

The sample configuration allows for the following:

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

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

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

When a user attempts to place a call to digits that match a pattern that is specified for call blocking during a time period that is defined for call blocking, a fast busy signal is played for approximately 10 seconds. The call is then terminated, and the line is placed back in on-hook status.

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

SUMMARY STEPS

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

DETAILED STEPS

	Command or Action	Purpose
Step 1	call-manager-fallback Example: Router(config)# call-manager-fallback	Enters call-manager-fallback configuration mode.
Step 2	after-hours block pattern tag pattern [7-24] Example: Router(config-cm-fallback)# after-hours block pattern 1 91900	Defines a pattern of outgoing digits to be blocked. Up to 32 patterns can be defined, using individual commands. <ul style="list-style-type: none"> • If the 7-24 keyword is specified, the pattern is always blocked, 7 days a week, 24 hours a day. • If the 7-24 keyword is not specified, the pattern is blocked during the days and dates that are defined using the after-hours day and after-hours date commands.
Step 3	after-hours day day start-time stop-time Example: Router(config-cm-fallback)# after-hours day mon 19:00 7:00	Defines a recurring time period based on the day of the week during which calls are blocked to outgoing dial patterns that are defined using the after-hours block pattern command. <ul style="list-style-type: none"> • <i>day</i>: Day of the week abbreviation. The following are valid day abbreviations: sun, mon, tue, wed, thu, fri, sat. • <i>start-time stop-time</i>: Beginning and ending times for call blocking, in an HH:MM format using a 24-hour clock. If the stop time is a smaller value than the start time, the stop time occurs on the day following the start time. For example, “mon 19:00 07:00” means “from Monday at 7 p.m. until Tuesday at 7 a.m.”
Step 4	after-hours date month date start-time stop-time Example: Router(config-cm-fallback)# after-hours date jan 1 0:00 0:00	Defines a recurring time period based on month and date during which calls are blocked to outgoing dial patterns that are defined using the after-hours block pattern command. <ul style="list-style-type: none"> • <i>month</i>: Month abbreviation. The following are valid month abbreviations: jan, feb, mar, apr, may, jun, jul, aug, sep, oct, nov, dec. • <i>date</i>: Date of the month. Range is from 1 to 31. • <i>start-time stop-time</i>: Beginning and ending times for call blocking, in an HH:MM format using a 24-hour clock. The stop time must be larger than the start time. The value 24:00 is not valid. If 00:00 is entered as an stop time, it is changed to 23:59. If 00:00 is entered for both start time and stop time, calls are blocked for the entire 24-hour period on the specified date.
Step 5	exit Example: Router(config-cm-fallback)# exit	Exits call-manager-fallback configuration mode.

Examples

The following example defines several patterns of digits for which outgoing calls are blocked. Patterns 1 and 2, which block calls to external numbers that begin with “1” and “011,” are blocked on Monday through Friday before 7 a.m. and after 7 p.m., on Saturday before 7 a.m. and after 1 p.m., and all day Sunday. Pattern 3 blocks calls to 900 numbers 7 days a week, 24 hours a day.

```
call-manager-fallback
after-hours block pattern 1 91
after-hours block pattern 2 9011
after-hours block pattern 3 91900 7-24
after-hours block day mon 19:00 07:00
after-hours block day tue 19:00 07:00
after-hours block day wed 19:00 07:00
after-hours block day thu 19:00 07:00
after-hours block day fri 19:00 07:00
after-hours block day sat 13:00 12:00
after-hours block day sun 12:00 07:00
```

How to Configure Cisco Unified SIP SRST

This section contains the following procedures:

- [Configuring SIP Phone Features, page 224](#) (optional)
- [Configuring SIP-to-SIP Call Forwarding, page 226](#) (required)
- [Configuring Call Blocking Based on Time of Day, Day of Week, or Date, page 228](#) (required)
- [SIP Call Hold and Resume, page 232](#) (no configuration necessary)
- [Examples, page 232](#)

Configuring SIP Phone Features

After a voice register pool has been set, this procedure adds optional features to increase functionality. Some features can be made per pool or globally.

In **voice register pool** configuration, you can now configure several new options per pool (a pool can be one phone or a group of phones). There is also a new **voice register global** configuration mode for Cisco Unified SIP SRST. In **voice register global** mode, you can globally assign characteristics to phones.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register global** *tag*
4. **max-pool** *max-voice-register-pools*
5. **application** *application-name*
6. **external ring** {**bellcore-dr1** | **bellcore-dr2** | **bellcore-dr3** | **bellcore-dr4** | **bellcore-dr5**}
7. **exit**
8. **voice register pool** *tag*
9. **no vad**

10. `codec codec-type [bytes]`
11. `end`

DETAILED STEPS

	Command or Action	Purpose
Step 1	<pre>enable</pre> <p>Example: Router> enable</p>	<p>Enables privileged EXEC mode.</p> <ul style="list-style-type: none"> Enter your password if prompted.
Step 2	<pre>configure terminal</pre> <p>Example: Router# configure terminal</p>	<p>Enters global configuration mode.</p>
Step 3	<pre>voice register global tag</pre> <p>Example: Router(config)# voice register global 12</p>	<p>Enters voice register global configuration mode to set global parameters for all supported Cisco SIP IP phones in a Cisco Unified SIP SRST environment.</p>
Step 4	<pre>max-pool max-voice-register-pools</pre> <p>Example: Router(config-register-global)# max-pool 10</p>	<p>Sets the maximum number of SIP voice register pools that are supported in a Cisco Unified SIP SRST environment. The <i>max-voice-register-pools</i> argument represents the maximum number of SIP voice register pools supported by the Cisco Unified SIP SRST router. The upper limit of voice register pools is version- and platform-dependent; see Cisco IOS command-line interface (CLI) help. Default is 0.</p>
Step 5	<pre>application application-name</pre> <p>Example: Router(config-register-global)# application global_app</p>	<p>Selects the session-level application for all dial peers associated with SIP phones. Use the <i>application-name</i> argument to define a specific interactive voice response (IVR) application.</p>
Step 6	<pre>external-ring {bellcore-dr1 bellcore-dr2 bellcore-dr3 bellcore-dr4 bellcore-dr5}</pre> <p>Example: Router(config-register-global)# external-ring bellcore-dr1</p>	<p>Specifies the type of ring sound used on Cisco SIP or Cisco SCCP IP phones for external calls. Each bellcore-dr 1-5 keyword supports standard distinctive ringing patterns as defined in the standard GR-506-CORE, <i>LSSGR: Signaling for Analog Interfaces</i>.</p>
Step 7	<pre>exit</pre> <p>Example: Router(config-register-global)# exit</p>	<p>Exits voice register global configuration mode.</p>
Step 8	<pre>voice register pool tag</pre> <p>Example: Router(config)# voice register pool 20</p>	<p>Enters voice register pool configuration mode for SIP phones.</p> <ul style="list-style-type: none"> Use this command to control which phone registrations are to be accepted or rejected by a Cisco Unified SIP SRST device.

	Command or Action	Purpose
Step 9	<pre>no vad</pre> <p>Example: Router(config-register-pool)# no vad</p>	<p>Disables voice activity detection (VAD) on the VoIP dial peer.</p> <ul style="list-style-type: none"> VAD is enabled by default. Because there is no comfort noise during periods of silence, the call may seem to be disconnected. You may prefer to set no vad on the SIP phone pool.
Step 10	<pre>codec codec-type [bytes]</pre> <p>Example: Router(config-register-pool)# codec g729r8</p>	<p>Specifies the codec supported by a single SIP phone or a VoIP dial peer in a Cisco Unified SIP SRST environment. The <i>codec-type</i> argument specifies the preferred codec and can be one of the following:</p> <ul style="list-style-type: none"> g711alaw: G.711 a-law 64,000 bps. g711ulaw: G.711 mu-law 64,000 bps. g729r8: G.729 8000 bps (default). <p>The <i>bytes</i> argument is optional and specifies the number of bytes in the voice payload of each frame</p>
Step 11	<pre>end</pre> <p>Example: Router(config-register-pool)# end</p>	<p>Returns to privileged EXEC mode.</p>

Configuring SIP-to-SIP Call Forwarding

SIP-to-SIP call forwarding (call routing) is available. Call forwarding is provided either by the phone or by using a back-to-back user agent (B2BUA), which allows call forwarding on any dial peer. Calls into a SIP device may be forwarded to other SIP or SCCP devices (including Cisco Unity, third-party voice-mail systems, or an auto attendant or IVR system such as IPCC and IPCC Express). In addition, SCCP IP phones may be forwarded to SIP phones.

Cisco Unity or other voice messaging systems connected by a SIP trunk or SIP user agent are able to pass a message-waiting indicator (MWI) when a message is left. The SIP phone then displays the MWI when indicated by the voice messaging system.



Note

SIP-to-H.323 call forwarding is not supported.

To configure SIP-to-SIP call forwarding, you must first allow connections between specific types of endpoints in a Cisco IP-to-IP gateway. The **allow-connections** command grants this capability. Once the SIP-to-SIP connections are allowed, you can configure call forwarding under an individual SIP phone pool. Any of the following commands can be used to configure call forwarding, according to your needs:

- Under **voice register pool**
 - call-forward b2bua all** *directory-number*
 - call-forward b2bua busy** *directory-number*
 - call-forward b2bua mailbox** *directory-number*
 - call-forward b2bua noan** *directory-number* [**timeout** *seconds*]

In a typical Cisco Unified SIP SRST setup, the **call-forward b2bua mailbox** command is not used; however, it is likely to be used in a Cisco Unified SIP Communications Manager Express (CME) environment. Detailed procedures for configuring the **call-forward b2bua mailbox** command are found in the *Cisco Unified Communications Manager (CallManager)* documentation on Cisco.com.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register pool tag**
4. **call-forward b2bua all directory-number**
5. **call-forward b2bua busy directory-number**
6. **call-forward b2bua mailbox directory-number**
7. **call-forward b2bua noan directory-number timeout seconds**
8. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. <ul style="list-style-type: none"> • Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	voice register pool tag Example: Router(config)# voice register pool 15	Enters voice register pool configuration mode. <ul style="list-style-type: none"> • Use this command to control which phone registrations are accepted or rejected by a Cisco Unified SIP SRST device.
Step 4	call-forward b2bua all directory-number Example: Router(config-register-pool)# call-forward b2bua all 5005	Enables call forwarding for a SIP back-to-back user agent (B2BUA) so that all incoming calls are forwarded to another non-SIP station extension (that is, SIP trunk, H.323 trunk, SCCP device or analog/digital trunk). <ul style="list-style-type: none"> • <i>directory-number</i>: Telephone number to which calls are forwarded. Represents a fully qualified E.164 number. Maximum length of the telephone number is 32.
Step 5	call-forward b2bua busy directory-number Example: Router(config-register-pool)# call-forward b2bua busy 5006	Enables call forwarding for a SIP B2BUA so that incoming calls to a busy extension are forwarded to another extension. <ul style="list-style-type: none"> • <i>directory-number</i>: Telephone number to which calls are forwarded. Represents a fully qualified E.164 number. Maximum length of the telephone number is 32.

	Command or Action	Purpose
Step 6	<pre>call-forward b2bua mailbox directory-number</pre> <p>Example: <pre>Router(config-register-pool)# call-forward b2bua mailbox 5007</pre></p>	<p>Controls the specific voice-mail box selected in a voice-mail system at the end of a call forwarding exchange.</p> <ul style="list-style-type: none"> <i>directory-number</i>: Telephone number to which calls are forwarded when the forwarded destination is busy or does not answer. Represents a fully qualified E.164 number. Maximum length of the telephone number is 32.
Step 7	<pre>call-forward b2bua noan directory-number timeout seconds</pre> <p>Example: <pre>Router(config-register-pool)# call-forward b2bua noan 5010 timeout 10</pre></p>	<p>Enables call forwarding for a SIP B2BUA so that incoming calls to an extension that does not answer after a configured amount of time are forwarded to another extension.</p> <p>This command is used if a phone is registered with a Cisco Unified SIP SRST router, but the phone is not reachable because there is no IP connectivity (there is no response to Invite requests).</p> <ul style="list-style-type: none"> <i>directory-number</i>: Telephone number to which calls are forwarded. Represents a fully qualified E.164 number. Maximum length of the telephone number is 32. timeout seconds: Duration, in seconds, that a call can ring with no answer before the call is forwarded to another extension. Range is 3 to 60000. The default value is 20.
Step 8	<pre>end</pre> <p>Example: <pre>Router(config-register-pool)# end</pre></p>	<p>Returns to privileged EXEC mode.</p>

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

This section applies to both SCCP and SIP SRST. Call blocking prevents the unauthorized use of phones and is implemented by matching a pattern of up to 32 digits during a specified time of day, day of week, or date. Cisco Unified SIP SRST provides SIP endpoints the same time-based call blocking mechanism that is currently provided for SCCP phones. The call blocking feature supports all incoming calls, including incoming SIP and analog FXS calls.



Note

Pin-based exemptions and the “Login” toll-bar override are not supported in Cisco Unified SIP SRST.

The commands used for SIP phone call blocking are the same commands that are used for SCCP phones on your Cisco Unified SRST system. The Cisco SRST session application accesses the current after-hours configuration under call-manager-fallback mode and applies it to calls originated by Cisco SIP phones that are registered to the Cisco SRST router. The commands used in call-manager-fallback mode that set block criteria (time/date/block pattern) are the following:

- **after-hours block pattern** *pattern-tag pattern* [7-24]
- **after-hours day** *day start-time stop-time*
- **after-hours date** *month date start-time stop-time*

When a user attempts to place a call to digits that match a pattern that has been specified for call blocking during a time period that has been defined for call blocking, the call is immediately terminated and the caller hears a fast busy.

In SRST (call-manager-fallback configuration mode), there is no phone- or pin-based exemption to after-hours call blocking. However, in Cisco Unified SIP SRST (voice register pool mode), individual IP phones can be exempted from all call blocking using the **after-hours exempt** command.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **call-manager-fallback**
4. **after-hours block pattern** *tag pattern* [7-24]
5. **after-hours day** *day start-time stop-time*
6. **after-hours date** *month date start-time stop-time*
7. **exit**
8. **voice register pool** *tag*
9. **after-hour exempt**
10. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Router> enable	Enables privileged EXEC mode. <ul style="list-style-type: none"> • Enter your password if prompted.
Step 2	configure terminal Example: Router# configure terminal	Enters global configuration mode.
Step 3	call-manager-fallback Example: Router(config)# call-manager-fallback	Enters call-manager-fallback configuration mode.
Step 4	after-hours block pattern <i>tag pattern</i> [7-24] Example: Router(config-cm-fallback)# after-hours block pattern 1 91900	Defines a pattern of outgoing digits to be blocked. Up to 32 patterns can be defined, using individual commands. <ul style="list-style-type: none"> • If the 7-24 keyword is specified, the pattern is always blocked, 7 days a week, 24 hours a day. • If the 7-24 keyword is not specified, the pattern is blocked during the days and dates that are defined using the after-hours day and after-hours date commands.

Command or Action	Purpose
<p>Step 5 <code>after-hours day day start-time stop-time</code></p> <p>Example: Router(config-cm-fallback)# after-hours day mon 19:00 07:00</p>	<p>Defines a recurring time period based on the day of the week during which calls are blocked to outgoing dial patterns that are defined using the after-hours block pattern command.</p> <ul style="list-style-type: none"> <i>day</i>: Day of the week abbreviation. The following are valid day abbreviations: sun, mon, tue, wed, thu, fri, sat. <i>start-time stop-time</i>: Beginning and ending times for call blocking, in an HH:MM format using a 24-hour clock. If the stop time is a smaller value than the start time, the stop time occurs on the day following the start time. For example, “mon 19:00 07:00” means “from Monday at 7 p.m. until Tuesday at 7 a.m.” <p>The value 24:00 is not valid. If 00:00 is entered as a stop time, it is changed to 23:59. If 00:00 is entered for both start time and stop time, calls are blocked for the entire 24-hour period on the specified date.</p>
<p>Step 6 <code>after-hours date month date start-time stop-time</code></p> <p>Example: Router(config-cm-fallback)# after-hours date jan 1 00:00 00:00</p>	<p>Defines a recurring time period based on month and date during which calls are blocked to outgoing dial patterns that are defined using the after-hours block pattern command.</p> <ul style="list-style-type: none"> <i>month</i>: Month abbreviation. The following are valid month abbreviations: jan, feb, mar, apr, may, jun, jul, aug, sep, oct, nov, dec. <i>date</i>: Date of the month. Range is from 1 to 31. <i>start-time stop-time</i>: Beginning and ending times for call blocking, in an HH:MM format using a 24-hour clock. The stop time must be larger than the start time. <p>The value 24:00 is not valid. If 00:00 is entered as a stop time, it is changed to 23:59. If 00:00 is entered for both start time and stop time, calls are blocked for the entire 24-hour period on the specified date.</p>
<p>Step 7 <code>exit</code></p> <p>Example: Router(config-cm-fallback)# exit</p>	<p>Exits call-manager-fallback configuration mode.</p>
<p>Step 8 <code>voice register pool tag</code></p> <p>Example: Router(config)# voice register pool 12</p>	<p>Enters voice register pool configuration mode.</p> <ul style="list-style-type: none"> Use this command to control which registrations are accepted or rejected by a Cisco Unified SIP SRST device.

	Command or Action	Purpose
Step 9	after-hour exempt Example: Router(config-register-pool)# after-hour exempt	Specifies that for a particular voice register pool, none of its outgoing calls are blocked even though call blocking is enabled.
Step 10	end Example: Router(config-register-pool)# end	Returns to privileged EXEC mode.

Examples

The following example defines several patterns of digits for which outgoing calls are blocked. Patterns 1 and 2, which block calls to external numbers that begin with 1 and 011, are blocked on Monday through Friday before 7 a.m. and after 7 p.m. Pattern 3 blocks calls to 900 numbers 7 days a week, 24 hours a day.

```
call-manager-fallback
after-hours block pattern 1 91
after-hours block pattern 2 9011
after-hours block pattern 3 91900 7-24
after-hours day mon 19:00 07:00
after-hours day tue 19:00 07:00
after-hours day wed 19:00 07:00
after-hours day thu 19:00 07:00
after-hours day fri 19:00 07:00
```

The following example exempts a Cisco SIP phone pool from the configured blocking criteria:

```
voice register pool 1
after-hour exempt
```

Verification

To verify the feature's configuration, enter one of the following commands:

- **show voice register dial-peer:** Displays all the dial peers created dynamically by phones that have registered. This command also displays configurations for after hours blocking and call forwarding.
- **show voice register pool <tag>:** Displays information regarding a specific pool.
- **debug ccsip message:** Debugs basic B2BUA calls.

For more information about these commands, see [Cisco Unified SRST and Cisco Unified SIP SRST Command Reference \(All Versions\)](#).

SIP Call Hold and Resume

Cisco Unified SRST supports the ability for SIP phones to place calls on hold and to resume from calls placed on hold. This also includes support for a consultative hold where A calls B, B places A on hold, B calls C, and B disconnects from C and then resumes with A. Support for call hold is signaled by SIP phones using “re-INVITE c=0.0.0.0” and also by the receive-only mechanism.

No configuration is necessary.

Examples

```
Router# show running-config
Building configuration...

Current configuration : 1462 bytes
configuration mode exclusive manual
version 12.4
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
service internal
!
boot-start-marker
boot-end-marker
!
logging buffered 8000000 debugging
!
no aaa new-model
!
resource policy
!
clock timezone edt -5
clock summer-time edt recurring
ip subnet-zero
!
!
!
ip cef
!
!
!
voice-card 0
no dspfarm
!
!
voice service voip
  allow-connections h323 to h323
  allow-connections h323 to sip
  allow-connections sip to h323
  allow-connections sip to sip
sip
  registrar server expires max 600 min 60
!
!
!
voice register global
  max-dn 10
  max-pool 10
!
! Define call forwarding under a voice register pool
voice register pool 1
```

```
id mac 0012.7F57.60AA
number 1 1000
call-forward b2bua busy 2413
call-forward b2bua noan 2414 timeout 30

codec g711ulaw
!
voice register pool 2
id mac 0012.7F3B.9025
number 1 2800
codec g711ulaw
!
voice register pool 3
id mac 0012.7F57.628F
number 1 2801
codec g711ulaw
!
!
!
interface GigabitEthernet0/0
ip address 10.0.2.99 255.255.255.0
duplex auto
speed auto
!
interface GigabitEthernet0/1
no ip address
shutdown
duplex auto
speed auto
!
ip classless
ip route 0.0.0.0 0.0.0.0 GigabitEthernet0/0
!
ip http server
!
!
!
control-plane
!
!
!
dial-peer voice 1000 voip
destination-pattern 24..
session protocol sipv2
session target ipv4:10.0.2.5
codec g711ulaw
!
! Define call blocking under call-manager-fallback mode
call-manager-fallback
max-conferences 4 gain -6
after-hours block pattern 1 2417

after-hours date Dec 25 12:01 20:00
!
!
line con 0
exec-timeout 0 0
line aux 0
line vty 0 4
login
!
scheduler allocate 20000 1000
```

```
ntp server 10.0.2.10
!  
end
```

How to Configure Optional Features

This section describe the following optional additional call features:

- Three-party G.711 ad hoc conferencing—Cisco Unified Survivable Remote Site Telephony (SRST) support for simultaneous three-party conferences.
- eXtensible Markup Language (XML) application program interface (API)— This interface supplies data from Cisco Unified SRST to management software.

The following sections describe how to configure these optional features:

- [Enabling Three-Party G.711 Ad Hoc Conferencing, page 234](#)
- [Defining XML API Schema, page 236](#)

Enabling Three-Party G.711 Ad Hoc Conferencing

Enabling three-party G.711 ad hoc conferencing involves configuring the maximum number of simultaneous three-party conferences supported by the Cisco Unified SRST router. For conferencing to be available, an IP phone must have a minimum of two lines connected to one or more buttons. See the [“Configuring a Secondary Dial Tone” section on page 155](#).

SUMMARY STEPS

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

DETAILED STEPS

	Command or Action	Purpose
Step 1	<code>call-manager-fallback</code> Example: Router(config)# <code>call-manager-fallback</code>	Enters call-manager-fallback configuration mode.
Step 2	<code>max-conferences</code> <i>max-conference-numbers</i> Example: Router(config-cm-fallback)# <code>max-conferences 16</code>	Sets the maximum number of simultaneous three-party conferences supported by the router. The maximum number possible is platform dependent: <ul style="list-style-type: none"> • Cisco 1751 router:8 • Cisco 1760 router:8 • Cisco 2600 series routers:8 • Cisco 2600-XM series routers:8 • Cisco 2801 router:8 • Cisco 2811, Cisco 2821, and Cisco 2851 routers:16 • Cisco 3640 and Cisco 3640A routers:8 • Cisco 3660 router: 16 • Cisco 3725 router: 16 • Cisco 3745 router: 16 • Cisco 3800 series router: 24
Step 3	<code>exit</code> Example: Router(config-cm-fallback)# <code>exit</code>	Exits call-manager-fallback configuration mode.

Examples

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

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

Defining XML API Schema

The Cisco IOS commands in this section allow you to specify parameters associated with the XML API. For more information, see [XML Provisioning Guide for Cisco CME/SRST](#). See the “[Enabling Consultative Call Transfer and Forward Using H.450.2 and H.450.3 with Cisco SRST 3.0](#)” section on [page 208](#) for configuration instructions.

SUMMARY STEPS

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

DETAILED STEPS

	Command or Action	Purpose
Step 1	call-manager-fallback	Enters call-manager-fallback configuration mode.
	Example: Router(config)# call-manager-fallback	
Step 2	xmlschema <i>schema-url</i>	Specifies the URL for an XML API schema to be used with this Cisco Unified SRST system.
	Example: Router(config-cm-fallback)# xmlschema http://server2.example.com/ schema/schema1.xsd	<ul style="list-style-type: none"> • <i>schema-url</i>: Local or remote URL as defined in RFC 2396.
Step 3	exit	Exits call-manager-fallback configuration mode.
	Example: Router(config-cm-fallback)# exit	

Configuration Examples for Call Handling

- [Example: Monitoring the Status of Key Expansion Modules, page 236](#)
- [Example: Configuring Voice Hunt Groups in Cisco Unified SIP SRST, page 237](#)

Example: Monitoring the Status of Key Expansion Modules

Show commands are used to monitor the status and other details of Key Expansion Modules (KEMs).

The following example demonstrates how the **show voice register all** command displays KEM details with all the Cisco Unified CME configurations and registration information:

```
show voice register all
VOICE REGISTER GLOBAL
=====
CONFIG [Version=9.1]
```

```

=====
.....
Pool Tag 5
Config:
  Mac address is B4A4.E328.4698
  Type is 9971 addon 1 CKEM
  Number list 1 : DN 2
  Number list 2 : DN 3
  Proxy Ip address is 0.0.0.0
  DTMF Relay is disabled
  Call Waiting is enabled
  DnD is disabled
  Video is enabled
  Camera is enabled
  Busy trigger per button value is 0
  keep-conference is enabled
  registration expires timer max is 200 and min is 60
  kpml signal is enabled
  Lpcor Type is none

```

The following example demonstrates how the **show voice register pool type** command displays all the phones configured with add-on KEMs in Cisco Unified CME:

```

Router# show voice register pool type CKEM
Pool ID          IP Address      Ln DN  Number          State
=====
4      B4A4.E328.4698  9.45.31.111    1 4    5589$          REGISTERED

```

Example: Configuring Voice Hunt Groups in Cisco Unified SIP SRST

The following example shows how to configure longest-idle hunt group 20 with pilot number 4701, final number 5000, and 6 numbers in the list. After a call is redirected six times (makes 6 hops), it is redirected to the final number 5000.

```

Router(config)# voice hunt-group 20 longest-idle
Router(config-voice-hunt-group)# pilot 4701
Router(config-voice-hunt-group)# list 4001, 4002, 4023, 4028, 4045, 4062
Router(config-voice-hunt-group)# final 5000
Router(config-voice-hunt-group)# hops 6
Router(config-voice-hunt-group)# timeout 20
Router(config-voice-hunt-group)# exit

```

Where to Go Next

If you need to configure security, see the [“Configuring Secure SRST for SCCP and SIP”](#) section on page 239, or if you need to configure voicemail, see the [“Integrating Voicemail with Cisco Unified SRST”](#) section on page 331. If you need to configure video parameters, see the [“Setting Video Parameters”](#) section on page 355. If you do not need any of those features, go to the [“Monitoring and Maintaining Cisco Unified SRST”](#) section on page 369.

For additional information, see the [“Additional References”](#) section on page 308 in the [“Cisco Unified SRST Feature Overview”](#) section on page 1 chapter.

