Command Reference: A through M

This chapter contains commands to configure and maintain Cisco Unified Survivable Remote Site Telephony (SRST) and Cisco Unified SIP SRST. The commands are presented in alphabetical order. Some commands required for configuring Cisco Unified SRST and Cisco Unified SIP SRST may be found in other Cisco IOS command references. Use the command reference primary index or search online to find these commands.

access-code

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

FXO and EandM Line Types
access-code{**fxo** | **e&m**}*dial-string*
no access-code{**fxo** | **e&m**}*dial-string*

BRI and PRI Line Types
access-code {**bri** | **pri**} *dial-string* [**direct-inward-dial**]
no access-code {**bri** | **pri**} [*dial-string*] [**direct-inward-dial**]

Syntax Description

fxo	Enables a Foreign Exchange Office (FXO) interface.
e&m	Enables an analog ear and mouth (E&M) interface.
dial-string	Sets up dial access codes for each specified line type by creating dial peers.
bri	Enables a BRI interface.
pri	Enables a PRI interface.
direct-inward-dial	(Optional) Enables direct inward dialing on a POTS dial peer.

Command Default

No default behavior or values.

Command Modes

Call-manager-fallback configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco Unified SRST 1.0	This command was introduced on the following platforms: Cisco 2600 series and Cisco 3600 series multiservice routers, and Cisco IAD2420 series IADs.
12.2(2)XT	Cisco Unified SRST 2.0	This command was implemented on Cisco 1750 and Cisco 1751 multiservice routers.
12.2(8)T	Cisco Unified SRST 2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers.

Usage Guidelines

The **access-code** command configures trunk access codes for each type of line—BRI, E&M, FXO, and PRI—so that the Cisco IP phones can access the trunk lines during Cisco Unified Communications Manager fallback when Cisco Unified SRST is enabled. This provides systemwide access.

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

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

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

Examples

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

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# access-code bri 8 direct-inward-dial
The following example sets the access-code
 command for E&M 8:
Router(config)# call-manager-fallback
Router(config-cm-fallback)# access-code e&m 8
```

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

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

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

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

Related Commands

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

addon

To define whether a phone type supports add-on modules, use the **addon** command in ephone-type configuration mode. To reset to the default value, use the **no** form of this command.

addon
no addon

Syntax Description

This command has no arguments or keywords.

Command Default

Disabled (phone type does not support add-on modules).

Command Modes

Ephone-type configuration (config-ephone-type)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)XZ	Cisco Unified CME 4.3 Cisco Unified SRST 4.3	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0 Cisco Unified SRST 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

This command specifies that add-on modules, such as the Cisco Unified IP Phone 7915 Expansion Module, are supported by the type of phone being added with the phone-type template.

Examples

The following example shows that expansion modules are supported for the phone being added with an ephone-type template:

```
Router(config)# ephone-type 7965
Router(config-ephone-type)# addon
```

Related Commands

Command	Description
device-id	Specifies the device ID for a phone type in an ephone-type template.
device-name	Assigns a name to a phone type in an ephone-type template.

address (voice emergency response location)

To define the civic address for an ERL that is used for the ALI database upload, use the **address** command in voice emergency response location mode. To remove this definition, use the **no** form of the command. This command is optional.

address *string*
no address

Syntax Description

string	String (1-247 characters) used to identify an ERL's civic address.

Command Default

The civic address is not defined.

Command Modes

Voice emergency response location configuration (cfg-emrgncy-resp-location)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)XY	Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1) Cisco Unified SIP SRST 4.2(1)	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0 Cisco Unified SRST 7.0 Cisco Unified SIP SRST 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

This command creates a comma separated text entry of the ERL's civic address. The address information must be entered to conform with the NENA-2 Data Record specifications or the recommendations by the service provider.

Examples

In this example, a civic address is displayed for ERL 60.

```
voice emergency response location 60
subnet 1 209.165.200.224 255.255.0.0
elin 1 4085550100
name Cookies and More Incorporated,
address I,408,5550100,,11902,,,Main Street,Emerald City,CA,Idina Menzel,1,,,,,,
```

Related Commands

Command	Description
elin	Specifies a PSTN number that will replace the caller's extension.
name	Specifies a string (up to 30-characters) used internally to identify or describe the emergency response location.
subnet	Defines which IP phones are part of this ERL.

after-hour exempt (voice register pool)

To specify that a particular voice register pool does not have any of its outgoing calls blocked, even though global system call blocking is enabled, use the **after-hours exempt** command in voice register pool configuration mode. To return to the default, use the **no** form of this command.

after-hour exempt
no after-hour exempt

Syntax Description

This command has no arguments or keywords.

Command Default

Disabled (global call blocking remains active, as configured).

Command Modes

Voice register pool configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4 Cisco SRST 3.4	This command was introduced.

Usage Guidelines

This command exempts individual Cisco SIP phones and phone extensions in a voice register pool from call blocking.

Call blocking on Cisco IP phones is defined in the following way. First, define one or more patterns of outgoing digits by using the **after-hours block pattern** command in either telephony-service configuration mode for Cisco Unified CME or in call-manager-fallback configuration mode for Cisco Unified SIP SRST. Next, define one or more time periods during which calls that match those patterns are to be blocked are by using the **after-hours date** or **after-hours day** command or both. By default, all Cisco IP phones in a Cisco Unified CME or Cisco Unified SIP SRST system are restricted during the specified time if at least one pattern and at least one time period are defined.

A phone extension is exempt as long as the **after-hour exempt** command is configured in voice register dn or in voice register pool configuration mode.

Note The **id** (voice register pool) command is required before Cisco Unified CME or Cisco Unified SIP SRST can accept registrations. Configure the **id** (voice register pool) command before any other voice register pool command.

Examples

The following example shows how to configure a SIP phone, specified by the **voice register pool** command, so that outgoing calls are not blocked:

```
Router(config)# voice register pool 23
Router(config-register-pool)# after-hour exempt
```

The following example shows how to specify that outgoing calls from extension 5001 under voice register pool 2 are not blocked:

```
Router(config)# voice register pool 2
Router(config-register-pool)# number 5001
Router(config-register-pool)# after-hour exempt
```

Related Commands

Command	Description
after-hours block pattern	Defines a pattern of digits for blocking outgoing calls from IP phones.
after-hours block pattern (call-manager- fallback)	Defines a pattern of digits for blocking outgoing calls from IP phones.
after-hours date	Defines a recurring period based on date during which outgoing calls that match defined block patterns are blocked on IP phones.
after-hours date (call-manager-fallback)	Defines a recurring period based on date during which outgoing calls that match defined block patterns are blocked on IP phones.
after-hours day	Defines a recurring period based on day of the week during which outgoing calls that match defined block patterns are blocked on IP phones.
after-hours day (call-manager-fallback)	Defines a recurring period based on day of the week during which outgoing calls that match defined block patterns are blocked on IP phones.
after-hour exempt (voice register dn)	Specifies that an individual extension on a SIP phone does not have any of its outgoing calls blocked even though global system call blocking is enabled.
call-manager-fallback	Enables Cisco Unified SIP SRST support and enter call-manager-fallback configuration mode.
id (voice register pool)	Explicitly identifies a locally available individual Cisco SIP IP phone, or when running Cisco Unified SIP SRST, set of Cisco Unified SIP IP phones.
telephony-service	Enters telephony-service configuration mode to configure a Cisco Unified Communications Manager Express (Cisco Unified CME) system.
voice register dn	Enters voice register dn configuration mode to define an extension for a SIP phone line.
voice register pool	Enters voice register pool configuration mode for Cisco SIP IP phones.

after-hours block pattern (call-manager-fallback)

To define a pattern of outgoing digits for call blocking from IP phones, use the **after-hours block pattern** command in call-manager-fallback configuration mode. To delete a call-blocking pattern, use the **no** form of this command.

after-hours block pattern *pattern-tag pattern* **[7-24]**
no after-hours block pattern *pattern-tag*

Syntax Description

pattern-tag	Identifier for a call-blocking pattern. Up to 32 call-blocking patterns can be defined in separate commands.
pattern	Outgoing call digits to be matched for blocking.
7-24	(Optional) 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 with the **after-hours day** and **after-hours date** commands.

Command Default

No pattern is defined.

Command Modes

Call-manager-fallback configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco Unified SRST 3.0	This command was introduced.
12.3(4)T	Cisco Unified SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
15.3(2)T	Cisco Unified CME 9.5	This command was modified to include regular expressions as a value for the *pattern* argument.

Usage Guidelines

Call blocking on IP phones is defined in the following way. First, one or more patterns of outgoing digits are defined using the **after-hours block pattern** command. Next, one or more time periods during which calls that match those patterns are to be blocked are defined using the **after-hours date** or **after-hours day** command or both. By default, all IP phones in a Cisco Unified SRST system are restricted during the specified time if at least one pattern and at least one time period are defined.

Blocked calls return a fast-busy tone to the user for approximately 10 seconds before the call is terminated and the line is returned to on-hook status.

In Cisco Unified SRST 9.5, support for afterhours pattern blocking is extended to regular expression patterns for dial plans on Cisco Unified SIP and Cisco Unified SCCP IP phones. With this support, users can add a combination of fixed dial plans and regular expression-based dial plans.

When a call is initiated after hours, the dialed number is matched against a combination of dial plans. If a match is found, the call is blocked.

To enable regular expression patterns to be included when configuring afterhours pattern blocking, the **after-hours block pattern** command is modified to include regular expressions as a value for the *pattern* argument.

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

Examples

The following example defines pattern 1, which blocks international calls after hours for a Cisco Unified SRST system that requires dialing 9 for external calls:

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# after-hours block pattern 1 9011
```

The following example shows how to configure several afterhours block patterns of regular expressions:

```
Router# configure terminal
Enter configuration commands, one per line.  End with CNTL/Z.
Router(config)# call-manager-fallback

Router(config-telephony)# after-hours block pattern 1 ?
  WORD  Specific block pattern or a regular expression for after-hour block
        pattern
Router(config-cm-fallback)# after-hours block pattern 1 1234
Router(config-cm-fallback)# after-hours block pattern 2 .T
Router(config-cm-fallback)# after-hours block pattern 3 987654([1-3])+
Router(config-cm-fallback)# after-hours block pattern 4 98765432[1-9]
Router(config-cm-fallback)# after-hours block pattern 5 98765(432|422|456)
```

Related Commands

Command	Description
after-hours date (call-manager-fallback)	Defines a recurring period based on date during which outgoing calls that match defined block patterns are blocked on IP phones.
after-hours day (call-manager-fallback)	Defines a recurring period based on day of the week during which outgoing calls that match defined block patterns are blocked on IP phones.

after-hours date (call-manager-fallback)

To define a recurring period based on date during which outgoing calls that match defined block patterns are blocked on IP phones, use the **after-hours date** command in call-manager-fallback configuration mode. To delete a defined time period, use the **no** form of this command.

after-hours date *month date start-time stop-time*
no after-hours date *month date*

Syntax Description

month	Abbreviated month. The following abbreviations for month are valid: **jan**, **feb**, **mar**, **apr**, **may**, **jun**, **jul**, **aug**, **sep**, **oct**, **nov**, **dec**.
date	Date of the month. Range is from 1 to 31.
start-time stop-time	Beginning and ending times for call blocking, in an HH:MM format using a 24-hour clock. The stop time must be greater than the start time. The value 24:00 is not valid. If 00:00 is entered as a stop time, it is changed to 23:59. If 00:00 is entered for both start time and stop time, calls are blocked for the entire 24-hour period on the specified date.

Command Default

No time period based on date is defined for call blocking.

Command Modes

Call-manager-fallback configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco Unified SRST 3.0	This command was introduced.
12.3(4)T	Cisco Unified SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Usage Guidelines

Call blocking on IP phones is defined in the following way. First, one or more patterns of outgoing digits are defined using the **after-hours block pattern** command. Next, one or more time periods during which calls that match those patterns are to be blocked are defined using the **after-hours date** or **after-hours day** command or both. By default, all IP phones in a Cisco Unified SRST system are restricted during the specified time if at least one pattern and at least one time period are defined.

Call blocking for the time period that is defined in this command recurs annually on the date specified in the command.

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

Examples

The following example defines January 1 as an entire day on which calls that match the pattern specified in the **after-hours block pattern** command are blocked:

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# after-hours date jan 1 00:00 00:00
```

Related Commands

Command	Description
after-hours block pattern (call-manager-fallback)	Defines a pattern of digits for blocking outgoing calls from IP phones.
after-hours day (call- manager-fallback)	Defines a recurring period based on day of the week during which outgoing calls that match defined block patterns are blocked on IP phones.

after-hours day (call-manager-fallback)

To define a recurring period based on date during which outgoing calls that match defined block patterns are blocked on IP phones, use the **after-hours date** command in call-manager-fallback configuration mode. To delete a defined time period, use the **no** form of this command.

after-hours date *month date start-time stop-time*
no after-hours date *month date*

Syntax Description

month	Abbreviated month. The following abbreviations for month are valid: **jan**, **feb**, **mar**, **apr**, **may**, **jun**, **jul**, **aug**, **sep**, **oct**, **nov**, **dec**.
date	Date of the month. Range is from 1 to 31.
start-time stop-time	Beginning and ending times for call blocking, in an HH:MM format using a 24-hour clock. The stop time must be greater than the start time. The value 24:00 is not valid. If 00:00 is entered as a stop time, it is changed to 23:59. If 00:00 is entered for both start time and stop time, calls are blocked for the entire 24-hour period on the specified date.

Command Default

No time period based on date is defined for call blocking.

Command Modes

Call-manager-fallback configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco Unified SRST 3.0	This command was introduced.
12.3(4)T	Cisco Unified SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Usage Guidelines

Call blocking on IP phones is defined in the following way. First, one or more patterns of outgoing digits are defined using the **after-hours block pattern** command. Next, one or more time periods during which calls that match those patterns are to be blocked are defined using the **after-hours date** or **after-hours day** command or both. By default, all IP phones in a Cisco Unified SRST system are restricted during the specified time if at least one pattern and at least one time period are defined.

Call blocking for the time period that is defined in this command recurs annually on the date specified in the command.

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

Examples

The following example defines January 1 as an entire day on which calls that match the pattern specified in the **after-hours block pattern** command are blocked:

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# after-hours date jan 1 00:00 00:00
```

Related Commands

Command	Description
after-hours block pattern (call-manager-fallback)	Defines a pattern of digits for blocking outgoing calls from IP phones.
after-hours day (call- manager-fallback)	Defines a recurring period based on day of the week during which outgoing calls that match defined block patterns are blocked on IP phones.

alias (call-manager-fallback)

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

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

Syntax Description

tag	Identifier for alias rule range. The range is from 1 to 50.
number-pattern	Pattern to match the incoming telephone number. This pattern may include wildcards. A *number-pattern* can be used multiple times.
to	Connects the tag number pattern to the alternate number.
alternate-number	Alternate telephone number to route incoming calls to match the number pattern. The alternate number has to be a specific extension that belongs to an IP phone that is actively registered on the Cisco Unified SRST router. The alternate telephone number can be used in multiple **alias** commands.
preference	(Optional) Assigns a dial-peer preference value to the alias. Use with the **max-dn** command.
preference-value	(Optional) Preference value of the associated dial peer. The range is from 0 to 10.
cfw number	(Optional) 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. The **cfw** keyword assumes that you want to forward calls to the same number for both busy and no-answer call forward cases. The *timeout-value* applies only to the no-answer call forward case. There is no timeout required for a busy call forward situation.
huntstop	(Optional) Stops call hunting after trying the *alternate-number*. This keyword only has an effect if the global command **no huntstop** is set.

Command Default

No default behavior or values.

Command Modes

Call-manager-fallback configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(2)XT	Cisco Unified SRST 2.0	This command was introduced on the following platforms: Cisco 1750, Cisco 1751, Cisco 2600 series and Cisco 3600 series multiservice routers, and Cisco IAD2420 series IADs.

Cisco IOS Release	Cisco Product	Modification
12.2(8)T	Cisco Unified SRST 2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers.
12.2(8)T1	Cisco Unified SRST 2.0	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
12.2(11)T	Cisco Unified SRST 2.01	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers. The **preference** keyword was added.
12.3(11)T	Cisco Unified SRST 3.2	The **cfw** keyword was added. The maximum number of **alias** commands used for creating calls to telephone numbers that are unavailable during Cisco Unified Communications Manager fallback was increased to 50. The *alternate-number* argument can be used in multiple **alias** commands.

Usage Guidelines

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).

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 few seconds and the call is not answered, it 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 the calls are subject to the forwarding as set by the **alias** command, instead of any call forwarding that may have been set globally.

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.

Note The **alias** command supports all port types and obsoletes the **default-destination** command. The alias command is recommended over the **default-destination** command.

Examples

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

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

In the following example, alias 1 is set up to route calls coming into extensions 5000 through 5999 to extension 6000. The routing occurs when extensions 5000 through 5999 are unavailable during Cisco Unified Communications Manager fallback.

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# alias 1 5... to 6000
```

The following example sets the **preference** keyword in the **alias** command to a lower preference value than the preference number created by the **max-dn** command. Setting a lower value 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.

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# alias 1 1000 to 1000 preference 1 cfw 2000 timeout 10
Router(config-cm-fallback)# max-dn 10 preference 2
```

```
Router(config-cm-fallback)# call-forward busy 3000
Router(config-cm-fallback)# call-forward noan 3000 timeout 10
```

Related Commands

Command	Description
call-manager-fallback	Enables Cisco Unified SRST feature support and enters call-manager-fallback configuration mode.
default-destination	Assigns a default destination number for incoming telephone calls on the Cisco Unified SRST router.
max-dn (call- manager-fallback)	Sets the maximum possible number of virtual voice ports that can be supported by a router and activates dual-line mode.
show dialplan number	Displays which outgoing dial peer is reached when a particular phone number is dialed.

alias (voice register pool)

To allow Cisco Session Initiation Protocol (SIP) IP phones to handle inbound PSTN calls to telephone numbers that are unavailable when the main proxy is not available, use the **alias** command in voice register pool configuration mode. To disable rerouting of unmatched call destination calls, use the **no** form of this command.

alias *tag pattern* **to** *target* [**preference** *value*]
no alias *tag*

Syntax Description

tag	Number from 1 to 5 and the distinguishing factor when there are multiple alias commands.
pattern	Prefix number that represents a pattern against which to match the incoming telephone number. It may include wildcards.
to	Connects the number *pattern* to the *target* (alternate number).
target	Target number. An alternate telephone number to route incoming calls that match the number pattern. The target must be a full E.164 number.
preference *value*	(Optional) Assigns a dial-peer preference value to the alias. The *value* argument is the value of the associated dial peer. The range is from 1 to 10. There is no default.

Command Default

None

Command Modes

Voice register pool configuration

Command History

Release	Product Version	Modification
12.2(15)ZJ	Cisco SIP SRST 3.0	This command was introduced to Cisco Unified.
12.3(4)T	Cisco SIP SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Usage Guidelines

The **alias** command services calls placed to telephone numbers that are unavailable because the main proxy is not available. The **alias** command is activated when a Cisco SIP IP phone that has an extension number matching the target number registers.

When a phone with the target number registers, calls that match the number pattern are rerouted to the target number. The target number must be a local phone number to enable rerouting of a range of number patterns. When a Cisco SIP IP phone registers with a target number, an additional VoIP dial peer is created using the target number IP address as a session target and destination pattern as configured with the **alias** *pattern* command. For the **alias** command to work, the VoIP dial peer must be set with a translation rule to translate the called number to the target number. Translation rules can be configured under voice register pool configuration mode.

If other Cisco SIP IP phones register that have specific phone numbers that fall within the alias range or if another static dial peer exists for this pattern, the call is routed using the appropriate dial peer in preference to being rerouted to this alternate alias number (according to normal dial-peer longest-match, preference, and huntstop rules).

Note The **id (voice register pool)** command must be configured before any other voice register pool commands, including the **alias** command. The **id** command identifies a locally available individual Cisco SIP IP phone or sets of Cisco SIP IP phones. Before the **alias** command is configured, translation rules must be set using the **translate-outgoing** (voice register pool) command. Translation rules are a general-purpose number-manipulation mechanism that perform operations such as automatically adding telephone area and prefix codes to dialed numbers.

Examples

The following example configures calls to numbers in the 5000 to 5099 range that are not otherwise explicitly resolved to a specific extension number to be routed to the phone with extension 5001. Phone calls intended for phones that are not given fallback service can then be redirected to the specified extension number.

```
Router(config)# voice register pool
Router(config-register-pool)# alias 1 50.. to 5001
```

Related Commands

Command	Description
id (voice register pool)	Explicitly identifies a locally available individual Cisco SIP IP phone or set of Cisco SIP IP phones.
show dial-peer voice	Displays information for voice dial peers.
translate-outgoing (voice register pool)	Applies a translation rule to manipulate dialed digits on an outbound POTS or VoIP call leg.
voice register pool	Enables SIP SRST voice register pool configuration commands.

allow-hash-in-dn(voice register global)

To allow the insertion of '#' at any place in voice register dn, use the **allow-hash-in-dn** command in voice register global mode. To disable this, use the **no** form of this command.

allow-hash-in-dn
no allow-hash-in-dn

Syntax Description

allow-hash-in-dn	Allow the insertion of hash at all places in voice register dn.

Command Default

This command is disabled by default.

Command Modes

Voice register global configuration

Command History

Cisco IOS Release	Cisco Product	Modification
15.6(2)T	Cisco SIP SRST 11.0	This command was introduced.

Usage Guidelines

Before this command was introduced, the characters supported in voice register dn were 0-9, +, and *. The new command is enabled whenever the user requires the insertion of # in voice register dn. The command is disabled by default. You can configure this command only in Cisco Unified SRST and Cisco Unified E-SRST modes. The character # can be inserted at any place in voice register dn. When this command is enabled, users are required to change the default termination character(#) to another valid character using **?dial-peer terminator?** command under configuration mode.

Examples

The following example shows how to enable the command in mode E-SRST, SRST and how to change the default terminator:

```
Router(config)#voice register global
Router(config-register-global)#mode esrst
Router(config-register-global)#allow-hash-in-dn

Router(config)#voice register global
Router(config-register-global)#no mode [Default SRST mode]
Router(config-register-global)#allow-hash-in-dn



Router(config)#dial-peer terminator ?
WORD Terminator character: '0'-'9', 'A'-'F', '*', or '#'

Router(config)#dial-peer terminator *
```

Related Commands

Command	Description
dial-peer terminator	Configures the character used as a terminator for variable-length dialed numbers.

application (call-manager-fallback)

To select the session-level application for all Cisco IP phone lines served by the Cisco Unified SRST router, use the **application** command in call-manager-fallback configuration mode. To disable this application, use the **no** form of this command.

application *application-name*
no application *application-name*

Syntax Description

application-name	Interactive voice response (IVR) application name.

Command Default

DEFAULT system session application

Command Modes

Call-manager-fallback configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco Unified SRST 3.0	This command was introduced.
12.3(4)T	Cisco Unified SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Usage Guidelines

Use this command to assign a Tool Command Language (Tcl) IVR application to all IP phones served by the Cisco Unified SRST router.

Examples

The following example selects a Tcl IVR application named "app-xfer" for all IP phones served by the Cisco Unified SRST router:

```
Router(config)# call-manager-fallback
Router(config-cm-fallback) application app-xfer
```

Related Commands

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

application (voice register global)

To select the session-level application for all dial peers associated with Session Initiation Protocol (SIP) phones, use the **application** command in voice register global configuration mode. To disable use of the application, use the **no** form of this command.

application *application-name*
no application

Syntax Description

application-name	Interactive voice response (IVR) application name.

Command Default

Default application on router

Command Modes

Voice register global configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4 Cisco SIP SRST 3.4	This command was introduced.

Usage Guidelines

During Cisco Unified Communications Manager Express (Cisco Unified CME) or Cisco Unified Session Initiation Protocol (SIP) Survivable Remote Site Telephony (SRST) registration, a dial peer is created for each SIP phone and that dial peer includes the default session application. The **application** command allows you to change the default application for all dial peers associated with the Cisco SIP IP phones, if desired. The applied application (or TCL IVR script) must support call redirection. Use the **show call application voice summary** command to display a list of applications.

The **application** command in voice register pool configuration mode takes precedence over this command in voice register global configuration mode.

Note Configure the **id** (voice register pool) command before any other voice register pool commands, including the **application** command. The **id** command identifies a locally available individual Cisco SIP IP phone or set of Cisco SIP IP phones.

Examples

The following example shows how to set the Tcl IVR application globally for all SIP phones:

```
Router(config)# voice register global
Router(config-register-global)# mode cme
Router(config-register-global)# application sipapp2
```

Related Commands

Command	Description
application (dial-peer)	Enables a specific application on a dial peer.

Command	Description
application (voice register pool)	Selects the session-level application for the dial peer associated an individual SIP phone in a Cisco Unified CME environment or for a group of phones in a Cisco Unified SIP SRST environment.
id (voice register pool)	Explicitly identifies a locally available individual Cisco SIP IP phone, or when running Cisco Unified SIP SRST, set of Cisco SIP IP phones.
mode (voice register global)	Enables the mode for provisioning SIP phones in a Cisco Unified CME system.
show call application voice summary	Displays information about voice applications.
show dial-peer voice	Displays information for dial peers.
voice register global	Enters voice register global configuration mode in order to set global parameters for all supported Cisco SIP phones in a Cisco Unified CME or Cisco Unified SIP SRST environment.
voice register pool	Enters voice register pool configuration mode for SIP phones.

application (voice register pool)

To select the session-level application for the dial peer associated with an individual Session Initiation Protocol (SIP) phone in a Cisco Unified Communications Manager Express (Cisco Unified CME) environment or for a group of phones in a Cisco Unified SIP Survivable Remote Site Telephony (SRST) environment, use the **application** command in voice register pool configuration mode. To disable use of the application, use the **no** form of this command.

application *application-name*
no application

Syntax Description

application-name	Name of the selected interactive voice response (IVR) application name.

Command Default

Default application on router

Command Modes

Voice register pool configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco SIP SRST 3.0	This command was introduced.
12.3(4)T	Cisco SIP SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.4(4)T	Cisco CME 3.4 and Cisco SIP SRST 3.4	This command was added to Cisco CME.

Usage Guidelines

During Cisco Unified CME or Cisco Unified SIP SRST registration, a dial peer is created for each SIP phone and that dial peer includes the default session application. The **application** command allows you to change the default application for all dial peers associated with the Cisco SIP IP phones, if desired. The applied application (or TCL IVR script) must support call redirection. Use the **show call application voice summary** command to display a list of applications.

The **application** command in voice register pool configuration mode takes precedence over this command in voice register global configuration mode.

Note Configure the **id** (voice register pool) command before any other voice register pool commands, including the **application** command. The **id** command identifies a locally available individual Cisco SIP IP phone or set of Cisco SIP IP phones.

Examples

The following example shows how to set the IVR application for the SIP phone specified by the **voice register pool** command:

```
Router(config)# voice register pool 1
Router(config-register-pool) application sipapp2
```

The following partial sample output from the **show running-config** command shows that voice register pool 1 has been set up to use the SIP.app application:

```
voice register pool 1
 id network 172.16.0.0 mask 255.255.0.0
 application SIP.app
 voice-class codec 1
```

Related Commands

Command	Description
application (dial-peer)	Enables a specific application on a dial peer.
application (voice register global)	Selects the session-level application for all dial peers associated with SIP phones.
id (voice register pool)	Explicitly identifies a locally available individual Cisco SIP IP phone, or when running Cisco Unified SIP SRST, set of Cisco SIP IP phones.
mode (voice register global)	Enables the mode for provisioning SIP phones in a Cisco Unified CME system.
show call application voice summary	Displays information about voice applications.
show dial-peer voice	Displays information for dial peers.
voice register global	Enters voice register global configuration mode in order to set global parameters for all supported Cisco SIP phones in a Cisco Unified CME or Cisco Unified SIP SRST environment.
voice register pool	Enters voice register pool configuration mode for SIP phones.

attempted-registrations size

To set the size of the table that shows a number of attempted-registrations, use the attempted- registrations command in voice register global mode. To set the size of attempted-registrations table to its default value, use the no form of this command.

attempted-registrations size *size*
no attempted-registrations size

Syntax Description

size	Number of entries in attempted registrations table. Size range from 0 to 50.

Command Default

The default size for attempted registration table is 10.

Command Modes

voice register global

Command History

Cisco IOS Release	Cisco Product	Modification
15.1(2)T	Cisco Unified CME 8.1 Cisco Unified SRST 8.1	This command was introduced.

Usage Guidelines

Use this command to define the size of the table that stores information related to SIP phones that attempt to register with Cisco Unified CME or Cisco Unified SRST and fail. The default size of an attempted registration table is 10. The minimum size of attempted registration table is 0. Use the attempted-registration size 0 when you do not wish to store any information about phones attempting to register with the Cisco Unified CME or Cisco Unified SRST and fail. The maximum size of attempted registration table is 50.

When the current number of entries in the table is more than the new size that is being configured, system prompts the user for the following confirmation, “This will remove x old entries from the table. Proceed? Yes/No ?”. The default user confirmation is “No”. Where “x” represents the number of entries that will be deleted. The old entries are classified on basis of the time-stamp of the latest register attempt made by the phone.

During rollback, the user confirmation is not sought and the target configuration is applied. If the current number of entries in the table is more than the default value of the table size, then entries in excess of the default table size are cleared before reverting to the target table size.

For example, if the configured table size is 40 and there are currently 35 entries in the table, any change in the size of the attempted registration table during rollback removes 25 oldest entries leaving only the default (10) entries before making the table size equal to the size in target configuration.

Examples

The following example shows attempted-registrations size:

```
Router# conf t
Router(config)#voice register global
Router(config-register-global)#attempted-registrations size 15
!
```

Related Commands

Command	Description
clear voice register attempted- registrations	Allows to delete entries in attempted-registration table.
show voice register attempted-registrations	Displays details of phones that attempted to register and failed.

audible-tone

To configure audible tones to indicate successful join and unjoin and login and logout from any hunt group, use the **audible-tone** command in ephone or ephone-template configuration mode. To revert to the default behavior of not playing any audible tone, use the **no** form of this command.

audible-tone
no audible-tone

Syntax Description

This command has no arguments or keywords.

Command Default

By default, this feature is disabled.

Command Modes

Ephone configuration (config-ephone)

Ephone-template configuration (config-ephone-template)

Command History

Cisco IOS Release	Cisco Product	Modification
15.4(3)M	Cisco Unified Enhanced SRST 10.5	This command was introduced.

Usage Guidelines

Use the **audible-tone** command to set an audible tone to confirm successful join and unjoin and log in and log out from specific hunt groups.

Example

The following example shows how to configure audible tone in ephone configuration mode:

```
Router(config)# ephone 1
Router(config-ephone)# audible-tone
```

The following example shows how to configure audible tone in ephone-template configuration mode:

```
Router(config)# ephone-template 1
Router(config-ephone-template)# audible-tone
```

Related Commands

Command	Description
ephone-hunt group	Configures an ephone hunt group in Cisco Unified SRST.
voice-hunt group	Configures a voice hunt group in Cisco Unified SRST.

authenticate (voice register global)

To define the authenticate mode for SIP phones in a Cisco Unified CME or Cisco Unified SRST system, use the **authenticate** command in voice register global configuration mode. To return to the default, use the **no** form of this command.

Cisco IOS Release 12.4(11)XJ and later releases
authenticate {**credential** *tag* *location* | **ood-refer** | **presence** | **realm** *string* | **register**}
no authenticate {**credential** *tag* *location* | **ood-refer** | **presence** | **realm** *string* | **register**}

Cisco IOS Release 12.4(4)T
authenticate [**all**] [**realm** *string*]
no authenticate [**all**] [**realm** *string*]

Syntax Description

credential *tag*	Number that identifies the credential file to use for out-of-dialog REFER (OOD-R) or presence authentication. Range: 1 to 5.
location	Name and location of the credential file in URL format. Valid storage locations are TFTP, HTTP, and flash memory.
ood-refer	Incoming OOD-R requests are authenticated using RFC 2617-based digest authentication.
presence	Incoming presence subscription requests from an external presence server are authenticated.
realm *string*	Realm parameter for challenge and response as specified in RFC 2617 is authenticated.
register	All incoming registration requests are challenged and authenticated. Valid for Cisco Unified CME only.

Command Default

Authenticate mode is disabled.

Command Modes

Voice register global configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.
12.4(11)XJ	Cisco Unified CME 4.1 Cisco Unified SRST 4.1	The **credential**, **ood-refer**, **presence**, and **register** keywords were added. The **register** keyword replaced the **all** keyword.
12.4(15)T	Cisco Unified CME 4.1 Cisco Unified SRST 4.1	The modifications to this command were integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

The **credential** keyword allows OOD-R and presence service to use credential files for authentication. Up to five text files containing username and password pairs can be defined and loaded into the system. The contents of these five files are mutually exclusive; the username and password pairs must be unique across all the files. For Cisco Unified CME, the username and password pairs cannot be the same ones defined for SCCP or SIP phones with the **username** command.

The **ood-refer** keyword specifies that any OOD-R request that passes authentication is authorized to setup calls between referee and refer-target if OOD-R is enabled with the **refer-ood enable** command.

The **presence** keyword enables digest authentication for external watchers. Credentials are verified against a credential file stored in flash. This applies to both OOD-R and presence. The default is to authenticate all SUBSCRIBE requests from external watchers. An external watcher that passes authentication is authorized to subscribe to presence service for all lines allowed to be watched.

The **register** keyword enables authentication for registration requests in which the MAC address of the SIP phone cannot be identified by using other methods. All incoming register requests are challenged and authenticated. The **realm** keyword with the *string* argument specifies the character string to be included in the challenge.

Examples

The following example shows that all registration requests from SIP phones in a Cisco Unified CME system must be authenticated:

```
Router(config)# voice register global
Router(config-register-global)# mode cme
Router(config-register-global)# authenticate register
```

Related Commands

Command	Description
credential load	Reloads a credential file into flash memory.
mode cme	Enables the mode for provisioning SIP phones in a Cisco Unified CME system.
presence-enable	Allows incoming presence subscribe requests from SIP trunks.
refer-ood enable	Enables OOD-R processing.
username (ephone)	Defines a username and password for SCCP phones.
username (voice register pool)	Defines a username and password for authenticating SIP phones.

b2bua

To configure a dial peer associated with an individual SIP phone in a Cisco Unified Communications Manager Express (Cisco Unified CME) environment or a group of phones in a Cisco Unified Session Initiation Protocol (SIP) Survivable Remote Site Telephony (SRST) environment to point to Cisco Unity Express, use the **b2bua** command in dial-peer configuration mode. To disable B2BUA call flow on the dial peer, use the **no** form of this command.

b2bua
no b2bua

Syntax Description

This command has no arguments or keywords.

Command Default

Disabled

Command Modes

Dial-peer configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4 and Cisco SIP SRST 3.4	This command was introduced.

Usage Guidelines

Use the **b2bua** command to set the Cisco Unified CME source address as the 302 redirect contact address for all calls forwarded to Cisco Unity Express.

Note Use the **b2bua** command to configure Cisco SIP SRST 3.4 only after using the **allow-connections** command to enable B2BUA call flow on the SRST gateway.

Examples

The following example shows b2bua included in the configuration for voice dial peer 1:

```
dial-peer voice 1 voip
 destination-pattern 4...
 session target ipv4:10.5.49.80
 session protocol sipv2
 dtmf-relay sip-notify
 b2bua
```

Related Commands

Command	Description
allow-connections	Enables calls between SIP endpoints in a VoIP network.
dial-peer voice	Defines a dial peer and enters dial-peer configuration mode.
mode (voice register global)	Enables the mode for provisioning SIP phones in a Cisco Unified CME system.
show dial-peer voice	Displays information for dial peers.

Command	Description
source-address (voice register global)	Identifies the IP address and port through which SIP phones communicate with a Cisco Unified CME router.
voice register global	Enters voice register global configuration mode in order to set global parameters for all supported Cisco SIP phones in a Cisco Unified CME or Cisco Unified SIP SRST environment.

call-forward b2bua all

To enable call forwarding for a Session Initiation Protocol (SIP) back-to-back user agent (B2BUA) so that all incoming calls are forwarded to another extension, use the **callforward b2bua all command in** voice register dn or voice register pool configuration mode. To disable call forwarding, use the **no** form of this command.

call-forward b2bua all *directory-number*
no call-forward b2bua all

Syntax Description

directorynumber	Telephone number to which calls are forwarded. Represents a fully qualified E.164 number. Maximum length of the telephone number is 32.

Command Default

Disabled (no incoming call forwarding to another extension).

Command Modes

Voice register dn configuration
Voice register pool configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4 Cisco SIP SRST 3.4	This command was introduced.
Cisco IOS XE Amsterdam 17.2.1r	Cisco Unified SIP SRST 12.8	Introduced support for YANG models.

Usage Guidelines

You can apply call forward to an individual SIP extension (voice register dn) or to the SIP phone on which the extension appears (voice register pool). Use this command in voice register pool configuration mode to enable call forwarding for all extensions on a SIP phone. Use this command in voice register dn configuration mode to enable call forwarding for an individual SIP extension.

If information is configured in both voice register dn and voice register pool mode, the information under voice register dn mode takes precedence.

We recommend that you do not use this command with hunt groups. If the command is used, consider removing the phone from any assigned hunt groups unless you want to forward calls to all phones in the hunt group.

The **call-forward b2bua all** command takes precedence over the **call-forward b2bua busy** and **call-forward b2bua noan** commands.

Note This command in voice register dn configuration mode is not commonly used for Cisco Unified SIP SRST.

Examples

The following example shows how to forward all incoming calls to extension 5001 on directory number 4, to extension 5005.

```
Router(config)# voice register dn 4
```

```
Router(config-register-dn)# number 5001
Router(config-register-dn)# call-forward b2bua all 5005
```

The following example shows how to forward all incoming calls for extension 5001 on pool number 4, to extension 5005.

```
Router(config)# voice register pool 4
Router(config-register-pool)# number 5001
Router(config-register-pool)# call-forward b2bua all 5005
```

Related Commands

Command	Description
call-forward b2bua busy	Enables call forwarding for a SIP B2BUA so that incoming calls to a busy extension are forwarded to another extension.
call-forward b2bua mailbox	Controls the specific voice-mail box selected in a voice-mail system at the end of a call forwarding exchange.
call-forward b2bua noan	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.
call-forward b2bua unreachable	Enables call forwarding for a SIP B2BUA so that incoming calls to an extension that is not registered in Cisco Unified CME are forwarded to another extension.
call-waiting (voice register pool)	Enables call waiting on a SIP phone.
number (voice register dn)	Associates an extension number with a voice register dn.
voice register dn	Enters voice register dn configuration mode to define an extension for a SIP phone line.
voice register pool	Enters voice register pool configuration mode for SIP phones.

call-forward b2bua busy

To enable call forwarding for a Session Initiation Protocol (SIP) back-to-back user agent (B2BUA) so that incoming calls to a busy extension are forwarded to another extension, use the **callforward b2bua busy command in** voice register dn or voice register pool configuration mode. To disable call forwarding, use the **no** form of this command.

call-forward b2bua busy *directory-number*
no call-forward b2bua busy

Syntax Description

directorynumber	Telephone number to which calls are forwarded. Represents a fully qualified E.164 number. Maximum length of the telephone number is 32.

Command Default

Disabled (no incoming calls to a busy extension are forwarded).

Command Modes

Voice register dn configuration
Voice register pool configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4 Cisco SIP SRST 3.4	This command was introduced.

Usage Guidelines

The call-forward b2bua busy response is triggered when a call is sent to a SIP phone using a VoIP dial peer and a busy response is received back from the phone. This command functions only with phones that are registered to a Cisco Unified SIP SRST or Cisco Unified CME router.

You can apply call forward to an individual SIP extension (voice register dn) or to the SIP phone on which the extension appears (voice register pool). Use this command in voice register pool configuration mode to enable call forwarding for all extensions on a SIP phone. Use this command in voice register dn configuration mode to enable call forwarding for a specific extension. If information is configured in both voice register dn and voice register pool mode, the information under voice register dn takes precedence.

We recommend that you do not use this command with hunt groups. If the command is used, consider removing the phone from any assigned hunt groups, unless you want to forward calls to all phones in the hunt group.

The **call-forward b2bua all** command takes precedence over the **call-forward b2bua busy** and **call-forward b2bua noan** commands.

Note This command in voice register dn configuration mode is not commonly used for Cisco Unified SIP SRST.

Cisco Unified CME

Call forward busy can also get invoked if a number is unreachable but the **call forward b2bua unreachable** command is not configured.

Examples

The following example shows how to forward calls from extension 5001 in pool 4 to extension 5005 when extension 5001 is busy.

```
Router(config)# voice register pool 4
Router(config-register-pool)# number 5001
Router(config-register-pool)# call-forward b2bua busy 5005
```

Related Commands

Command	Description
call-forward b2bua all	Enables call forwarding for a SIP B2BUA so that all incoming calls are forwarded to another extension.
call-forward b2bua mailbox	Controls the specific voice-mail box selected in a voice-mail system at the end of a call forwarding exchange.
call-forward b2bua noan	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.
call-forward b2bua unreachable	Enables call forwarding for a SIP B2BUA so that incoming calls to an extension that is not registered in Cisco Unified CME are forwarded to another extension.
call-waiting (voice register pool)	Enables call waiting on a SIP phone.
number (voice register dn)	Associates an extension number with a voice register dn.
voice register dn	Enters voice register dn configuration mode to define an extension for a SIP phone line.
voice register pool	Enters voice register pool configuration mode for SIP phones.

call-forward b2bua mailbox

To control the specific voice-mail box selected in a voice-mail system at the end of a call forwarding exchange, use the **callforward b2bua mailbox command in** voice register dn or voice register pool configuration mode. To disable call forwarding, use the **no** form of this command.

call-forward b2bua mailbox *directory-number*
no call-forward b2bua mailbox

Syntax Description

directorynumber	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.

Command Default

Disabled (no voice-mail box is selected for call forwarding).

Command Modes

Voice register dn configuration
Voice register pool configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4 Cisco SIP SRST 3.4	This command was introduced.

Usage Guidelines

Use this command to denote the voice-mail box to use at the end of a chain of call forwards to busy or no answer destinations. It can be used to forward calls to a voice-mail box that has a different number than the forwarding extension. A sample of this would be in the case of a shared voice-mail box, for instance one between a manager and her assistant. This command functions only with phones that are registered to a Cisco Unified CME or Cisco Unified SIP SRST router.

If information is configured in both voice register dn and voice register pool mode, the information under voice register dn takes precedence.

It is recommended that you do not use the **call-forward b2bua mailbox** command with hunt groups. If the command is used, consider removing the phone from any assigned hunt groups, unless you want to forward calls to all phones in the hunt group.

This command is used in conjunction with the **call-forward b2bua all**, **call-forward b2bua busy**, and **call-forward b2bua noan** commands.

Note This command in voice register dn configuration mode is not commonly used for Cisco Unified SIP SRST.

Examples

The following example shows how to forward calls to extension 5005 if an incoming call is forwarded to extension 5001 on pool number 4 and extension 5001 is busy or does not answer.

```
Router(config)# voice register pool 4
```

```
Router(config-register-pool)# number 5001
Router(config-register-pool)# call-forward b2bua mailbox 5005
```

Related Commands

Command	Description
call-forward b2bua all	Enables call forwarding for a SIP B2BUA so that all incoming calls are forwarded to another extension.
call-forward b2bua busy	Enables call forwarding for a SIP B2BUA so that incoming calls to a busy extension are forwarded to another extension.
call-forward b2bua noan	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.
call-forward b2bua unreachable	Enables call forwarding for a SIP B2BUA so that incoming calls to an extension that is not registered in Cisco Unified CME are forwarded to another extension.
call-waiting (voice register pool)	Enables call waiting on a SIP phone.
number (voice register dn)	Associates an extension number with a voice register dn.
voice register dn	Enters voice register dn configuration mode to define an extension for a SIP phone line.
voice register pool	Enters voice register pool configuration mode for SIP phones.

call-forward b2bua noan

To enable call forwarding for a Session Initiation Protocol (SIP) back-to-back user agent (B2BUA) so that incoming calls to an extension that does not answer after a configured amount of time are forwarded to another extension, use the **callforward b2bua noan command in** voice register dn or voice register pool configuration mode. To disable call forwarding, use the **no** form of this command.

call-forward b2bua noan *directory-number* **timeout** *seconds*
no call-forward b2bua noan

Syntax Description

directorynumber	Telephone number to which calls are forwarded. Represents a fully qualified E.164 number. Maximum length of the telephone number is 32.
timeout *seconds*	Number of seconds that a call can ring with no answer before the call is forwarded to another extension. Range is 3 to 60000. Default is 20.

Command Default

Disabled (no incoming calls to an extension that does not answer are forwarded).

Command Modes

Voice register dn configuration
Voice register pool configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4 Cisco SIP SRST 3.4	This command was introduced.

Usage Guidelines

This command functions only with phones that are registered to a Cisco Unified SIP SRST or Cisco Unified CME router. You can apply call forward to an individual SIP extension (voice register dn) or to the SIP phone on which the extension appears (voice register pool). Use this command in voice register pool configuration mode to enable call forwarding for all extensions on a SIP phone. Use this command in voice register dn configuration mode to enable call forwarding for a specific extension.

If information is configured in both voice register dn and voice register pool mode, the information under voice register dn takes precedence.

We recommend that you do not use this command with hunt groups. If the command is used, consider removing the phone from any assigned hunt groups, unless you want to forward calls to all phones in the hunt group.

The **call-forward b2bua all** command takes precedence over the **call-forward b2bua busy** and **call-forward b2bua noan** commands.

Note This command in voice register dn configuration mode is not commonly used for Cisco Unified SIP SRST.

Examples

The following example shows how to forward calls to extension 5005 when extension 5001 on pool number 4 is unanswered. The timeout before the call is forwarded to extension 5005 is 10 seconds.

```
Router(config)# voice register pool 4
Router(config-register-pool)# number 5001
Router(config-register-pool)# call-forward b2bua noan 5005 timeout 10
```

Related Commands

Command	Description
call-forward b2bua all	Enables call forwarding for a SIP B2BUA so that all incoming calls are forwarded to another extension.
call-forward b2bua busy	Enables call forwarding for a SIP B2BUA so that incoming calls to a busy extension are forwarded to another extension.
call-forward b2bua mailbox	Controls the specific voice-mail box selected in a voice-mail system at the end of a call forwarding exchange.
call-forward b2bua unreachable	Enables call forwarding for a SIP B2BUA so that incoming calls to an extension that is not registered in Cisco Unified CME are forwarded to another extension.
call-waiting (voice register pool)	Enables call waiting on a SIP phone.
number (voice register dn)	Associates an extension number with a voice register dn.
voice register dn	Enters voice register dn configuration mode to define an extension for a SIP phone line.
voice register pool	Enters voice register pool configuration mode for SIP phones.

call-forward busy (call-manager-fallback)

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

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

Syntax Description

directory-number	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.

Command Default

No default behavior or values.

Command Modes

Call-manager-fallback configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(2)XT	Cisco Unified SRST 2.0	This command was introduced on the following platforms: Cisco 1750, Cisco 1751, Cisco 2600 series and Cisco 3600 series multiservice routers, and Cisco IAD2420 series IADs.
12.2(8)T	Cisco Unified SRST 2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers.
12.2(8)T1	Cisco Unified SRST 2.0	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
12.2(11)T	Cisco Unified SRST 2.1	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.

Usage Guidelines

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

Examples

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

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

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

The following example transforms an extension number for call forwarding when the extension number is busy. 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. 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.

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

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

Related Commands

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

call-forward noan (call-manager-fallback)

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

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

Syntax Description

directory-number	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	Sets the ringing no answer timeout duration, which is the waiting time (in seconds) before the call is forwarded to another phone.
seconds	Time (in seconds) before call forwarding starts. The range is from 3 to 60000.

Command Default

No default behavior or values.

Command Modes

Call-manager-fallback configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(2)XT	Cisco SRST 2.0	This command was introduced on the following platforms: Cisco 1750, Cisco 1751, Cisco 2600 series and Cisco 3600 series multiservice routers, and Cisco IAD2420 series IADs.
12.2(8)T	Cisco SRST 2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers.
12.2(8)T1	Cisco SRST 2.0	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
12.2(11)T	Cisco SRST 2.1	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.

Usage Guidelines

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

Examples

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

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

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

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

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

Related Commands

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

call-forward pattern (call-manager-fallback)

To specify a pattern for calling-party numbers that are able to support the ITU-T H.450.3 standard for call forwarding, use the **call-forward pattern** command in call-manager-fallback configuration mode. To remove the pattern, use the **no** form of this command.

call-forward pattern *pattern*
no call-forward pattern *pattern*

Syntax Description

pattern	String that consists of one or more digits and wildcard markers or dots (.) to define a specific pattern. Calling parties that match a defined pattern use the H.450.3 standard if they are forwarded. A pattern of .T specifies the H.450.3 forwarding standard for all incoming calls.

Command Default

No default behavior or values.

Command Modes

Call-manager-fallback configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco SRST 3.0	This command was introduced.
12.3(4)T	Cisco SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Usage Guidelines

The pattern match in this command is against the phone number of the calling party. When a directory number has forwarded its calls and an incoming call is received for that number, the SRST router sends an H.450.3 response back to the original calling party to request that the call be placed again using the forward-to destination.

Calling numbers that do not match the patterns defined with this command are forwarded using Cisco-proprietary call forwarding for backward compatibility.

Examples

The following example specifies that all 4-digit directory numbers beginning with 4 should use the H.450.3 standard whenever they are forwarded:

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# call-forward pattern 4...
```

The following example forwards all calls using the H.450.3 standard:

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# call-forward pattern .T
```

Related Commands

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

call-manager-fallback

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

call-manager-fallback
no call-manager-fallback

Syntax Description

This command has no arguments or keywords.

Command Default

No default behavior or values.

Command Modes

Global configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco SRST 1.0	This command was introduced on the following platforms: Cisco 2600 series and Cisco 3600 series multiservice routers, and Cisco IAD2420 series IADs.
12.2(2)XT	Cisco SRST 2.0	This command was implemented on Cisco 1750 and Cisco 1751 multiservice routers.
12.2(8)T	Cisco SRST 2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers.
12.2(8)T1	Cisco SRST 2.0	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
12.2(11)T	Cisco SRST 2.1	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.

Examples

The following example shows how to enter call-manager-fallback configuration mode:

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

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

Related Commands

Command	Description
access-code	Configures trunk access codes for each type of line so that the Cisco IP phones can access the trunk lines.
alias	Provides a mechanism for servicing calls to telephone numbers that are unavailable during Cisco Unified Communications Manager fallback.

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

clear voice moh-group statistics

To clear the display of MOH subsystem statistics information and reset the packet counters, use the **clear voice moh-group statistics** command in privileged EXEC mode.

clear voice moh-group statistics

Syntax Description

This command has no arguments or keywords

Command Modes

Privileged EXEC (#)

Command History

Cisco IOS Release	yCisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0 Cisco Unified SRST 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0 Cisco Unified SRST 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

Use this command to clear the display of MOH subsystem statistics information displayed by the show voice moh-group statistics command.

We recommend that the clear voice moh-group statistics should be used once every two years to reset the packet counters. Each packet counter is of 32 bit size limit and the largest count a packet counter can hold is 4294967296 intervals. This means that with 20 milliseconds packet interval (for G.711), the counters will restart from 0 any time after 2.72 years (2 years and 8 months).

Examples

```
Router# clear voice moh-group statistics
All moh group stats are cleared
```

Related Commands

Command	Description
show voice moh-group statistics	Displays the MOH subsystem statistics information
show voice moh-group	Displays the MOH groups configured

codec g722-64k

To specify that the G.722 codec should be used for Cisco Unified Survivable Remote Site Telephony (SRST) mode, use the **codec g722-64k** command in call-manager-fallback configuration mode. To disable this command and restore G.711 as the supported codec for SRST mode, use the **no** form of this command.

codec g722-64k
no codec g722-64k

Syntax Description

This command has no arguments or keywords.

Command Default

If the **codec g722-64k** command is not enabled, the G.711 codec is the default for SRST mode.

Command Modes

Call-manager-fallback configuration (config-cm-fallback)

Command History

Release	Cisco Products	Modification
15.0(1)M	Cisco Unified SRST 8.0 Cisco Unified SIP SRST 8.0	This command was introduced.

Usage Guidelines

The G.722 codec should be used for the SRST codec provided the phone supports that codec capability. For phones that do not support G.722 codec, the phones will fall back to the G.711 codec.

Examples

The following example shows how to enable support for the G.722 codec for SRST mode:

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# codec g722-64k
```

Related Commands

Command	Description
call-manager-fallback	Enters call-manager-fallback configuration mode.

conference max-length

To allow Cisco Unified SRST conferencing only if the number of dialed digits are within the maximum length limit, use the **conference max-length** command. To remove the configuration, use the **no** form of this command.

conference max-length *<value>*
no conference max-length

Syntax Description

value	Maximum number of digits that can be dialed. The range is from 3 to 16.

Command Default

No default value is defined for conferencing.

Command Modes

voice register pool (config-register-pool)

Command History

Cisco IOS Release	Cisco Product	Modification
15.4(3)M	Cisco Unified Enhanced SRST 10.5	This command was introduced.

Usage Guidelines

Use the **conference max-length** command to configure, the Cisco Unified SRST to allow conferencing, only if the dialed digits are within the maximum length limit.

Example

The following example shows how to configure the maximum number of digits that can be dialed to make a conference call:

```
Router(config)# voice register pool 1
Router(config-register-pool)# conference max-length 4
```

Related Commands

Command	Description
conference-pattern blocked	Blocks extensions on an ephone or voice register pool from making conference calls to patterns defined in the **transfer-pattern** command.
conference transfer-pattern	Allows to conference the transferred calls to phones within the local Cisco Unified SRST network.
transfer max-length	Allows the transfer of calls to phones within the local Cisco Unified SRST network.
transfer-pattern (call-manager fallback)	Allows the transfer of calls to phones outside the local Cisco Unified SRST network.

conference-pattern blocked

To prevent extensions on an ephone or a voice register pool from initiating a conference to external numbers, use the **conference-pattern blocked** command. Note that the **conference-pattern blocked** command does not impact call transfer functions. To remove the configuration, use the **no** form of this command.

conference-pattern blocked
no conference-pattern blocked

Syntax Description

This command has no arguments or keywords.

Command Default

No default behavior or values.

Command Modes

Ephone configuration (config-ephone)

Voice register pool configuration (config-register-pool)

Command History

Cisco IOS Release	Cisco Product	Modification
15.4(3)M	Cisco Unified Enhanced SRST 10.5	This command was introduced.

Usage Guidelines

Use the**conference-pattern blocked** command to prevent specific extensions from making conference calls to patterns generally allowed through the **transfer-pattern** command.

Example

This example shows how this command prevents extensions from making conference calls to patterns using the **transfer-pattern** command.

```
Router(config)# voice regsiter pool 1
Router(config-registetr-pool)# conference-pattern blocked
```

Related Commands

Command	Description
conference max-length	Allows the conferences only if the dialed digits are within the maximum length limit.
conference transfer-pattern	Allows to conference the transferred calls to phones within the local Cisco Unified SRST network.
transfer-pattern blocked	Block extensions on an ephone or voice register pool from making transferring calls to patterns defined in the **conference pattern-blocked** command.
transfer-pattern (call-manager fallback)	Allows the transfer of calls to phones outside the local Cisco Unified SRST network.

conference transfer-pattern (call-manager-fallback)

To apply transfer patterns to a conference call using conference softkeys or feature buttons in Cisco Unified SRST, use the **conference transfer-pattern** command in call-manager-fallback configuration mode. To disable the transfer patterns, use the **no** form of this command.

conference transfer-pattern
no conference transfer-pattern

Syntax Description

This command has no arguments or keywords.

Command Default

Transfer patterns do not apply to conference calls.

Command Modes

Call-manager-fallback configuration (config-cm-fallback)

Command History

Release	Modification
15.3(2)T	This command was introduced.

Usage Guidelines

There is no check for the conference numbers for call conferencing. To apply transfer patterns for call conferencing, use the **conference transfer-pattern** command.

When both the **transfer-pattern** and **conference transfer-pattern** commands are configured and dialed digits match the configured transfer pattern, conference calls are allowed. However, when the dialed digits do not match any of the configured transfer pattern, the conference call is blocked.

Examples

The following example applies transfer patterns to conference calls:

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# transfer-pattern 1234
Router(config-cm-fallback)# conference transfer-pattern
```

Related Commands

Command	Description
call-manager-fallback	Enters call-manager-fallback configuration mode to enable Cisco Unified SRST support.
transfer-pattern	Allows Cisco Unified IP phones to transfer telephone calls from callers outside the local IP network to another Cisco Unified IP phone.

cor (call-manager-fallback)

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

cor {**incoming** | **outgoing**} *cor-list-name* [{*cor-list-number* *starting-number-ending-number* | **default**}]
no **cor** *cor-list-name* *cor-list-number*

Syntax Description

incoming	COR list to be used by incoming dial peers.
outgoing	COR list to be used by outgoing dial peers.
cor-list-name	COR list name.
cor-list-number	COR list identifier. The maximum number of COR lists that can be created is 20, comprised of incoming or outgoing dial peers. The first six COR lists are applied to a range of directory numbers. The directory numbers that do not have a COR configuration are assigned to the default COR list, provided that a default COR list has been defined.
starting-number - ending-number	Directory number range; for example, 2000 - 2025.
default	Instructs the COR list to assume behavior according to a predefined default COR list.

Command Default

No default behavior or values.

Command Modes

Call-manager-fallback configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(2)XT	Cisco SRST 2.0	This command was introduced on the following platforms: Cisco 1750, Cisco 1751, Cisco 2600 series and Cisco 3600 series multiservice routers, and Cisco IAD2420 series IADs.
12.2(8)T	Cisco SRST 2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers.
12.2(8)T1	Cisco SRST 2.0	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
12.2(11)T	Cisco SRST 2.01	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers. The **default** keyword was added.
12.3(11)T	Cisco SRST 3.2	The maximum number of COR lists that can be created was increased to 20.

Usage Guidelines

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

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

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

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 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 to be 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 an outgoing call.

By default, an incoming call leg has the highest COR priority. The outgoing COR list has the lowest. If there is no COR configuration for incoming calls on a dial peer, 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. Incoming and outgoing lists are shown in Table 1: Combinations of COR List and Results, on page 56.

Table 1: Combinations of COR List and Results

COR List on Incoming Dial Peer	COR List on Outgoing Dial Peer	Result
No COR	No COR	Call will succeed.
No COR	COR list applied for outgoing calls	Call will succeed. By default, the incoming dial peer has the highest COR priority when no COR is applied. If you apply no COR for an incoming call leg to a dial peer, 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 on Incoming Dial Peer	COR List on Outgoing Dial Peer	Result
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.

Examples

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

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

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

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# cor outgoing LockforPhoneC 1 5010 - 5020
```

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

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

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

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

```
Router(config)# dial-peer cor list call911
Router(config-dp-corlist)# member 911
Router(config)# dial-peer cor list call1800
Router(config-dp-corlist)# member 1800
Router(config)# dial-peer cor list call1900
Router(config-dp-corlist)# member 1900
Router(config)# dial-peer cor list calllocal
Router(config-dp-corlist)# member local_call
Router(config)# dial-peer cor list engineering
```

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

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

```
Router(config)# dial-peer voice 1 voip
Router(config-dial-peer)# destination pattern 734....
Router(config-dial-peer)# session target ipv4:1.1.1.1
Router(config-dial-peer)# cor outgoing calllocal
Router(config)# dial-peer voice 2 voip
Router(config-dial-peer)# destination pattern 1800.......
Router(config-dial-peer)# session target ipv4:1.1.1.1
Router(config-dial-peer)# cor outgoing call1800
Router(config)# dial-peer voice 3 pots
Router(config-dial-peer)# destination pattern 1900.......
Router(config-dial-peer)# port 1/0/0
Router(config-dial-peer)# cor outgoing call1900
Router(config)# dial-peer voice 4 pots
Router(config-dial-peer)# destination pattern 911
Router(config-dial-peer)# port 1/0/1
Router(config-dial-peer)# cor outgoing call911
Router(config)# dial-peer voice 5 pots
Router(config-dial-peer)# destination pattern 316....
Router(config-dial-peer)# port 1/1/0
! No cor is applied.
```

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

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

The example configuration allows for the following:

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

Related Commands

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

Command	Description
corlist-outgoing	Specifies the COR list to be used by outgoing dial peers.
dial-peer cor list	Defines a COR list name.

cor (voice register pool)

To configure a class of restriction (COR) on the VoIP dial peers associated with directory numbers, use the **cor command in** voice register pool configuration mode. To disable a COR associated with directory numbers, use the **no** form of this command.

cor {**incoming** | **outgoing** }*cor-list-name* {*cor-list-number starting-number [- ending-number]* | **default** }
no cor {**incoming** | **outgoing**}*cor-list-name cor-list-name* {*cor-list-number starting-number [- ending-number]* | **default**}

Syntax Description

incoming	COR list to be used by incoming dial peers.
outgoing	COR list to be used by outgoing dial peers.
cor-list-name	COR list name.
cor-list-number	COR list identifier.
starting-number	Start of a directory number range, if an ending number is included. Can also be a standalone number.
-	(Optional) Indicator that a full range is configured.
ending-number	(Optional) End of a directory number range.
default	Instructs the COR list to assume behavior according to a predefined default COR list.

Command Default

None

Command Modes

Voice register pool configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco SIP SRST 3.0	This command was introduced.
12.3(4)T	Cisco SIP SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.4(4)T	Cisco CME 3.4 Cisco SIP SRST 3.4	This command was added to Cisco Communications Manager Express (Cisco CME).

Usage Guidelines

The **cor** command sets the dial-peer COR parameter for dynamically created VoIP dial peers. A list-based mechanism assigns COR parameters to specific set of number ranges. The COR functionality provides the ability to deny certain call attempts on the basis of the incoming and outgoing class of restrictions provisioned on the dial peers. This functionality provides flexibility in network design, allows users to block calls (for example, calls to 900 numbers), and applies different restrictions to call attempts from different originators.

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

A default COR is assigned to the directory numbers that do not match any COR list number or number range. During Cisco Unified Session Initiation Protocol (SIP) Survivable Remote Site Telephony (SRST) registration, a dial peer is created and that dial peer includes a default COR value. The **cor** command allows you to change the automatically selected default.

In dial-peer configuration mode, build your COR list and add members. Then in voice register pool configuration mode, use the **cor** command to apply the name of the dial-peer COR list.

You can have up to four COR lists for the Cisco Unified SIP SRST configuration, comprised of incoming or outgoing dial peers. The first four COR lists are applied to a range of phone numbers. The phone numbers that do not have a COR list configuration are assigned to the default COR list, providing that a default COR list has been defined.

Note Configure the **id** (voice register pool) command before any other voice register pool commands, including the **cor** command. The **id** command identifies a locally available individual Cisco SIP IP phone or set of Cisco SIP IP phones.

Examples

The following is sample output from the **show running-config** command:

```
..
voice register pool 1
 id mac 0030.94C2.A22A
 preference 5
 cor incoming call91 1 91011
 translate-outgoing called 1
 proxy 10.2.161.187 preference 1 monitor probe icmp-ping
 alias 1 94... to 91011 preference 8
 voice-class codec 1
.
.
.
dial-peer cor custom
 name 95
 name 94
 name 91
!
dial-peer cor list call91
 member 91
!
dial-peer voice 91500 pots
 corlist incoming call91
 corlist outgoing call91
 destination-pattern 91500
 port 1/0/0
.
.
.
```

Related Commands

Command	Description
dial-peer cor custom	Specifies that named CORs apply to dial peers.
dial-peer cor list	Defines a COR list name.

Command	Description
id (voice register pool)	Explicitly identifies a locally available individual Cisco SIP IP phone, or when running Cisco Unified SIP SRST, set of Cisco SIP IP phones.
member (dial-peer cor list)	Adds a member to a dial-peer COR list.
name (dial-peer custom cor)	Provides a name for a custom COR.
show dial-peer voice	Displays information for voice dial peers.
voice register pool	Enables Cisco Unified SIP SRST voice register pool configuration commands.

credentials

To enter credentials configuration mode to configure a certificate for a Cisco Unified CME CTL provider or for Cisco Unified SRST router communication to Cisco Unified Communications Manager, use the **credentials** command in global configuration mode. To set all commands in credentials configuration mode to the default of nonsecure, use the **no** form of this command.

credentials
no credentials

Syntax Description

This command has no arguments or keywords.

Command Default

Credentials are not provided.

Command Modes

Global configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(14)T	Cisco SRST 3.3	This command was introduced for Cisco Unified SRST.
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced for Cisco Unified CME.
Cisco IOS XE Fuji Release, 16.7.1	Unified SRST 12.1	This command was introduced for Unified SRST support on Cisco 4000 Series Integrated Services Router.
Cisco IOS XE Dublin 17.10.1a	Cisco Unified SRST 14.3	Introduced support for YANG models.

Usage Guidelines

This command is used to configure credentials service for Cisco Unified CME and Cisco Unified SRST.

Cisco Unified CME

This command is used with Cisco Unified CME phone authentication to configure a CTL provider on each Cisco Unified CME router on which the CTL client is not running. That is, if there is a primary and a secondary Cisco Unified CME router and the CTL client is running on the primary router, a CTL provider must be configured on the secondary router, and vice versa. If the CTL client is running on a router that is not a Cisco Unified CME router, CTL providers must be configured on all Cisco Unified CME routers.

Credentials service for Cisco Unified CME runs on default port 2444.

Cisco Unified SRST

The credential server provides certificates to any device that requests a certificate. The credentials server does not request any data from a client; thus no authentication is necessary. When the client, Cisco Unified Communications Manager, requests a certificate, the credentials server provides the certificate. Cisco Unified Communications Manager exports the certificate to the phone, and the Cisco Unified IP phone holds the SRST router certificate in its configuration file. The device certificate for secure SRST routers is placed in the configuration file of the Cisco Unified IP phone because the entry limit in the certificate trust list (CTL) of Cisco Unified Communications Manager is 32.

Credentials service for SRST runs on default port 2445. Cisco Unified Communications Manager connects to port 2445 on the secure SRST router and retrieves the secure SRST device certificate during the TLS handshake.

Activate this command on all SRST routers.

Caution For security reasons, credentials service should be deactivated on all SRST routers after provisioning to Cisco Unified Communications Manager is completed.

Examples

Cisco Unified CME

The following example configures a CTL provider on the Cisco Unified CME router with the IP address 172.19.245.1. CTL providers must be configured on all Cisco Unified CME routers on which the CTL client is not running.

```
Router(config)# credentials
Router(config-credentials)# ip source-address 172.19.245.1 port 2444
Router(config-credentials)# trustpoint cmeca
Router(config-credentials)# ctl-service admin user4 secret 0 c89L8o
```

Cisco Unified SRST

The following example enters credentials configuration mode and sets the IP source address and the trustpoint:

```
Router(config)# credentials
Router(config-credentials)# ip source-address 10.6.21.4 port 2445
Router(config-credentials)#
trustpoint srstca
```

Related Commands

Command	Description
ctl-service admin	Specifies a user name and password to authenticate the CTL client during the CTL protocol.
debug credentials	Sets debugging on the credentials service that runs between a Cisco Unified CME CTL provider the CTL client or between an SRST router and Cisco Unified Communications Manager.
ip source-address (credentials)	Enables the Cisco Unified CME or SRST router to receive messages through the specified IP address and port.
show credentials	Displays the credentials settings on a Cisco Unified CME or SRST router.
trustpoint (credentials)	Specifies the name of the trustpoint to be associated with a Cisco Unified CME CTL provider certificate or with an SRST router certificate.

date-format (call-manager-fallback)

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

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

Syntax Description

mm-dd-yy	Sets the date format to month, day, year. Each element requires a two-digit number. This format is the default setting.
dd-mm-yy	Sets the date format to day, month, year. Each element requires a two-digit number.
yy-dd-mm	Sets the date format to year, day, month. Each element requires a two-digit number.
yy-mm-dd	Sets the date format to year, month, day. Each element requires a two-digit number.

Command Default

mm-dd-yy

Command Modes

Call-manager-fallback configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(2)XT	Cisco SRST 2.0	This command was introduced on the following platforms: Cisco 1750, Cisco 1751, Cisco 2600 series and Cisco 3600 series multiservice routers, and Cisco IAD2420 series IADs.
12.2(8)T	Cisco SRST 2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers.
12.2(8)T1	Cisco SRST 2.0	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
12.2(11)T	Cisco SRST 2.01	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.
12.2(15)ZJ	Cisco SRST 3.0	This command was integrated into Cisco IOS Release 12.2(15)ZJ.
12.3(4)T	Cisco SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Examples

The following example sets the date format:

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

Related Commands

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

default-destination

To create a default call routing path for incoming calls on Foreign Exchange Office (FXO) ports during a WAN outage, use the **default-destination** command in call-manager-fallback configuration mode. To delete the default destination number on the Cisco Unified SRST router, use the **no** form of this command.

default-destination *telephone-number*
no default-destination *telephone-number*

Syntax Description

telephone-number	Telephone number of the default destination.

Command Default

No default behavior or values.

Command Modes

Call-manager-fallback configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco SRST 1.0	This command was introduced on the following platforms: Cisco 2600 series and Cisco 3600 series multiservice routers, and Cisco IAD2420 series IADs.
12.2(2)XT	Cisco SRST 2.0	This command was implemented on Cisco 1750 and Cisco 1751 multiservice routers.
12.2(8)T	Cisco SRST 2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers.
12.2(8)T1	Cisco SRST 2.0	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
12.2(11)T	Cisco SRST 2.01	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.

Usage Guidelines

The **default-destination** command creates a temporary Private Line Auto Ringdown (PLAR) connection configuration on FXO ports during fallback. The **default-destination** command has been obsoleted by the **alias** command, which applies to all port types. Use of the **alias** command is recommended over the **default-destination** command.

Examples

The following example sets the default destination to 40802:

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

Related Commands

Command	Description
alias (call-manager- fallback)	Provides a mechanism for servicing calls to telephone numbers that are unavailable during Cisco Unified Communications Manager fallback.
call-manager-fallback	Enables Cisco Unified SRST feature support and enters call-manager-fallback configuration mode.

description (voice moh-group)

To display a brief description specific to a MOH group, use the **description** command in voice moh-group configuration mode. To remove the description, use the **no** form of this command.

description *string*
no description

Syntax Description

string	An alphanumeric string to add a brief description specific to a MOH group. Maximum length: 80 characters including spaces.

Command Default

No MOH group description is configured.

Command Modes

V
oice moh-group configuration (config-voice-moh-group)

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0 Cisco Unified SRST 8.0	This command was introduced
15.1(1)T	Cisco Unified CME 8.0 Cisco Unified SRST 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

This command allows you to type a brief text describing a specific voice-moh-group. You can use maximum 80 characters, including spaces to describe a MOH group.

Examples

The following example provides a description for voice-moh-group1:

```
Router(config)#
Router(config-voice-moh-group)#
Router(config-voice-moh-group) descri
ption this is a moh group for sales
```

Related Commands

Command	Description
voice-moh-group	Enter voice-moh-group configuration mode.
moh	Enables music on hold from a flash audio feed
multicast moh	Enables multicast of the music-on-hold audio stream.
extension-range	Specifies the extension range for a clients calling a voice-moh-group.

device-id

To specify the device ID of a phone type, use the **device-id** command in ephone-type configuration mode. To reset to the default value, use the **no** form of this command.

device-id *number*
no device-id

Syntax Description

number	Device ID of the phone type. Range: 1 to 2,147,483,647. Default: 0. SeeTable 2: Supported Values for Ephone-Type Commands, on page 70 for a list of supported device IDs.

Command Default

Device ID is 0.

Command Modes

Ephone-type configuration (config-ephone-type)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)XZ	Cisco Unified CME 4.3 Cisco Unified SRST 4.3	This command was introduced.

Usage Guidelines

This command specifies the device ID of the type of phone being added with the ephone-type template. If this command is set to the default value of 0, the ephone-type is invalid.

Table 2: Supported Values for Ephone-Type Commands

Supported Device	device-id	num-buttons	max-presentation
Cisco Unified IP Conference Station 7937G	431	1	6
Nokia E61	376	1	1

Examples

The following example shows the device ID is set to 376 for the Nokia E61 when creating the ephone-type template:

```
Router(config)# ephone-type E61
Router(config-ephone-type)# device-id 376
Router(config-ephone-type)# device-name E61 Mobile Phone
```

Related Commands

Command	Description
device-name	Assigns a name to a phone type in an ephone-type template.
load	Associates a type of phone with a phone firmware file.
type	Assigns the phone type to a SCCP phone.

device-name

To assign a name to a phone type in an ephone-type template, use the **device-name** command in ephone-type configuration mode. To remove the name, use the **no** form of this command.

device-name *name*
no device-name

Syntax Description

name	String that identifies this phone type. Value is any alphanumeric string up to 32 characters.

Command Default

No name is assigned to this phone type.

Command Modes

Ephone-type configuration (config-ephone-type)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)XZ	Cisco Unified CME 4.3 Cisco Unified SRST 4.3	This command was introduced.

Usage Guidelines

This command specifies a device name for the type of phone being added with the ephone-type template.

Examples

The following example shows that the name "E61 Mobile Phone" is assigned to a phone type when creating the ephone-type template:

```
Router(config)# ephone-type E61
Router(config-ephone-type)# device-id 376
Router(config-ephone-type)# device-name E61 Mobile Phone
```

Related Commands

Command	Description
device-id	Specifies the device ID for a phone type in an ephone-type template.

device-type

To specify the phone type, use the **device-type** command in ephone-type configuration mode. To reset to the default value, use the **no** form of this command.

device-type *phone-type*
no device-type

Syntax Description

phone-type	Device type of the phone. See Table 3: Supported Values for Ephone-Type Commands, on page 72 for a list of supported device types. Default value is the same value entered with the **ephone-type** command.

Command Default

Device type is the same value that is entered with the **ephone-type** command.

Command Modes

Ephone-type configuration (config-ephone-type)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)XZ	Cisco Unified CME 4.3 Cisco Unified SRST 4.3	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0 Cisco Unified SRST 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

This command specifies the device type of the phone being added with the ephone-type template. The device type is set to the same value as the **ephone-type** command unless you use this command to change the value.

This command must be set to one of the following supported values.

Table 3: Supported Values for Ephone-Type Commands

Supported Device	device-id	device-type	num-buttons	max-presentation
Cisco Unified IP Phone 7915 Expansion Module with 12 buttons	227	7915	12	0 (default)
Cisco Unified IP Phone 7915 Expansion Module with 24 buttons	228	7915	24	0
Cisco Unified IP Phone 7916 Expansion Module with 12 buttons	229	7916	12	0
Cisco Unified IP Phone 7916 Expansion Module with 24 buttons	230	7916	24	0
Cisco Unified IP Conference Station 7937G	431	7937	1	6
Cisco Unified IP Phone 8941	586	8941	4	3
Cisco Unified IP Phone 8945	585	8945	4	3

Supported Device	device-id	device-type	num-buttons	max-presentation
Nokia E61	376	E61	1	1

Examples

The following example shows the device type set to 7915 in the ephone-type template for the Cisco Unified IP Phone 7915 Expansion Module with 12 buttons:

```
Router(config)# ephone-type 7915-12 addon
Router(config-ephone-type)# device-id 227
Router(config-ephone-type)# device-name 7915-12
Router(config-ephone-type)# device-type 7915
```

Related Commands

Command	Description
device-name	Assigns a name to a phone type in an ephone-type template.
ephone-type	Adds a Cisco Unified IP phone type by defining an ephone-type template.
load	Associates a type of phone with a phone firmware file.
type	Assigns the phone type to a SCCP phone.

dialplan-pattern (call-manager-fallback)

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

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

Syntax Description

tag	Dial-plan string tag used before a ten-digit telephone number. The tag number is from 1 to 5.
pattern	Dial-plan pattern, such as the area code, the prefix, and the first one or two digits of the extension number, plus wildcard markers or dots (.) for the remainder of the extension number digits.
extension-length	Sets the number of extension digits that will appear as a caller ID.
extension-length	The number of extension digits. The extension length must match the setting for IP phones in Cisco Unified Communications Manager mode. The range is from 1 to 32.
extension-pattern	(Optional) Sets an extension number's leading digit pattern when it is different from the E.164 telephone number's leading digits defined in the *pattern* variable.
extension-pattern	(Optional) The extension number's leading digit pattern. Consists of one or more digits and wildcard markers or dots (.). For example, 5.. would include extensions 500 to 599; 5... would include extensions 5000 to 5999. The extension pattern must match the setting for IP phones in Cisco Unified Communications Manager mode.
no-reg	(Optional) Prevents the E.164 numbers in the dial peer from registering with the gatekeeper.

Command Default

No default behavior or values.

Command Modes

Call-manager-fallback configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco SRST 1.0	This command was introduced on the Cisco 2600 series and Cisco 3600 series multiservice routers and on the Cisco IAD2420 series.
12.2(2)XT	Cisco SRST 2.0	This command was implemented on the Cisco 1750 and Cisco 1751 multiservice routers.

Cisco IOS Release	Cisco Product	Modification
12.2(8)T	Cisco SRST 2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745 routers.
12.2(8)T1	Cisco SRST 2.0	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
12.2(11)T	Cisco SRST 2.01	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.
12.2(11)YT	Cisco SRST 2.1	The **extension-pattern** keyword was added.
12.4(20)T	Cisco Unified CME 7.0 Cisco Unified SRST 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

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

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

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

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

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

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

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

For outbound calls, the **dialplan-pattern** command converts the calling party’s extension number to an E.164 calling party number. Outbound calls that do not use an E.164 number and go through a PRI connection to the PSTN may be rejected by the PRI link as the calling party identifier.

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

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

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

The number of *extension-pattern* argument characters must match the number set for the *extension-length* argument. For example, if the *extension-length* is 3, the *extension-pattern* can be 8.., 1.., 51., and so forth.

A dial-plan pattern is required to register the Cisco IP phone lines with a gatekeeper. The **no-reg** keyword provides the option of not registering specific numbers to the gatekeeper so that those numbers can be used for other telephony services.

Examples

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

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

In the following example, the **dialplan-pattern** command creates dial-plan pattern 1 for extensions 800 to 899 with the telephone prefix starting with 4085559. As each number in the extension pattern is declared with the **number** command, two POTs dial peers are created. In the example, they are 801 (an internal office number) and 4085559001 (an external number).

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

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

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

Related Commands

Command	Description
call-manager-fallback	Enables Cisco Unified SRST support and enters call-manager-fallback configuration mode.
show dial-peer voice	Displays information for voice dial peers.

digit collect kpml

To enable Key Press Markup Language (KPML) digit collection on a SIP phone, use the **digit collect kpml** command in voice register pool or voice register template configuration mode. To disable KPML, use the **no** form of this command.

digit collect kpml
no digit collect kpml

Syntax Description

This command has no arguments or keywords.

Command Default

KPML digit collection is enabled.

Command Modes

Voice register pool configuration
Voice register template configuration

Command History

Release	Cisco Product	Modification
12.4(11)XJ	Cisco Unified CME 4.1 Cisco Unified SRST 4.1	This command was introduced.
12.4(15)T	Cisco Unified CME 4.1 Cisco Unified SRST 4.1	This command was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

KPML is enabled by default for all directory numbers on the phone. A dial plan assigned to a phone has priority over KPML. Use the **no digit collect kpml** command to disable KPML on a phone.

If you use a voice register template to apply a command to a phone and you also use the same command in voice register pool configuration mode for the same phone, the value that you set in voice register pool configuration mode has priority.

KPML is not supported on the Cisco IP Phone 7905, 7912, 7940, or 7960.

Examples

The following example shows KPML enabled on SIP phone 4:

```
Router(config)# voice register pool 4
Router(config-register-pool)# digit collect kpml
```

Related Commands

Command	Description
dialplan	Assigns a dial plan to a SIP phone.
show voice register pool	Displays all configuration information associated with a SIP phone.
voice register dialplan	Enters voice register dialplan configuration mode to define a dial plan for SIP phones.

dtmf-relay (voice register pool)

To specify the list of DTMF relay methods that can be used to relay dual-tone multifrequency (DTMF) audio tones between Session Initiation Protocol (SIP) endpoints, use the **dtmf-relay** command in voice register pool configuration mode. To send the DTMF audio tones as part of an audio stream, use the **no** form of this command.

dtmf-relay [**cisco-rtp**] [**rtp-nte**] [**sip-notify**]
no dtmf-relay

Syntax Description

cisco-rtp	Forwards DTMF audio tones by using Real-Time Transport Protocol (RTP) with a Cisco proprietary payload type. This keyword is supported only for dial peers that are created by incoming REGISTERs from a SIP gateway. It is not supported for dial peers that are created by a SIP Cisco IP phone.
rtp-nte	Forwards DTMF audio tones by using Real-Time Transport Protocol (RTP) with a Named Telephone Event (NTE) payload.
sip-notify	Forwards DTMF audio tones by using SIP-NOTIFY messages. This keyword is supported only for dial peers that are created by incoming REGISTERs from a SIP gateway. It is not supported for dial peers that are created by a SIP Cisco IP phone.

Command Default

DTMF tones are disabled and sent in-band. That is, they remain in the audio stream.

Command Modes

Voice register pool configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(4)T	Cisco SIP SRST 3.0	This command was introduced.
12.4(4)T	Cisco CME 3.4 Cisco SIP SRST 3.4	This command was added to Cisco Unified CME.
Cisco IOS XE Amsterdam 17.2.1r	Cisco Unified SIP SRST 12.8	Introduced support for YANG models.

Usage Guidelines

During Cisco Unified Session Initiation Protocol (SIP) Survivable Remote Site Telephony (SRST) or Cisco Unified CME registration, a dial peer is created and that dial peer has a default DTMF relay of in-band.

This command command allows you to change the default to a desired value. You must use one or more keywords when configuring this command.

DTMF audio tones are generated when you press a button on a Touch-Tone phone. The tones are compressed at one end of the call and when the digits are decompressed at the other end, there is a risk that they can become distorted. DTMF relay reliably transports the DTMF audio tones generated after call establishment out-of-band.

The SIP Notify method sends Notify messages bidirectionally between the originating and terminating gateways for a DTMF event during a call. If multiple DTMF relay mechanisms are enabled on a SIP dial peer and are negotiated successfully, the SIP Notify method takes precedence.

SIP Notify messages are advertised in an Invite message to the remote end only if the **dtmf-relay** command is set.

For SIP calls, the most appropriate methods to transport DTMF tones are RTP-NTE or SIP-NOTIFY.

Note The **cisco-rtp** keyword is a proprietary Cisco implementation. If the proprietary Cisco implementation is not supported, the DTMF relay feature does not function, and the gateway sends DTMF tones in-band.

- The **sip-notify** keyword is available only if the VoIP dial peer is configured for SIP.

Examples

Cisco Unified CME

The following example shows how to enable the RTP-NTE and SIP-NOTIFY mechanisms for DTMF relay for SIP phone 4:

```
Router(config)# voice register pool 4
Router(config-register-pool)# dtmf-relay rtp-nte sip-notify
```

Cisco Unified SIP SRST

The following is sample output from the **show running-config** command that shows that voice register pool 1 has been set up to send DTMF tones:

```
voice register pool 1
 application SIP.app
 incoming called-number 308
 voice-class codec 1
 dtmf-relay rtp-nte
```

Related Commands

Command	Description
dtmf-relay (voice over IP)	Specifies how an H.323 or SIP gateway relays DTMF tones between telephony interfaces and an IP network.

elin

To create a PSTN number that replaces a 911 caller's extension, use the **elin** command in voice emergency response location configuration mode. To remove the number, use the **no** form of this command.

elin [{**1** | **2**}] *number*
no elin [{**1** | **2**}]

Syntax Description

number	PSTN number that replaces a 911 caller's extension.

Command Default

No replacement number is created.

Command Modes

Voice emergency response location configuration (cfg-emrgncy-resp-location)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)T	Cisco Unified CME 4.1 Cisco Unified SRST 4.1 Cisco Unified SIP SRST 4.1	This command was introduced. For Cisco Unified CME, this command is supported in SRST fallback mode only.
12.4(15)XY	Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1) Cisco Unified SIP SRST 4.2(1)	This command was added for Cisco Unified CME.
12.4(20)T	Cisco Unified CME 7.0 Cisco Unified SRST 7.0 Cisco Unified SIP SRST 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

Use this command to specify the ELIN, a PSTN number that will replace the caller's extension. The PSAP will see this number and use it to query the ALI database to locate the caller. It is also used by the PSAP for callbacks. You can optionally configure a second ELIN using the **elin 2** command. If two ELINs are configured, the system selects an ELIN using a round-robin algorithm. If an ELIN is not defined for the ERL, the PSAP will see the original calling number.

Examples

In the following example, all IP phones with the IP address of 10.X.X.X or 192.168.X.X are automatically associated with this ERL. If one of the phones dials 911, its extension is replaced with 408 555-0100 before it goes to the PSAP. The PSAP will see that the caller's number is 408 555-0100.

```
voice emergency response location 1
 elin 1 4085550100
 subnet 10.0.0.0 255.0.0.0
 subnet 2 192.168.0.0 255.255.0.0
```

Related Commands

Command	Description
voice emergency response location	Creates a tag for identifying an ERL for Enhanced 911 Services.
subnet	Defines which IP phones are part of this ERL.

elin (voice emergency response settings)

To create a default ELIN that is used if no ERL has a subnet mask that matches the current 911 caller's IP phone address, use the **elin** command in voice emergency response settings configuration mode. To remove the number, use the **no** form of this command.

elin *number*
no elin

Syntax Description

number	An E.164 number to be used as the default ELIN.

Command Default

No default ELIN number is created.

Command Modes

Voice emergency response settings configuration (cfg-emrgncy-resp-settings)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)XY	Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1) Cisco Unified SIP SRST 4.2(1)	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0 Cisco Unified SRST 7.0 Cisco Unified SIP SRST 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

This command specifies an E.164 number to be the default ELIN if the 911 caller's IP phone address does not match the subnet of any location in any ERL zone. The default ELIN can be an existing ELIN already defined in an ERL or it can be unique.

Examples

In this example, the ELIN (4085550101) defined in the voice emergency response settings configuration is used if the 911 caller's IP phone address does not match any of the voice emergency response locations. After the 911 call is placed to the PSAP, the PSAP has 120 minutes to call back 408 555-0101 to reach the 911 caller. If the call history has expired (after 120 minutes), any callback is routed to extension 7500.

```
voice emergency response settings
callback 7500
elin 4085550101
expiry 120
```

Related Commands

Command	Description
callback	Default phone number to contact if a 911 callback cannot find the last 911 caller from the ERL.
expiry	Number of minutes a 911 call is associated to an ELIN in case of a callback from the 911 operator.

Command	Description
logging	Syslog informational message printed to the console each time an emergency call is made.
voice emergency response settings	Creates a tag for identifying settings for E911 behavior.

emergency response callback

To define a dial peer that is used for 911 callbacks from the PSAP, use the emergency response callback command in voice dial-peer configuration mode. To remove the definition of the dial peer as an incoming link from the PSAP, use the **no** form of this command.

emergency response callback
no emergency response callback

Syntax Description

This command has no arguments or keywords.

Command Default

The dial peer is not defined as an incoming link from the PSAP.

Command Modes

Dial-peer configuration (config-dial-peer)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)T	Cisco Unified CME 4.1 Cisco Unified SRST 4.1 Cisco Unified SIP SRST 4.1	This command was introduced. For Cisco Unified CME, this command is supported in SRST fallback mode only.
12.4(15)XY	Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1) Cisco Unified SIP SRST 4.2(1)	This command was added for Cisco Unified CME.
12.4(20)T	Cisco Unified CME 7.0 Cisco Unified SRST 7.0 Cisco Unified SIP SRST 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

This command defines which dial peer is used for 911 callbacks from the PSAP. You can define multiple dial peers with this command.

Examples

The following example shows a dial peer defined as an incoming link from the PSAP. If 408 555-0100 is configured as the ELIN for an ERL, this dial peer recognizes that an incoming call from 408 555-0100 is a 911 callback.

```
dial-peer voice 100 pots
 incoming called-number 4085550100
 port 1/1:D
 direct-inward-dial
 emergency response callback
```

Related Commands

Command	Description
emergency response location	Associates an ERL to either a SIP phone, ephone, or dial peer.
emergency response zone	Defines a dial peer that is used by the system to route emergency calls to the PSAP.
voice emergency response location	Creates a tag for identifying an ERL for the enhanced 911 service.

emergency response location

To associate an emergency response location (ERL) for Enhanced 911 Services with a dial peer, ephone, ephone-template, voice register pool, or voice register template, use the **emergency response location** command in dial peer, ephone, ephone-template, voice register pool, or voice register template configuration mode. To remove the association, use the **no** form of this command.

emergency response location *tag*
no emergency response location *tag*

Syntax Description

tag	Unique number that identifies an existing ERL tag defined by the **voice emergency response location** command.

Command Default

No ERL tag is associated with a dial peer, ephone, ephone-template, voice register pool, or voice register template.

Command Modes

Dial-peer configuration (config-dial-peer)
Ephone configuration (config-ephone)
Ephone-template configuration (config-ephone-template)
Voice register pool configuration (config-register-pool)
Voice register template configuration (config-register-template)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)T	Cisco Unified CME 4.1 Cisco Unified SRST 4.1 Cisco Unified SIP SRST 4.1	This command was introduced. For Cisco Unified CME, this command is supported in SRST fallback mode only.
12.4(15)XY	Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1) Cisco Unified SIP SRST 4.2(1)	This command was added to Cisco Unified CME in the ephone-template and voice register template configuration modes.

Usage Guidelines

Use this command to assign an ERL to phones individually. Depending on the type of phones (endpoints) that you have, you can assign an ERL to a phone's:

- Dial-peer configuration
- Ephone
- Ephone-template
- Voice register pool
- Voice register template

These methods of associating a phone with an ERL are alternatives to assigning a group of phones that are on the same subnet as an ERL.

The tag used by this command is an integer from 1 to 2147483647 and refers to an existing ERL tag that is defined by the **voice emergency response location** command. If the tag does not refer to a valid ERL configuration, the phone is not associated to an ERL. For IP phones, the IP address is used to find the inclusive ERL subnet. For phones is on a VoIP trunk or FXS/FXO trunk, the PSAP gets a reorder tone.

Examples

The following example shows how to assign an ERL to a phone's dial peer:

```
dial-peer voice 12 pots
 emergency response location 18
```

The following example shows how to assign an ERL to a phone's ephone:

```
ephone  41
 emergency response location 22
```

The following example shows how to assign an ERL to one or more SCCP phones:

```
ephone-template 6
 emergency response location 8
```

The following example shows how to assign an ERL to a phone's voice register pool:

```
voice register pool 4
 emergency response location 21
```

The following example shows how to assign an ERL to one or more SIP phones:

```
voice register template 4
 emergency response location 8
```

Related Commands

Command	Description
emergency response callback	Defines a dial peer that is used for 911 callbacks from the PSAP.
emergency response zone	Defines a dial peer that is used by the system to route emergency calls to the PSAP.
voice emergency response location	Creates a tag for identifying an ERL for the enhanced 911 service.

emergency response zone

To define a dial peer that is used by the system to route emergency calls to a PSAP, use the emergency response zone command in voice dial-peer configuration mode. To remove the definition of the dial peer as an outgoing link to the PSAP, use the **no** form of this command.

emergency response zone *zone-tag*
no emergency response zone

Syntax Description

zone-tag	Identifier (1-100) for the emergency response zone.

Command Default

The dial peer is not defined as an outgoing link to the PSAP. Therefore, E911 services are not enabled.

Command Modes

Dial-peer configuration (config-dial-peer)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)T	Cisco Unified CME 4.1 Cisco Unified SRST 4.1 Cisco Unified SIP SRST 4.1	This command was introduced. For Cisco Unified CME, this command is supported in SRST fallback mode only.
12.4(15)XY	Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1) Cisco Unified SIP SRST 4.2(1)	The zone tag option was added. This command was added for Cisco Unified CME.

Usage Guidelines

Use this command to specify that any calls using this dial peer are processed by the E911 software. To enable any E911 processing, the emergency response zone command must be enabled under a dial peer.

If no zone tag is specified, the system looks for a matching ELIN to the E911 caller's phone by searching each *emergency response location* that was configured using the **emergency response location** command.

If a zone tag is specified, the system looks for a matching ELIN using sequential steps according to the contents of the configured zone. For example, if the E911 caller's phone has an explicit ERL assigned, the system first looks for that ERL in the zone. If not found, it then searches each location within the zone according to assigned priority numbers, and so on. If all steps fail to find a matching ELIN, the default ELIN is assigned to the E911 caller's phone. If no default ELIN is configured, the E911 caller's automatic number identification (ANI) number is communicated to the Public Safety Answering Point (PSAP).

This command can be defined in multiple dial peers. The zone tag option allows only ERLs defined in that zone to be routed on this dial peer. Also, this command allows callers dialing the same emergency number to be routed to different voice interfaces based on the zone that includes their ERL.

Examples

The following example shows a dial peer defined as an outgoing link to the PSAP. Emergency response zone 10 is created and only calls from this zone are routed through 1/0/0.

```
dial-peer voice 911 pots
 destination-pattern 9911
 prefix 911
 emergency response zone 10
```

```
port 1/0/0
```

Related Commands

Command	Description
emergency response callback	Defines a dial peer that is used for 911 callbacks from the PSAP.
emergency response location	Associates an ERL to either a SIP phone, ephone, or dial peer.
voice emergency response location	Creates a tag for identifying an ERL for E911 services.
voice emergency response zone	Creates an emergency response zone within which ERLs can be grouped.

encrypt password

To encrypt the password that is configured on Unified SRST, use the **encrypt password** command in **call-manager-fallback** configuration mode. To disable password encryption, use the **no** form of this command.

encrypt pasword
no encrypt password

Syntax Description

This command has no arguments or keywords.

Command Default

This command is enabled by default.

Command Modes

Call Manager Fallback configuration (config-cm-fallback)

Command History

Cisco IOS Release	Cisco Product	Modification
Cisco IOS XE Gibraltar 16.11.1a Release	Unified SRST 12.6	The command is introduced.

Usage Guidelines

The CLI command **encrypt password** is enabled by default on Unified SRST router. However, it is mandatory to configure **key config-key password-encrypt** *[Master key]* and **password encryption aes** along with **encrypt password** to support encryption on Unified SRST router.

Note If the key used to encrypt the password is replaced with a new key (replace key or re-key), then the password is re-encrypted with the new key.

Related Commands

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

ephone-type

To add a Cisco Unified IP phone type by defining an ephone-type template, use the **ephone-type** command in global configuration mode. To remove an ephone type, use the **no** form of this command.

ephone-type *phone-type* [**addon**]
no **ephone-type** *phone-type*

Syntax Description

phone-type	Unique label that identifies the type of phone. Value is any alphanumeric string up to 32 characters.
addon	(Optional) Phone type is an add-on module, such as the Cisco Unified IP Phone 7915 Expansion Module.

Command Default

Ephone type is not defined.

Command Modes

Global configuration (config)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)XZ	Cisco Unified CME 4.3 Cisco Unifieid SRST 4.3	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0 Cisco Unified SRST 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

This command adds a user-defined template for a phone type to a Cisco Unified CME system. This configuration template defines a set of attributes that describe the features of the new phone type. Use this command to add phone types that are not already defined in the system.

If you use a phone-type argument that matches a system-defined phone type, a message displays notifying you that the ephone-type is built-in and cannot be changed. For a list of system-defined phone types, see the **type** command.

Use the **create cnf-files** command for the new phone type to take effect.

Examples

The following example shows the Nokia E61 added with an ephone-type template, which is then assigned to ephone 2:

```
ephone-type  E61
 device-id 376
 device-name E61 Mobile Phone
 num-buttons 1
 max-presentation 1
 no utf8
 no phoneload
!
!
telephony-service
 load E61 SCCP61.8-2-2SR2S
```

```
 max-ephones 100
 max-dn 240
 ip source-address 15.7.0.1 port 2000
 cnf-file location flash:
 cnf-file perphone
 voicemail 8900
 max-conferences 8 gain -6
 transfer-system full-consult
 create cnf-files version-stamp 7960 Sep 25 2007 21:25:47
!
!
ephone  2
 mac-address 001C.821C.ED23
 type E61
 button  1:2
```

Related Commands

Command	Description
create cnf-files	Builds the eXtensible Markup Language (XML) configuration files that are required for IP phones.
device-id	Specifies the device ID for a phone type in an ephone-type template.
device-name	Assigns a name to a phone type in an ephone-type template.
load	Associates a type of Cisco Unified IP phone with a phone firmware file.
type	Assigns a phone type to an SCCP phone.

expiry

To set the time after which emergency callback history expires, use the **expiry** command in voice emergency response settings configuration mode. To remove the number, use the **no** form of this command.

expiry *time*
no expiry

Syntax Description

time	Identifier (2-2880) in minutes for the maximum time the 911 caller history is available for callback.

Command Default

The default expiry time is 180 minutes.

Command Modes

Voice emergency response settings configuration (cfg-emrgncy-resp-settings)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)XY	Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1) Cisco Unified SIP SRST 4.2(1)	This command was introduced.

Usage Guidelines

Use this command to specify the amount of time (in minutes) to keep emergency caller history for each ELIN. The time can be an integer in the range of 2 to 2880 minutes. The default value is 180 minutes.

Examples

In this example, the ELIN (4085550101) defined in the voice emergency response settings configuration will be used if the 911 caller's IP phone address does not match any of the voice emergency response locations. After the 911 call is placed to the PSAP, the PSAP has 120 minutes to call back 408 555-0101 to reach the 911 caller. If the call history has expired (after 120 minutes), any callback is routed to extension 7500.

```
voice emergency response settings
callback 7500
elin 4085550101
expiry 120
```

Related Commands

Command	Description
callback	Default phone number to contact if a 911 callback cannot find the last 911 caller from the ERL.
elin	E.164 number used as the default ELIN if no matching ERL to the 911 caller's IP phone address is found.
logging	Syslog informational message printed to the console every time an emergency call is made.
voice emergency response settings	Creates a tag for identifying settings for E911 behavior.

extension-range

To define a range of extension numbers for a specific MOH group in Cisco Unified CME or Cisco Unified SRST, use the **extension-range** command in voice-moh-group configuration mode. To define a range of extension numbers for a specific directory number in Cisco Unified CME, use the **extension-range** command in ephone-dn configuration mode. To disable the extension-range command, use the **no** form of this command.

extension-range *starting-extension to ending-extension*
no extension-range starting-extension to ending-extension

Syntax Description

starting-extension	Hexidecimal digits (0-9 or A-F) that define the starting extension number in an extension range. Maximum length: 32 digits.
ending-extension	Hexidecimal digits (0-9 or A-F) that define the last extension number in an extension range. Value of the ending extensing must be larger than value of the starting extension. Maximum length: 32 digits.

Command Default

No extension- range is configured.

Command Modes

Voice MOH group configuration (config-voice-moh-group)
Ephone-dn configuration (config-ephone-dn)

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0 Cisco Unified SRST 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0 Cisco Unified SRST 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

This command configured in voice moh-group configuration mode identifies MOH clients calling extension numbers specified under the extension range configured for a MOH group in Cisco Unified CME or Cisco Unified SRST. This command in ephone-dn configuration mode identifies MOH clients calling extension numbers specified under the extension range configured for a directory number in Cisco Unified CME

You can define multiple extension-ranges in the same MOH group or directory number.

The extension can be expressed in hexadecimal digits ranging from 0-9 or A-F and should not exceed the limit of 32 digits.

The starting-extension and ending-extension numbers must contain the same number of digits.

The ending extension number must be of a greater value than the starting extension number.

Extension-range for a MOH group must not overlap with any other extension-range configured in any other MOH group.

Note If an extension range is defined in a MOH group and it is also defined under ephone-dn, the extension range defined under ephone-dn takes precedence.

Examples

The following example shows two extension ranges configured under voice moh-group 1:

```
Router(config)# voice moh-group 1
Router(config-voice-moh-group)# moh flahs:/minuet.wav
Router(config-voice-moh-group)# description Marketing
Router(cconfig-voice-moh-group)# extension range 1000 to 1999
Router(config-voice-moh-group)# extension range 3000 to 3999
Router(config-voice-moh-group)#
```

Related Commands

Command	Description
moh	Enables music on hold from an audio file.
voice-moh-group	Enters voice moh-group configuration mode.

external-ring (voice register global)

To specify the type of ring sound used on Cisco Session Initiation Protocol (SIP) or Cisco SCCP IP phones for external calls, use the **external-ring** command in voice register global configuration mode. To return to the default ring sound, use the **no** form of this command.

external-ring {**bellcore-dr1** | **bellcore-dr2** | **bellcore-dr3** | **bellcore-dr4** | **bellcore-dr5**}
no external-ring

Syntax Description

bellcore-dr1 bellcore-dr2 bellcore-dr3 bellcore-dr4 bellcore-dr5	Each **bellcore-dr** keyword supports standard distinctive ringing patterns as defined in the standard GR-506-CORE, *LSSGR: Signaling for Analog Interfaces* .

Command Default

The default ring sound is an internal ring pattern.

Command Modes

Voice register global configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4 Cisco SIP SRST 3.4	This command was introduced.

Usage Guidelines

When set, this command defines varying ring tones so that you can discriminate between internal and external calls from Cisco SIP or Cisco SCCP IP phones.

Examples

The following example shows how to specify that Bellcore DR1 be used for external ringing on Cisco SIP IP phones:

```
Router(config)# voice register global
Router(config-register-global)# external-ring bellcore-dr1
```

Related Commands

Command	Description
voice register global	Enters voice register global configuration mode in order to set global parameters for all supported Cisco SIP phones in a Cisco Unified Communications Manager Express (Cisco Unified CME) or Cisco Unified SIP Survivable Remote Site Telephony (SRST) environment.

group phone

To add a phone, including a TAPI-based client application, or a softphone on a PC to a VRF group for Cisco Unified CME, use the **group phone** command in ephone or ephone-template configuration mode. To remove the configuration, use the **no** form of this command.

group phone *group-tag* [**tapi** *group-tag*]
no group phone

Syntax Description

group-tag	Unique identifier of VRF group ranges from 1 to 5.
tapi	(Optional) Add TAPI-based client on the phone being configured to a VRF group.

Command Default

By default, this feature is disabled.

Command Modes

Ephone configuration (config-ephone)

Ephone-template configuration (config-ephone-template)

Voice register pool

Voice register template

Command History

Cisco IOS Release	Cisco Products	Modification
12.4(22)T	Cisco Unified SRST 7.0(1)	This command was introduced.
15.4(3)M	Cisco Unified SRST 10.5	This command was modified.

Usage Guidelines

This command enables you to configure the voice VRF group for SIP phones.

This command adds a softphone on a PC, an IP phone, or a TAPI client on an IP phone to a VRF group.

VRF groups for users and phones in Cisco Unified CME are created by using the **group** command in telephony-service configuration mode.

All SCCP and SIP phones connected to Cisco Unified CME must register through the global voice VRF.

TAPI-based client on an IP phone and softphones on a PC must register in Cisco Unified CME through a data VRF.

Before you can use this command, the MAC address for the IP phone being configured must be configured by using the **mac-address** command in ephone configuration mode.

If you use an ephone template to apply a command to an ephone and you also use the same command in ephone configuration mode, the value that you set in ephone configuration mode has priority over the ephone-template configuration.

Examples

The following example shows four phones in three VRF groups, two on data VRFs and one on a global voice VRF.

```
telephony-service
 sdspfarm conference mute-on # mute-off #
 sdspfarm units 4
 sdspfarm transcode sessions 10
 sdspfarm tag 1 xcode101
 sdspfarm tag 2 conf103
 group  1
  ip source-address 209.165.201.1 port 2000
  url directories http://209.165.201.1/localdirectory
 !
 group  2 vrf data-vrf1
  ip source-address 209.165.201.2 port 2000
 !
group  3 vrf data-vrf2
  ip source-address 209.165.201.3 port 2000
 !
.
.
!
ephone-template 1
  group phone 1 tapi 2
ephone-template 2
  group phone 2
...
ephone 1
  mac-address 1111.2222.3333
  ephone-template 1
ephone 2
  mac-address 2222.2222.3333
  ephone-template 2
ephone 3
  mac-address 1111.3333.3333
  group phone 1 tapi 3
ephone 4
  mac-address 1111.2222.4444
  group phone 3
!
```

The following example shows four phones in three VRF groups, two on data VRFs and one on a global voice VRF.

```
Router(config)# voice register template
Router(config-telephony)# group-phone <group-tag>
```

Related Commands

Command	Description
voice register pool	Enters voice register pool configuration mode.
voice register template	Enters voice register template configuration mode.
ephone-template (ephone)	Applies an ephone template to an ephone configuration.
group (telephony-service)	Creates a VRF group for phones and users in Cisco Unified CME.
mac-address	Associates the MAC address of a Cisco IP phone with an ephone configuration.

h450 h450-2 timeout (voice service voip)

To specify timeout values for call transfers using the ITU-T H.450.2 standard, use the **h450 h450-2 timeout** command in H.323 voice service configuration mode. To return to the default values, use the **no** form of this command.

h450 h450-2 timeout {**T1** | **T2** | **T3** | **T4**} *milliseconds*
no h450 h450-2 timeout {**T1** | **T2** | **T3** | **T4**}

Syntax Description

T1	Timer for identification.
T2	Timer for call setup.
T3	Timer for response initiation.
T4	Timer for response setup.
milliseconds	Number of milliseconds. Range is from 500 to 60000.

Command Default

T1 timer is 2000 milliseconds. T2 timer is 5000 milliseconds. T3 timer is 5000 milliseconds. T4 timer is 5000 milliseconds.

Command Modes

H.323 voice service configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco SRST 3.0	This command was introduced.
12.3(4)T	Cisco SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Usage Guidelines

Use this command with Cisco SRST V3.0 or a later version.

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.

Examples

The following example defines a T1 timeout of 3000 milliseconds:

```
Router(config)# voice service voip
Router(conf-voi-serv)# h323
Router(conf-serv-h323)# h450 h450-2 timeout T1 3000
```

Related Commands

Command	Description
h323	Enables H.323 voice service configuration commands.
voice service	Enters voice-service configuration mode.

huntstop (call-manager-fallback)

To set the huntstop attribute for the dial peers associated with a Cisco Unified IP phone during Cisco Unified Communications Manager fallback, use the **huntstop** command in call-manager-fallback configuration mode. To disable huntstop, use the **no** form of this command.

huntstop [channel] *1-8*
no huntstop

Syntax Description

channel	(Optional) For dual-line ephone-dns, keeps incoming calls from hunting to the second channel if the first channel is busy or does not answer.
1-8	(Optional) For octo-line ephone-dns, keeps incoming calls from hunting to the next channel if the last allowable channel is busy or does not answer. The default is 8.

Command Default

Huntstop is enabled by default. For octo-line mode, the default number is 8.

Command Modes

Call-manager-fallback configuration (config-cm-fallback)

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco SRST 1.0	This command was introduced on the following platforms: Cisco 2600 series and Cisco 3600 series multiservice routers, and Cisco IAD2420 series IADs.
12.2(2)XT	Cisco SRST 2.0	This command was implemented on Cisco 1750 and Cisco 1751 multiservice routers.
12.2(8)T	Cisco SRST 2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers.
12.2(8)T1	Cisco SRST 2.0	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
12.2(11)T	Cisco SRST 2.01	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.
12.2(15)ZJ	Cisco SRST 3.0	The **channel** keyword was added.
12.3(4)T	Cisco SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.4(15)XZ	Cisco Unified SRST 4.3	The *1-8* argument was added for octo-line mode.
12.4(20)T	Cisco Unified CME 7.0 Cisco Unified SRST 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

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

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

Examples

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

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

The following example shows that octo-line is enabled, the maximum directory numbers is 12, and huntstop is limited at 6 channels:

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

Router(config-cm-fallback)# **max-dn 12 octo-line 6**

Related Commands

Command	Description
call-manager-fallback	Enables Cisco Unified SRST support and enters call-manager-fallback configuration mode.
huntstop (dial-peer)	Disables all further dial-peer hunting if a call fails using hunt groups.
max-dn (call- manager-fallback)	Sets the maximum possible number of virtual voice ports that can be supported by a router and activates dual-line mode, octo-line mode, or both modes.

id (voice register pool)

To explicitly identify a locally available individual Cisco SIP IP phone, or when running Cisco Unified Session Initiation Protocol (SIP) Survivable Remote Site Telephony (SRST), set of Cisco SIP IP phones, use the **id** command in voice register pool configuration mode. To remove local identification, use the **no** form of this command.

id [network *address* **mask** *mask* | *address* **mask** *mask* **] [ip** *address* **mask** *mask* *address* **mask** *mask* **] [mac** *address* **]**

[device-id-name *devicename* **]**

[**phone-number** *e164-number* | **extension-number** *extension-number*]

no id { **[network** *address* **mask** *mask* | *address* **mask** *mask* **]** | **[ip** *address* **mask** *mask* *address* **mask** *mask* **]** | **[mac** *address* **]** } **[device-id-name** *devicename* **]**

[**phone-number** *e164-number* | **extension-number** *extension-number*]

Syntax Description

network *address* **mask** *mask* | *address* **mask** *mask*	This keyword/argument combination is used to accept SIP Register messages for the indicated phone numbers from any IP phone within the specified IPv4 and IPv6 subnets. *ipv6 address* can only be configured with an IPv6 address or a dual-stack mode.
ip *address* **mask** *mask* | *address* **mask** *mask*	This keyword/argument combination is used to identify an individual phones IPv4 or IPv6 address. *ipv6 address* can only be configured with an IPv6 address or a dual-stack mode.
mac *address*	The **mac** *address* keyword/argument combination is used to identify the MAC address of a particular Cisco IP phone.
device-id-name *devicename*	Defines the device name to be used to download the phone's configuration file.
phone-number*e164-number*	Configures the phone-number in E.164 format for webex calling user.
extension-number*extension-number*	Configures extension number for Webex Calling users.

Command Default

No SIP IP phone is configured.

Command Modes

Voice register pool configuration (config-register-pool)

Command History

Release	Cisco Product	Modification
12.2(15)ZJ	Cisco SIP SRST 3.0	This command was introduced.
12.3(4)T	Cisco SIP SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4 Cisco SIP SRST 3.4	This command was added to Cisco CME.
15.3(3)T	Cisco Unified CME 10.0	This command was modified to add the **device-id-name** *devicename* keyword-argument combination.
Cisco IOS XE Everest 16.6.1	Unified SRST 12.0	This command was modified to add the following keyword-argument combinations for **network** and **ip** to include support for IPv6 address: *address* **mask** *mask*.
Cisco IOS XE Amsterdam 17.2.1r	Cisco Unified SIP SRST 12.8	Introduced support for YANG models.
Cisco IOS XE Cupertino 17.9.1	Cisco Unified SRST 14.3	This command was modified to add the following: **phone-number***e164-number* and **extension-number***extension-number*

Usage Guidelines

Configure this command before configuring any other command in voice register pool configuration mode.

This command allows explicit identification of an individual Cisco SIP IP phone to support a degree of authentication, which is required to accept registrations, based upon the following:

- Verification of the local Layer 2 MAC address using the router's Address Resolution Protocol (ARP) cache.
- Verification of the known single static IP address (or DHCP dynamic IP address within a specific subnet) of the Cisco SIP IP phone.

When the **mac** *address* keyword and argument are used, the IP phone must be in the same subnet as that of the router's LAN interface, such that the phone's MAC address is visible in the router's ARP cache. Once a MAC address is configured for a specific voice register pool, remove the existing MAC address before changing to a new MAC address.

Note For Cisco Unified SIP SRST, this command also allows explicit identification of locally available set of Cisco SIP IP phones.

Examples

The following is partial sample output from the **show running-config** command. The **id** command identifies the MAC address of a particular Cisco IP phone. The output shows that voice register pool 1 has been set up to accept SIP Register messages from a specific IP phone through the use of the **id** command.

```
voice register pool 1
 id mac 0030.94C2.A22A
 preference 5
 cor incoming call91 1 91011
 translate-outgoing called 1
 proxy 10.2.161.187 preference 1 monitor probe icmp-ping
```

```
 alias 1 94... to 91011 preference 8
 voice-class codec 1
```

The following is sample output from the **show** command after configuring IPv6 address on Cisco Unified SRST router.

```
voice register pool 1
 id network 2001:420:54FF:13::312:0/117
```

The following is sample output from the **show phone-number***e164-number* .

```
voice register pool 10
 id phone-number +15139413701
 dtmf-relay rtp-nte
 voice-class codec 10
```

The following is sample output from the **show** and **extension-number***extension-number*.

```
voice register pool 10
 id extension-number 3701
 dtmf-relay rtp-nte
 voice-class codec 10
```

Related Commands

Command	Description
mode (voice register global)	Enables the mode for provisioning SIP phones in a Cisco Unified CallManager Express (Cisco Unified CME) system.

incoming called-number (voice register pool)

To apply incoming called-number parameters to dynamically created dial peers, use the **incoming called-number** command in voice register pool configuration mode. To remove incoming called-number parameters from a dial peer, use the **no** form of this command.

incoming called-number [*numbe* **r**]
no incoming called-number

Syntax Description

number	(Optional) Sequence of digits that represent a phone number prefix.

Command Default

None

Command Modes

Voice register pool configuration

Command History

Release	Cisco Product	Modification
12.2(15)ZJ	Cisco SIP SRST 3.0	This command was introduced.
12.3(4)T	Cisco SIP SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Usage Guidelines

The **id** (voice register pool) command must be configured before any other voice register pool commands, including the **incoming called-number** command. The **id** command identifies a locally available individual Cisco SIP IP phone or a set of Cisco SIP IP phones.

Examples

The following is partial sample output from the **show running-config** command that applies the prefix 308 to dynamically created dial peers:

```
voice register pool 1
 application SIP.app
 incoming called-number 308
 voice-class codec 1
```

Related Commands

Command	Description
id (voice register pool)	Explicitly identifies a locally available individual Cisco SIP IP phone or set of Cisco SIP IP phones.
incoming called-number (dial-peer)	Specifies an incoming called number of an MMoIP or POTS dial peer.
show dial-peer voice	Displays information for voice dial peers.
voice register pool	Enables SIP SRST voice register pool configuration commands.

ip qos dscp (call-manager-fallback)

To set the Differentiated Services Code Point (DSCP) for marking the quality of service (QoS) requirements for each packet, use the **ip qos dscp** command in call-manager-fallback configuration mode. To reset to the default value, use the **no** form of this command.

ip qos dscp {*numberafcs* | **default** | **ef**} {**media** | **service** | **signaling** | **video**}
no ip qos dscp {*numberafcs* | **default** | **ef**} {**media** | **service** | **signaling** | **video**}

Syntax Description

number	DSCP value. Range: 0 to 63.
af	Sets DSCP to assured forwarding bit pattern. • **af11**—bit pattern 001010 • **af12**—bit pattern 001100 • **af13**— bit pattern 001110 • **af21**— bit pattern 010010 • **af22**— bit pattern 010100 • **af23**— bit pattern 010110 • **af31**— bit pattern 011010 • **af32**— bit pattern 011100 • **af33**— bit pattern 011110 • **af41**—bit pattern 100010 • **af42**—bit pattern 100100 • **af43**—bit pattern 100110
cs	Sets DSCP to class-selector codepoint. • **cs1**—codepoint 1 (precedence 1) • **cs2**—codepoint 2 (precedence 2) • **cs3**—codepoint 3 (precedence 3) • **cs4**—codepoint 4 (precedence 4) • **cs5**—codepoint 5 (precedence 5) • **cs6**—codepoint 6 (precedence 6) • **cs7**—codepoint 7 (precedence 7)
default	Sets DSCP to default bit pattern of 000000.
ef	Sets DSCP to expedited forwarding bit pattern 101110.
media	Applies DSCP to media payload packets for SCCP phones.
service	Not supported for Cisco Unified SRST.
signaling	Not supported for Cisco Unified SRST.
video	Not supported for Cisco Unified SRST.

Command Default

DSCP for media is **ef**.

Command Modes

Call-manager-fallback configuration (config-cm-fallback)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(22)YB	Cisco Unified SRST 7.1	This command was introduced.
12.4(24)T	Cisco Unified SRST 7.1	This command was integrated into Cisco IOS Release 12.4(24)T.

Usage Guidelines

This command allows you to set different priority levels for different types of network traffic sent by the Cisco Unified SRST router. Differentiated Services is a method of prioritizing specific network traffic based on the QoS specified by each packet.

Because Cisco Unified IP Phones get their DSCP information from the configuration file that is downloaded to the device, and Cisco Unified SRST does not download configuration files to phones during fallback, only the value set with the **media** keyword takes effect for SCCP phones.

If the DSCP is configured for the gateway interface using the **service-policy** command or in the dial peer using the **ip qos dscp** command, the value set with those commands takes precedence over the DSCP value configured with this command.

Examples

The following example shows DSCP set for different types of packets .

```
call-manager-fallback
 ip qos dscp cs2 media
 ip qos dscp cs1 signal
 ip qos dscp cs3 video
 ip qos dscp cs4 service
```

Related Commands

Command	Description
ip qos dscp	Sets the DSCP for QoS in a dial peer.
service-policy	Assigns a policy map to an interface that will be used as the service policy for the interface.

ip source-address (call-manager-fallback)

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

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

Syntax Description

ip-address	The preexisting SRST router IP address, typically one of the addresses of the Ethernet port of the local router.
port	(Optional) The port to which the gateway router connects to receive messages from the Cisco IP phones.
port	(Optional) The port number. The range is from 2000 to 9999. The default is 2000.
any-match	(Optional) Disables strict IP address checking for registration. This is the default.
strict-match	(Optional) Requires strict IP address checking for registration.

Command Default

Default port number: 2000 Default server address match: **any-match**

Command Modes

Call-manager-fallback configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco SRST 1.0	This command was introduced on the following platforms: Cisco 2600 series and Cisco 3600 series multiservice routers, and Cisco IAD2420 series IADs.
12.2(2)XT	Cisco SRST 2.0	This command was implemented on Cisco 1750 and Cisco 1751 multiservice routers.
12.2(8)T	Cisco SRST 2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers.
12.2(8)T1	Cisco SRST 2.0	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
12.2(11)T	Cisco SRST 2.01	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.

Usage Guidelines

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

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

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

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

Examples

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

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

Related Commands

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

ip source-address (credentials)

To enable the Cisco Unified CME or SRST router to receive credential service messages through the specified IP address and port, use the **ip source-address** command in credentials configuration mode. To disable the router from receiving messages, use the **no** form of this command.

ip source-address *ip-address* [**port** [*port*]]
no ip source-address

Syntax Description

ip-address	Router IP address, typically one of the addresses of the Ethernet port of the local router.
port *port*	(Optional) TCP port for credentials service communication. Range is from 2000 to 9999. Cisco Unified CME default is 2444. SRST default is 2445.

Command Default

Cisco Unified CME default port number: 2444 Cisco Unified SRST default port number: 2445

Command Modes

Credentials configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(14)T	Cisco SRST 3.3	This command was introduced for Cisco Unified SRST.
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced for Cisco Unified CME.

Usage Guidelines

Cisco Unified CME

This command is used with Cisco Unified CME phone authentication to identify a Cisco Unified CME router on which a CTL provider is being configured.

Cisco Unified SRST

The **ip source-address** command is a mandatory command to enable secure SRST. If the port number is not provided, the default value (2445) is used. The IP address is usually the IP address of the secure SRST router.

Examples

Cisco Unified CME

The following example creates a CTL provider on a Cisco Unified CME router that is not running the CTL client.

```
Router(config)# credentials
Router(config-credentials)# ip source-address 172.19.245.1 port 2444
Router(config-credentials)# trustpoint ctlpv
Router(config-credentials)# ctl-service admin user4 secret 0 c89L8o
```

Cisco Unified SRST

The following example enters credentials configuration mode and sets the IP source address and port:

```
Router(config)# credentials
Router(config-credentials)# ip source-address 10.6.21.4 port 2445
```

Related Commands

Command	Description
credentials	Enters credentials configuration mode to configure a Cisco Unified CME CTL provider certificate or an SRST router certificate.
ctl-service admin	Specifies a user name and password to authenticate the CTL client during the CTL protocol.
debug credentials	Sets debugging on the credentials service that runs between a Cisco Unified CME CTL provider and the CTL client or between an SRST router and Cisco Unified Communications Manager.
show credentials	Displays the credentials settings on a Cisco Unified CME or SRST router.
trustpoint (credentials)	Specifies the name of the trustpoint to be associated with a Cisco Unified CME CTL provider certificate or with an SRST router certificate.

keepalive (call-manager-fallback)

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

keepalive *seconds*
no keepalive *seconds*

Syntax Description

seconds	Keepalive message transmission interval, in seconds. The valid range is from 10 to 65535. The default timeout value is 30.

Command Default

30 seconds

Command Modes

Call-manager-fallback configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco SRST 1.0	This command was introduced on the following platforms: Cisco 2600 series and Cisco 3600 series multiservice routers; and Cisco IAD2420 series IADs.
12.2(2)XT	Cisco SRST 2.0	This command was implemented on Cisco 1750 and Cisco 1751 multiservice routers.
12.2(8)T	Cisco SRST 2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers.
12.2(8)T1	Cisco SRST 2.0	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
12.2(11)T	Cisco SRST 2.01	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.

Usage Guidelines

The **keepalive** command configures the time interval between successive keepalive messages from Cisco IP phones to the Cisco Unified SRST router. If the router fails to receive three successive keepalive messages, it considers the Cisco IP phone to be out of service until the phone reregisters.

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

Examples

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

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

Related Commands

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

limit-dn (call-manager-fallback)

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

limit-dn phone-type *max-lines*
no limit-dn phone-type

Syntax Description

phone-type	Type of phone. The following phone types are predefined in the system:
12SP	12SP+ and 30VIP phones
6901	**Cisco Unified IP Phone 6901**
6911	**Cisco Unified IP Phone 6911**
6921	**Cisco Unified IP Phone 6921**
6941	Cisco Unified IP Phone 6941
6961	Cisco Unified IP Phone 6961
7902	Cisco Unified IP Phone 7902
7905	Cisco Unified IP Phone 7905
7906	Cisco Unified IP Phone 7906
7910	Cisco Unified IP Phone 7910
7911	Cisco Unified IP Phone 7911
7912	Cisco Unified IP Phone 7912
7920	Cisco Unified IP Phone 7920
7921	Cisco Unified IP Phone 7921
7925	Cisco Unified IP Phone 7925
7926	Cisco Unified IP Phone 7926
7931	Cisco Unified IP Phone 7931
7935	Cisco Unified IP Phone 7935
7936	Cisco Unified IP Phone 7936
7937	Cisco Unified IP Phone 7937
7940	Cisco Unified IP Phone 7940
7941	Cisco Unified IP Phone 7941

7941GE	Cisco Unified IP Phone 7941GE
7942	Cisco Unified IP Phone 7942
7945	Cisco Unified IP Phone 7945
7960	Cisco Unified IP Phone 7960
7961	Cisco Unified IP Phone 7961
7961GE	Cisco Unified IP Phone 7961GE
7962	Cisco Unified IP Phone 7962
7965	Cisco Unified IP Phone 7965
7970	Cisco Unified IP Phone 7970
7971	Cisco Unified IP Phone 7971
7975	Cisco Unified IP Phone 7975
7985	Cisco Unified IP Phone 7985
CIPC	CIPC
IP-STE	IP-STE
anl	anl
ata	ata
bri	bri
vgc-phone	vgc-phone
max-lines	Maximum lines setting. The range is from 1 to 34.

Command Default

No default behavior or values.

Command Modes

Call-manager-fallback configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(2)XT	Cisco SRST 2.0	This command was introduced on the following platforms: Cisco 1750, Cisco 1751, Cisco 2600 series and Cisco 3600 series multiservice routers, and Cisco IAD2420 series IADs.
12.2(8)T	Cisco SRST 2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers.
12.2(8)T1	Cisco SRST 2.0	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.

Cisco IOS Release	Cisco Product	Modification
12.2(11)T	Cisco SRST 2.01	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.

Usage Guidelines

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

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

The range of the maximum number of lines is from 1 to 34. If there is any active phone with a number greater than the specified limit, warning information is displayed for phone reset.

Examples

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

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

Related Commands

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

listen-port (SIP)

To configure the listen ports used for SIP protocols, use the **listen-port** command in **voice service voip/sip** configuration mode. To reset port use to its default value, use the **no** form of this command.

listen-port [**non-secure** | **secure** | **secure no-client-validation**] *port-number*
no listen-port [**non-secure** | **secure** | **secure no-client-validation**]

Syntax Description

secure	Specifies the TLS port value.
non-secure	Specifies the TCP/UDP port value.
no-client-validation	Disables mTLS and available only for secure ports.
port-number	Port number. Range: 1–65535. The default for UDP/TCP is 5060; the default for TLS is 5061.

Command Default

The port number is set to the default value based on the transport layer protocol used.

Command Modes

SIP configuration mode (config-serv-sip)

Command History

Release	Modification
12.4(15)XY	This command was introduced.
12.4(20)T	This command was integrated into Cisco IOS Release 12.4(20)T.
Cisco IOS XE Cupertino 17.8.1a	Enhanced **listen-port secure** command to disable validation of client certificate and configure TLS port for SIP OAuth-based registrations from remote SIP endpoints.

Usage Guidelines

The **listen-port** command is configurable on incoming SIP calls, and is applicable for both TDM-IP gateway and CUBE (IPIPGW). The CUBE gateway port number defined in global configuration will be used for incoming call legs. Before configuring the SIP listen port for TCP/UDP/TLS, the SIP service should be shut down using the **shutdown** command in SIP configuration mode. If the SIP service is not shut down, the **listen-port** command flashes an error message saying "shutdown SIP service before changing SIP listen port". This ensures that there are no active calls when the SIP listen port is changed. The **non-secure** keyword is supported with all IOS images, and both the **secure** and **non-secure** keywords are supported with licensed Crypto images.

From Cisco IOS XE Cupertino 17.8.1a onwards, **listen-port secure** is modified to create a different TLS port to listen to SIP OAuth registration and disable validation of client certificate.

The following restrictions apply:

- Configuring the SIP listen port on a dial-peer basis is not supported.
- Configuring same listening port for both UDP/TCP and TLS is not allowed.

- Configuring the SIP listen port to a port that is already in use is not supported and results in an error message.
- Changing SIP listen port when Transport services (TCP/UDP/TLS) are shut down, will not close or reopen the port. The result is that only the new port number is updated. The new port will be bound when Transport services (TCP/UDP/TLS) is enabled.

Examples

The following example shows the port number on a Crypto image being changed to port 2000:

```
Router(config-serv-sip)# listen-port secure 10000
```

The following example shows the port number being reset to the TLS default port:

```
Router(config-serv-sip)# no listen-port
```

The following example shows **listen-port secure** that is modified to create a different TLS port to listen to SIP OAuth registration and disable validation of client certificate:

```
Router(config)#voice service voip
Router(conf-voi-serv)#sip
Router(conf-serv-sip)#listen-port ?
  non-secure  Change UDP/TCP SIP listen Port
  secure      Change TLS SIP listen Port

Router(conf-serv-sip)# listen-port secure ?
<0-65535>             Listen port for mTLS service
no-client-validation  TLS service without client validation

Router(conf-serv-sip)# listen-port secure no-client-validation ?
<cr>           Use default port 5090
<1024-49151>   Specify TLS listen-port
Router(conf-serv-sip)#
```

Related Commands

Command	Description
shutdown	Disables the port.

location (voice emergency response zone)

To include a location within an emergency response zone, use the **location** command in voice emergency response zone mode. To assign specific priorites to the locations, use the priority tag. To remove the location, use the **no** form of this command.

location *location-tag* **[priority <1-100>]**
no location *location-tag*

Syntax Description

location-tag	Identifier for the emergency response zone location.
priority *1-100*	Identifier (1-100) for the priority ranking of locations, 1 being the highest priority.

Command Modes

Voice emergency response zone configuration (cfg-emrgncy-resp-zone)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)XY	Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1) Cisco Unified SIP SRST 4.2(1)	This command was introduced.

Usage Guidelines

This command creates locations within emergency response zones. The tag must be the same as the tag that is defined using the **voice emergency response location** command. This allows routing of 911 calls to different PSAPs. Priority is optional and allows searching locations in a specified priority order. If there are locations with assigned priorities and locations configured without priorities, the prioritized locations are searched before those without an assigned priority.

Examples

The following example shows an assignment of ERLs to two zones, 10 and 11, to route callers to two different PSAPs. The locations for ERLs in zone 10 are searched in sequential order for a phone address match. The calls from zone 10 have an ELIN from ERLs 8, 9, and 10. The calls from zone 11 have an ELIN from ERLs 2, 3, 4, and 5. The locations for ERLs in zone 11 have priorities assigned and is searched in order of the assigned priority and not the ERL tag number.

```
voice emergency response zone 10
location 8
location 9
location 10
voice emergency response zone 11
location 5 priority 1
location 3 priority 2
location 4 priority 3
location 2 priority 10
```

Related Commands

Command	Description
emergency response callback	Defines a dial peer that is used for 911 callbacks from the PSAP.
emergency response location	Associates an ERL to either a SIP phone, ephone, or dial peer.

Command	Description
voice emergency response location	Creates a tag for identifying an ERL for the enhanced 911 service.
voice emergency response zone	Creates an emergency response zone within which ERLs can be grouped.

logging (voice emergency response settings)

To enable sylog messages to capture emergency call data, use the **logging** command in voice emergency response settings configuration mode. To disable logging, use the **no** form of this command.

logging
no logging

Syntax Description

This command has no arguments or keywords.

Command Default

This command is enabled by default.

Command Modes

Voice emergency response settings configuration (cfg-emrgncy-resp-settings)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)XY	Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1) Cisco Unified SIP SRST 4.2(1)	This command was introduced.

Usage Guidelines

This command enables syslog messages to be announced for every 911 emergency call that is made. The syslog messages can be used by third party applications to send pager or e-mail notifications to an in-house support number. This command is optional and is enabled by default.

Examples

In this example, the ELIN (4085550101) defined in the voice emergency response settings configuration is used if the 911 caller's IP phones address does not match any of the voice emergency response locations. After the 911 call is placed to the PSAP, the PSAP has 120 minutes to call back 408 555-0101 to reach the 911 caller. If the call history has expired (after 120 minutes), any callback is routed to extension 7500. The outbound 911 calls do not emit a syslog message to the logging facility (for example, a local buffer, console, or remote host).

```
voice emergency response settings
callback 7500
elin 4085550101
expiry 120
no logging
```

Related Commands

Command	Description
callback	Default phone number to contact if a 911 callback cannot find the last 911 caller from the ERL.
elin	E.164 number used as the default ELIN if no matching ERL to the 911 caller's IP phone address is found.
expiry	Number of minutes a 911 call is associated to an ELIN in case of a callback from the 911 operator.

Command	Description
voice emergency response settings	Creates a tag for identifying settings for E911 behavior.

max registrations (voice register pool)

> **Note** Effective with Cisco IOS Release 12.4(4)T, the **max registrations** command is not visible in Cisco IOS software. For similar functionality, use the **maximum bit-rate (cm-fallback-video)** command.

To set the maximum number of registrations accepted by the voice register pool, use the **max registrations** command in voice register pool configuration mode. To disable registration setup, use the **no** form of this command.

max registrations *value*
no max registrations

Syntax Description

value	Digit, beginning with 0, that represents the maximum number of registrations. The maximum registration value is platform dependent.

Command Default

The maximum number of IP phones that can be configured per platform

Command Modes

Voice register pool configuration

Command History

Release	Modification
12.2(15)ZJ	This command was introduced.
12.3(4)T	This command was integrated into Cisco IOS Release 12.3(4)T.
12.4(4)T	This command was removed.

Usage Guidelines

The **id** (voice register pool) command must be configured before any other voice register pool commands, including the **max registrations** command. The **id** command identifies a locally available individual Cisco SIP IP phone or sets of Cisco SIP IP phones.

If two phones attempt to register the same phone number, only the first phone can register the number. You can control which phone is accepted by using multiple voice register pools. In general, the best usage is one pool per phone; with multiple pools, some flexibility is granted.

Examples

The following partial sample output from the **show running-config** command shows that 5 is the maximum number of SIP telephone registrations accepted.

```
voice register pool 3
 id network 10.2.161.0 mask 255.255.255.0
 number 1 95... preference 1
 cor outgoing call95 1 95011
 max registrations 5
 voice-class codec 1
```

Related Commands

Command	Description
id (voice register pool)	Explicitly identifies a locally available individual Cisco SIP IP phone or set of Cisco SIP IP phones.
voice register pool	Enables SIP SRST voice register pool configuration commands.

max-conferences (call-manager-fallback)

To set the maximum number of simultaneous three-party conferences supported by the router, use the **max-conferences** command in call-manager-fallback configuration mode. To return to the default number of conferences, use the **no** form of this command.

max-conferences *max-no-of-conferences* **[gain -6 |0 |3 | 6]**
no max-conferences *max-no-of-conferences*

Syntax Description

max-no-of-conferences	The maximum number of simultaneous three-party conferences allowed by the router. The maximum number of three-party conferences is platform dependent: • Cisco 1751 router—8 • Cisco 1760 router—8 • Cisco 2600 series routers—8 • Cisco 2600-XM series routers—8 • Cisco 2801 router—8 • Cisco 2811, Cisco 2821, and Cisco 2851 routers—16 • Cisco 3640 and Cisco 3640A routers—8 • Cisco 3660 router—16 • Cisco 3725 router—16 • Cisco 3745 router—16 • Cisco 3800 series router—24
gain	(Optional) Increases the sound volume of VoIP and public switched telephony network (PSTN) parties joining a conference call. The allowable decibel units are -6 db, 0 db, 3 db, and 6 db. The default is -6 db.

Command Default

The default is half of the number of maximum simultaneous three-party conferences allowed per platform.

Command Modes

Call-manager-fallback configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco SRST 3.0	This command was introduced.
12.3(4)T	Cisco SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.3(8)T4	Cisco SRST 3.1	The Cisco 2800 series routers were added.
12.3(11)T	Cisco SRST 3.2	The Cisco 3800 series routers were added.
12.3(11)XL	Cisco SRST 3.2.2	The **gain** keyword was added.

Usage Guidelines

The **max-conferences** command supports three-party conferences for local and on-net calls only when all conference participants are using the G.711 codec. Conversion between G.711 u-law and a-law is supported. Mixing of the media streams is supported by the Cisco IOS processor. The maximum number of simultaneous conferences is limited to the platform-specific maximum.

The **gain** keyword's functionality is applied to inbound audio packets, so conference participants can more clearly hear a remote PSTN or VoIP caller joining their call. Note that this functionality cannot discriminate between a remote VoIP/foreign exchange office (FXO) source, which requires a volume gain, and a remote VoIP/IP phone, which does not require a volume gain and may therefore incur some sound distortions.

Examples

The following example sets the maximum number of conferences for a Cisco IP phone to 4 and configures a gain of 6 db for inbound audio packets from remote PSTN or VoIP calls joining a conference:

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# max-conferences
4 gain 6
```

Related Commands

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

max-dn (call-manager-fallback)

To set the maximum possible number of directories or virtual voice ports that can be supported by a router and to activate dual-line mode, octo-line mode, or both modes, use the **max-dn** command in call-manager-fallback configuration mode. To return to the default number of directories or virtual voice ports and to deactivate the dual-line mode or octo-line mode, use the **no** form of this command.

max-dn *max-no-of-directories* [{**dual-line** | **octo-line**}] [**preference** *preference-order*] [*number* **octo-line**]
no max-dn

Syntax Description

max-no-of-directories	Maximum number of directory numbers (dns) or virtual voice ports supported by the router. The maximum possible number is platform-dependent; see the Cisco IOS command-line interface (CLI) help. The default is 0 directory numbers and 1 channel per virtual port.
dual-line	(Optional) Sets all Cisco Unified IP phones connected to a Cisco Unified SRST router to one virtual voice port with two channels.
octo-line	(Optional) Sets all Cisco Unified IP phones connected to a Cisco Unified SRST router to one virtual voice port with eight channels.
preference *preference-order*	(Optional) Sets the global preference for creating the VoIP dial peers for all directory numbers that are associated with the primary number. Range is from 0 to 10. Default is 0, which is the highest preference.
number **octo-line**	(Optional) Sets the maximum number of octo-line directory numbers.

Command Default

0 directory numbers, 1 channel per virtual port, 8 channels per virtual port, 0 preference, 8 numbers

Command Modes

Call-manager-fallback configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco SRST 1.0	This command was introduced on the following platforms: Cisco 2600 series and Cisco 3600 series multiservice routers, and Cisco IAD2420 series IADs.
12.2(2)XT	Cisco SRST 2.0	This command was implemented on Cisco 1750 and Cisco 1751 multiservice routers.
12.2(8)T	Cisco SRST 2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers.
12.2(8)T1	Cisco SRST 2.0	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
12.2(11)T	Cisco SRST 2.01	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco SRST 3.0	The **dual-line** keyword was added.
12.3(4)T	Cisco SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.3(11)T	Cisco SRST 3.2	The **preference** keyword was added.
12.4(15)XZ	Cisco Unified SRST 4.3	The **octo-line** keyword and its *number* argument were added.
12.4(20)T	Cisco Unified SRST 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

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

The **dual-line** keyword facilitates call waiting, call transfer, and conference functions by allowing two calls to occur on one line simultaneously. In dual-line mode, all Cisco Unified IP phones on the Cisco Unified SRST router support two channels per virtual voice port.

The **octo-line** keyword facilitates call waiting, call transfer, and conference functions by allowing eight calls to occur on one line simultaneously. In octo-line mode, all Cisco Unified IP phones on the Cisco Unified SRST router support eight channels per virtual voice port.

Note After you specify the maximum number of available directory numbers, you cannot reduce that number of directory numbers or virtual voice ports without rebooting the router.

During Cisco Unified SRST registration, a dial peer is created and that dial peer includes a default preference. The **preference** keyword allows you to change the default value, if desired.

Setting the preference enables the desired dial peer to be selected when multiple dial peers within a hunt group are matched for a dial string.

The **alias** command also has a **preference** keyword that sets **alias** command preference values. Setting the **alias** command **preference** keyword allows the default preference set with the **max-dn** command to be overridden. When configuring call rerouting with the **alias** command, set the **preference** keyword of the **max-dn** command to a higher numeric preference value than the preference set with the **alias** command.

Examples

The following example sets the maximum number of directory numbers or virtual voice ports to 12, activates dual-line mode, activates octo-line mode, and sets the maximum number of dns for octo-mode to 6:

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# max-dn 12 dual-line preference 1 octo-line 6
```

Related Commands

Command	Description
alias (call-manager- fallback)	Provides a mechanism for rerouting calls to telephone numbers that are unavailable during Cisco Unified Communications Manager fallback.

Command	Description
call-forward busy (call-manager-fallback)	Configures call forwarding to another number when a Cisco Unified IP phone is busy.
call-forward noan (call-manager-fallback)	Configures call forwarding to another number when no answer is received from a Cisco Unified IP phone.
call-manager-fallback	Enables Cisco Unified SRST feature support and enters call-manager-fallback configuration mode.
huntstop (call-manager- fallback)	Sets the huntstop attribute for the dial peers associated with a Cisco Unified IP phone during Cisco Unified Communications Manager fallback.
huntstop chan num (call-manager-fallback)	Sets the huntstop attribute for the dial peers, for dual-line mode or octo-line mode (or both), associated with a Cisco Unified IP phone during Cisco Unified Communications Manager fallback.
max-conferences (call-manager-fallback)	Sets the maximum number of simultaneous three-party conferences supported by the router.
preference (dial-peer)	Indicates the preferred order of a dial peer within a hunt group.
transfer-pattern	Allows Cisco Unified IP phones to transfer telephone calls from callers outside the local IP network to another Cisco Unified IP phone.
transfer-system (call-manager-fallback)	Specifies the call transfer method for all Cisco Unified IP phones on a Cisco Unified SRST router using the ITU-T H.450.2 standard.

max-dn (voice register global)

To set the maximum number of SIP phone directory numbers (extensions) that are supported by a Cisco router, use the **max-dn** command in voice register global configuration mode. To reset to the default, use the **no** form of this command.

max-dn *max-directory-numbers*
no max-dn

Syntax Description

maxdirectorynumbers	Maximum number of extensions (ephone-dns) supported by the Cisco router. The maximum number is version and platform dependent; type **?** to display range. • In Cisco CME 3.4 to Cisco Unified CME 7.0 and in Cisco SIP SRST 3.4 to Cisco Unified SIP SRST 7.0: Default is maximum number supported by platform. • In Cisco Unified CME 7.1 and Cisco Unified SIP SRST 7.1 and later versions: Default is 0.

Command Default

Before Cisco Unified CME 7.1 and Cisco Unified SIP SRST 7.1, default is maximum number supported by platform.

In Cisco Unified CME 7.1 and Cisco Unified SIP SRST 7.1 and later versions, default is 0.

Command Modes

Voice register global configuration (config-register-global)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4 Cisco SIP SRST 3.4	This command was introduced.
12.4(22)YB	Cisco Unified CME 7.1 Cisco Unified SIP SRST 7.1	The default value was changed to 0.
12.4(24)T	Cisco Unified CME 7.1 Cisco Unified SIP SRST 7.1	This command was integrated into Cisco IOS release 12.4(24)T.
Cisco IOS XE Amsterdam 17.2.1r	Cisco Unified SIP SRST 12.8	Introduced support for YANG models.

Usage Guidelines

This command limits the number of SIP phone directory numbers (extensions) available in a Cisco Unified CME system. The **max-dn** command is platform specific. It defines the limit for the **voice register dn** command. The **max-pool** command similarly limits the number of SIP phones in a Cisco CME system.

You can increase the number of allowable extensions to the maximum; but after the maximum allowable number is configured, you cannot reduce the limit without rebooting the router. You cannot reduce the number of allowable extensions without removing the already-configured directory numbers with dn-tags that have a higher number than the maximum number to be configured.

Note This command can also be used for Cisco Unified SIP SRST.

Examples

The following example shows how to set the maximum number of directory numbers to 48:

```
Router(config)# voice register global
Router(config-register-global)# max-dn 48
```

Related Commands

Command	Description
voice register dn	Enters voice register dn configuration mode to define an extension for a SIP phone line.
max-pool (voice register global)	Sets the maximum number of SIP voice register pools that are supported in a Cisco SIP SRST or Cisco CME environment.

max-pool (voice register global)

To set the maximum number of Session Initiation Protocol (SIP) voice register pools that are supported in Cisco Unified SIP SRST, use the **max-pool** command in voice register global configuration mode (**voice register global**). To reset the maximum number to the default, use the **no** form of this command.

max-pool *max-voice-register-pools*
no **max-pool**

Syntax Description

max-voice-register-pools	Maximum number of SIP voice register pools supported by the Cisco router. The upper limit of voice register pools is platform-dependent; type **?** for range.

Command Modes

Voice register global configuration (config-register-global)

Command History

Cisco IOS Release	Modification
Cisco IOS XE Release 17.2.1v	Command qualified for use in Cisco vManage CLI templates.
Cisco IOS XE Amsterdam 17.2.1r	Introduced support for YANG models.

Usage Guidelines

This command limits the number of SIP phones that are supported by Cisco Unified SIP SRST. The **max-pool** command is platform-specific and defines the limit for the **voice register pool** command.

The **max-dn** command similarly limits the number of directory numbers (extensions) in Cisco Unified SIP SRST.

Examples

```
voice register global
  max-dn   200
  max-pool 100
  system message "SRST mode"
```

Related Commands

Command	Description
voice register dn	Enters voice register dn configuration mode to define an extension for a SIP phone line.
max-dn (voice register global)	Sets the maximum number of SIP voice register pools that are supported in a Cisco SIP SRST or Cisco CME environment.

max-ephones (call-manager-fallback)

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

max-ephones *max-no-of-phones*
no max-ephones

Syntax Description

max-no-of-phones	Maximum number of Cisco IP phones supported by the router. The maximum possible number is platform-dependent; refer to Cisco IOS command-line interface (CLI) help. The default is 0.

Command Default

0 Cisco IP phones

Command Modes

Call-manager-fallback configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco SRST 1.0	This command was introduced on the following platforms: Cisco 2600 series and Cisco 3600 series multiservice routers, and Cisco IAD2420 series IADs.
12.2(2)XT	Cisco SRST 2.0	This command was implemented on Cisco 1750 and Cisco 1751 multiservice routers.
12.2(8)T	Cisco SRST 2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers.
12.2(8)T1	Cisco SRST 2.0	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
12.2(11)T	Cisco SRST 2.01	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.

Usage Guidelines

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

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

Examples

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

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

Related Commands

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

maximum bit-rate (cm-fallback-video)

To set the maximum IP phone video bandwidth, use the **maximum bit-rate** command in call-manager-fallback video configuration mode. To restore the default maximum bit-rate, use the **no** form of this command.

maximum bit-rate *value*
no maximum bit-rate

Syntax Description

value	Sets the maximum IP phone video bandwidth, in kbps. The range is 0 to 10000000. The default value is 10000000.

Command Default

Maximum bit-rate is 1000000.

Command Modes

Call-manager-fallback video configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified SRST 4.0	This command was introduced.

Usage Guidelines

Use this command to set the maximum bit-rate for all video-capable phones in a Cisco Unified SRST system.

Examples

The following example sets a maximum bit-rate of 256 kbps.

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

Router(config-call-manager-fallback)# video
Router(conf-cm-fallback-video)# maximum bit-rate 256
```

Related Commands

Command	Description
maximum bit-rate (telephony-service)	Sets the maximum bit-rate for all video-capable phones associated with a Cisco Unified CME router.

max-presentation

To set the number of call presentation lines supported by a phone type, use the **max-presentation** command in ephone-type configuration mode. To reset to the default, use the **no** form of this command.

max-presentation *number*
no max-presentation

Syntax Description

number	Number of presentation lines. Range: 1 to 100. Default: 0. See Table 4: Supported Values for Ephone-Type Commands, on page 134 for the number of presentation lines supported by each phone type.

Command Default

No display lines are supported by the phone type.

Command Modes

Ephone-type configuration (config-ephone-type)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)XZ	Cisco Unified CME 4.3 Cisco Unified SRST 4.3	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0 Cisco Unified SRST 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

This command defines the number of presentation lines that are supported for the type of phone being added with an ephone-type template.

Table 4: Supported Values for Ephone-Type Commands

Supported Device	device-id	num-buttons	max-presentation
Cisco Unified IP Conference Station 7937G	431	1	6
Nokia E61	376	1	1

Examples

The following example shows that 1 presentation line is specified for the Nokia E61 when creating the ephone-type template.

```
Router(config)# ephone-type E61
Router(config-ephone-type)# max-presentation 1
```

Related Commands

Command	Description
device-id	Specifies the device ID for a phone type in an ephone-type template.
num-buttons	Sets the number of line buttons supported by a phone type.
type	Assigns the phone type to an SCCP phone.

mode esrst

To enable Enhanced SRST mode with additional feature support for SCCP phones, use the **mode esrst** command. To disable the ESRST mode, use the **no** form of this command.

mode esrst
no mode esrst

Syntax Description

This command has no arguments or keywords.

Command Default

By default, ESRST mode is disabled.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
15.4(3)M	Cisco Unified Enhanced SRST 10.5	This command was introduced.

Usage Guidelines

This command enables the enhanced SRST mode for SCCP phones.

Example

The following example shows that esrst mode is enabled:

```
Router(config)# telephony-service
Router(config-telephony)# mode esrst
```

Related Commands

Command	Description
telephony-service	Enters telephony-service configuration mode.
voice register global	Enters voice register global configuration mode.

moh (call-manager-fallback)

To enable music on hold (MOH), use the **moh** command in call-manager-fallback configuration mode. To disable music on hold, use the **no** form of this command.

moh *filename*
no moh *filename*

Syntax Description

filename	Filename of the music file. The music file must be in the system flash.

Command Default

MOH is enabled.

Command Modes

Call-manager-fallback configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(2)XT	Cisco SRST 2.0	This command was introduced on the following platforms: Cisco 1750, Cisco 1751, Cisco 2600 series and Cisco 3600 series multiservice routers, Cisco IAD2420 series IADs.
12.2(8)T	Cisco SRST 2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3 routers.
12.2(8)T1	Cisco SRST 2.0	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
12.2(11)T	Cisco SRST 2.01	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.
Cisco IOS XE Amsterdam 17.2.1r	Cisco Unified SIP SRST 12.8	Introduced support for YANG models.

Usage Guidelines

The **moh** command allows you to specify the .au and .wav format music files that are played to callers who have been put on hold. MOH works only for G.711 calls and on-net VoIP and PSTN calls. For all other calls, callers hear a periodic tone. For example, internal calls between Cisco IP phones do not get MOH; instead callers hear a tone.

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

MOH can be used as a fallback MOH source when using MOH live feed. See the **moh-live (call-manager-fallback)** command for more information.

Examples

The following example enables MOH and specifies the music files:

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

Related Commands

Command	Description
call-manager-fallback	Enables Cisco Unified SRST support and enters call-manager-fallback configuration mode.
moh-live (call- manager-fallback)	Specifies that a particular telephone number is to be used for an outgoing call that is to be the source for an MOH stream for SRST.

moh (voice moh-group)

To enable music on hold (MOH) for a MOH group, use the **moh** command in voice moh-group configuration mode. To disable music on hold, use the no form of this command.

moh *filename*
no moh *filename*

Syntax Description

filename	Name of the music file. The music file must be in the system flash.

Command Default

No MOH is enabled

Command Modes

V
oice moh-group configuration (config-voice-moh-group)

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME/SRST/SIP SRST 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME/SRST/SIP SRST 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

The **moh** command allows you to specify the .au and .wav format music files that are played to callers who have been put on hold. MOH works only for G.711 calls and on-net VoIP and PSTN calls. For all other calls, callers hear a periodic tone. You must provide the directory and filename of the MOH file in URL format. For example: moh flash:/minuet.au

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

Examples

The following example enables MOH for voice moh group 1 and specifies the music files:

```
Router(config)#
Router(config)#voice moh-group 1
Router(config-voice-moh-group)# moh flash:/minuet.wav
```

Related Commands

voice moh-group	Enters voice moh-group configuration mode.
extension-range	Defines extension range for a clients calling a voice-moh-group.
moh	Enables music on hold from a flash audio file.
multicast moh	Enables multicast of the music-on-hold audio stream.

moh-file-buffer (cm-fallback)

To specify a MOH file buffer size, use the **moh-file-buffer** command in call-manager-fallback configuration mode. To delete the moh-file-buffer size, use the **no** form of this command.

moh-file-buffer *file_size*
no moh-file-buffer

Syntax Description

file_size	Specifies a numeric value for the buffer MOH file size between 64 KB and 10000 KB.

Command Default

No moh-file buffer is configured.

Command Modes

Call-manager-fallback configuration (config-cm-fallback)

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified SRST 8.0	This command was introduced.
15.1(1)T	Cisco Unified SRST 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

This command allows to set a buffer MOH file size limit for new MOH files. You can allocate a MOH file buffer size between 64 KB (8 seconds) and 10000 KB (20 minutes, approximately). A large buffer size is desirable to cache the largest MOH file and better MOH performance. During memory allocation the buffer size is aligned to 16KB. For more information on MOH file caching, see show ephone moh command.

The default maximum file buffer size is 64 KB. If the MOH file size is too large, it cannot be cached and the buffer size falls back to 64 KB.

Note When live-feed is enabled there is no file caching for MOH-group 0.

Examples

The following example shows a moh-file-buffer size of 2000 KB assigned for future moh files under the call-manager-fallback configuration mode.

```
!
!
!
!
call-manager-fallback
 max-conferences 8 gain -6
 transfer-system full-consult
 moh-file-buffer 2000
!
!
line con 0
 exec-timeout 0 0
line aux 0
 --More--
```

Related Commands

Command	Description
moh-group	Allows to configure a MOH group.
moh	Enables music on hold from a flash audio feed
multicast moh	Enables multicast of the music-on-hold audio stream.
extension-range	Specifies the extension range for a clients calling a voice-moh-group.

moh-live (call-manager-fallback)

To specify that a particular telephone number is to be used for an outgoing call that is to be the source for a music on hold (MOH) stream for SRST, use the **moh-live** command in call-manager-fallback configuration mode. To disable the source for the MOH stream, use the **no** form of this command.

moh-live **dn-number** *calling-number* **out-call** *outcall-number*
no **moh-live** **dn-number** *calling-number* **out-call** *outcall-number*

Syntax Description

dn-number *calling-number*	Sets the MOH telephone number. The *calling-number* is a sequence of digits that represent a telephone number.
out-call *outcall-number*	Indicates that the router is calling out for a live feed that is to be used for MOH and specifies the number to be called. The *outcall-number* is a sequence of digits that represent a telephone number, typically of an E&M port.

Command Default

No default behavior or values.

Command Modes

Call-manager-fallback configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)T	Cisco SRST 3.3	This command was introduced.

Usage Guidelines

The **moh-live** command in SRST mode provides live-feed MOH streams from an audio device connected to an E&M or FXO port to Cisco IP phones.

During SRST, do not change an existing **moh-live** command. Instead, first remove the existing command, and then add in the new command.

To configure MOH from a live feed, you first establish a voice port and dial peer for the call and then create a “dummy” phone or directory number. The dummy number allows for making and receiving calls, but the number is not assigned to a physical phone. It is that number that the MOH system automatically dials to establish the MOH feed.

The **moh-live** command allocates one of the virtual voice ports from the pool of virtual voice ports created by the **max-dn** command. The virtual voice port places an outgoing call to the dummy number; that is, the directory number specified in the **moh-live** command. The audio stream obtained from the MOH call provides the music-on-hold audio stream.

To activate MOH live feed, set up and connect the physical voice port. After setting up the voice port, create a dial peer and give the voice port a directory number with the **destination-pattern** command. The directory number is the number that the system uses to access the MOH. To establish the MOH feed, connect the music source, such as a CD player to autodial the directory number.

MOH can be used as a fallback MOH source when using MOH live feed. See the **moh (call-manager-fallback)** command for more information.

Examples

The following example configures MOH from a live feed. Note that the dial peer references the E&M port that was set with the **voice-port** command and that the dial-peer number (7777) matches the out-call value of the **moh-live** command.

```
.
.
.
voice-port 1/0/0
 input gain 3
 auto-cut-through
 operation 4-wire
 signal immediate
!
dial-peer voice 7777 pots
 destination-pattern 7777
 port 2/0/0
!
!
call-manager-fallback
 max-conferences 8
 max-dn 1
 moh-live dn-number 3333 out-call 7777
!
.
.
.
```

Related Commands

Command	Description
call-manager-fallback	Enables Cisco Unified SRST support and enters call-manager-fallback configuration mode.
destination-pattern	Specifies either the prefix or the full E.164 telephone number to be used for a dial peer.
moh (call-manager-fallback)	Enables MOH.

multicast moh (call-manager-fallback)

To enable continuous IP multicast output of music on hold (MOH) from a flash MOH file in a branch Cisco Unified SRST router, use the **multicast moh** command in call-manager-fallback configuration mode. To disable continuous IP multicast output of MOH from a flash MOH file in a branch Cisco Unified SRST router, use the no form of this command.

multicast moh *multicast-address* **port** *port* [**route** *ip-address-list*]
no multicast moh *multicast-address* **port** *port* [**route** *ip-address-list*]

Syntax Description

multicast-address	Declares the multicast-address header value of the MOH packets that are to be multicast.
port	Declares the port header value of the MOH packets that are to be multicast.
port	The desired port number.
route	(Optional) Sets the explicit IP address or addresses from which the multicast packets will be transmitted.
ip-address-list	(Optional) Declares the IP address or addresses. The ip-address-list argument accepts a maximum of four IP address entries separated by a space.

Command Default

If the **route** keyword is not in the **multicast moh** command, the **ip source-address** command value configured for Cisco Unified SRST will be used.

Command Modes

Call-manager-fallback configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco SRST 3.0	This command was introduced.
12.3(4)T	Cisco SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Usage Guidelines

MOH multicast operates during Cisco Unified Communications Manager fallback and normal Cisco Unified Communications Manager service.

Because all branch IP phones are instructed by the central Cisco Unified Communications Manager to select and multicast a specific MOH feed, you must configure the IP address and port header values to match the MOH multicast IP address and port used by the central Cisco Unified Communications Manager's MOH. This makes the MOH from Cisco Unified SRST router appear as if it were coming from the central Cisco Unified Communications Manager.

MOH multicast operates with both MOH and MOH live input sources.

Examples

The following example provides multicast MOH output to the IP phones local to SRST. The MOH packets are broadcast to the 239.10.20.30 multicast address using port 2000. The MOH packets use the **ip source-address** command value configured for Cisco Unified SRST. MOH packets are output only through the router interface that matches the SRST router's IP source-address:

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# moh music-on-hold-file.au
Router(config-cm-fallback)# multicast moh 239.10.20.30 port 2000
```

The following example provides multicast MOH output to the IP phones local to SRST. The MOH packets are broadcast to the 239.10.20.30 multicast address using port 2000. The MOH packets use the **ip source-address** command value configured for Cisco Unified SRST. MoH packets are output only through the router interfaces that match the IP addresses listed using the **route** keyword option:

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# moh music-on-hold-file.au
Router(config-cm-fallback)# multicast moh 239.10.20.30 port 2000 route 10.10.20.1 10.10.21.1
 10.10.22.1 10.10.23.1
```

Related Commands

Command	Description
call-manager-fallback	Enables Cisco Unified SRST and enters **call-manager-fallback** configuration mode.
ip source-address (call-manager-fallback)	Enables a router to receive messages from Cisco IP phones through the specified IP addresses and ports.
moh (call-manager- fallback)	Enables MOH.
moh-live (call- manager-fallback)	Specifies that a particular telephone number is to be used for an outgoing call that is to be the source for an MOH stream for SRST.

mwi expires (call-manager-fallback)

Effective with Cisco IOS Releases 12.3(11)T7 and 12.4, the **mwi expires** command was replaced with the **mwi-server** command in SIP user-agent configuration mode.

To set the expiration timer for registration for the message-waiting indication (MWI) client or server, use the **mwi expires** command in call-manager-fallback configuration mode. To disable the timer, use the **no** form of this command.

mwi expires *seconds*
no mwi expires *seconds*

Syntax Description

seconds	Expiration time, in seconds. Range is from 600 to 99999. Default is 86400 (24 hours).

Command Default

86400 seconds (24 hours)

Command Modes

Call-manager-fallback configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco SRST 3.0	This command was introduced.
12.3(4)T	Cisco SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.3(11)T7 12.4	Cisco SRST 3.3	This command was replaced by the **mwi-server** command in SIP user-agent configuration mode.

Examples

The following example sets the expiration timer to 1000 seconds:

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# mwi expires 1000
```

Related Commands

Command	Description
mwi relay (call-manager-fallback)	Enables a Cisco Unified SRST router to relay MWI notification to remote Cisco IP phones.

mwi reg-e164 (call-manager-fallback)

To register E.164 numbers rather than extension numbers with a Session Interface Protocol (SIP) proxy or registrar, use the **mwi reg-e164** command in call-manager-fallback configuration mode. To return to the default, use the **no** form of this command.

mwi reg-e164
no mwi reg-e164

Syntax Description

This command has no keywords or arguments.

Command Default

Registering extension numbers with the SIP proxy or registrar.

Command Modes

Call-manager-fallback configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)T7 12.4	Cisco SRST 3.3	This command was introduced.

Usage Guidelines

This command is used when setting up extensions to use an external SIP-based message-waiting indication (MWI) server. The **mwi-server** command in SIP user-agent configuration mode specifies other settings for MWI service.

Examples

The following example specifies that E.164 numbers should be used for registration with the SIP proxy or registrar:

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# mwi reg-e164
```

Related Commands

Command	Description
mwi-server (SIP user-agent)	Specifies voice-mail server settings on a voice gateway or user agent (UA).

mwi relay (call-manager-fallback)

To enable a Cisco Unified SRST router to relay message-waiting indication (MWI) notification to remote Cisco IP phones, use the **mwi relay** command in call-manager-fallback configuration mode. To disable MWI relay, use the **no** form of this command.

mwi relay
no mwi relay

Syntax Description

This command has no arguments or keywords.

Command Default

MWI is not enabled.

Command Modes

Call-manager-fallback configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco SRST 3.0	This command was introduced.
12.3(4)T	Cisco SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Usage Guidelines

Use this command to enable the Cisco Unified SRST router to relay MWI notification to remote Cisco IP phones. The router at the central site acts as a notifier after this command is used.

Examples

The following example enables MWI relay:

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# mwi relay
```

Related Commands

Command	Description
mwi expires (call-manager- fallback)	Sets the expiration timer for registration for the client or the server.

mwi sip-server (call-manager-fallback)

Effective with Cisco IOS Releases 12.3(11)T7 and 12.4, the **mwi sip-server** command was replaced with the **mwi-server** command in SIP user-agent configuration mode and the **mwi reg-e164** command in call-manager-fallback configuration mode.

To configure the IP address and port number for an external SIP-based message-waiting indication (MWI) server, use the **mwi sip-server** command in call-manager-fallback configuration mode. To disable the MWI server functionality, use the **no** form of this command.

mwi sip-server *ip-address* [{**transport tcp** | **transport udp**}] [**port** *port-number*] [**reg-e164**] [**unsolicited**]
no mwi sip-server *ip-address*

Syntax Description

ip-address	IP address of the MWI server.
transport tcp	(Optional) Selects TCP as the transport layer protocol. This is the default transport protocol.
transport udp	(Optional) Selects UDP as the transport layer protocol.
port port-number	(Optional) Specifies port number for the MWI server. Range is from 2000 to 9999. Default is 5060.
reg-e164	(Optional) Registers an E.164 number with a Session Interface Protocol (SIP) proxy or registrar rather than an extension number. Registering with an extension number is the default.
unsolicited	(Optional) Sends SIP NOTIFY for MWI without any need to send a SUBSCRIBE from the Cisco Unified SRST router.

Command Default

Transport layer protocol: TCP Port number: 5060 (SIP standard port) Registration: with an extension number

Command Modes

Call-manager-fallback configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco SRST 3.0	This command was introduced.
12.3(4)T	Cisco SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.3(11)T7 12.4	Cisco SRST 3.3	This command was replaced by the **mwi-server** command in SIP user-agent configuration mode and the **mwi reg-e164** command in call-manager-fallback configuration mode.

Usage Guidelines

Use this command to configure the IP address of an external SIP MWI server.

The **transport tcp** keyword is the default setting. The **transport udp** keyword allows you to integrate with a SIP MWI client. The optional **port** keyword is used to specify a port number other than 5060, the default. The default registration is with an extension number, so the **reg-e164** keyword allows you to register with an E.164 ten-digit number.

Examples

The following example sets MWI for the SIP server and sets individual ephone-dn extension numbers to the MWI SIP server's notification list:

```
Router(config) call-manager-fallback
Router(config-cm-fallback) mwi sip-server 192.168.0.5 transport udp
```

Related Commands

Command	Description
mwi relay (call-manager-fallback)	Enables a Cisco Unified SRST router to relay MWI notification to remote Cisco IP phones.