Command Reference: N through Z

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.

name (voice emergency response location)

To describe or identify an emergency response location, use the **name** command in voice emergency response location mode. To remove this definition, use the **no** form of this command.

name *string*
no name

Syntax Description

string	String (30 characters) used to describe or identify an ERL's location.

Command Default

The location is not described.

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

Use this command to enable a word or description of the ERL for administrative purposes. Most commonly, use this command to identify the location for the network administrator.

Examples

In this example, the location description is Widget Incorporated.

```
voice emergency response location 60
subnet 1 209.165.200.224 255.255.0.0
elin 1 4085550101
name Widget Incorporated,
```

Related Commands

Command	Description
address	Specifies a comma separated text entry (up to 247 characters) of an ERL's civic address.
elin	Specifies a PSTN number that will replace the caller's extension.
subnet	Defines which IP phones are part of this ERL.
voice emergency response location	Creates a tag for identifying an ERL for E911 services.

name (voice hunt-group)

To associate a name with a called voice hunt group, use the **name** command in voice hunt-group configuration mode. To dissociate the name of the called voice hunt group, use the **no** form of this command.

name *"primary pilot name"* **[secondary** *"secondary pilot name"* **]**
no name*"primary pilot name "* **[secondary** *"secondary pilot name"* **]**

Syntax Description

"primary pilot name"	Name of primary pilot number.
secondary *"secondary pilot name"*	(Optional) Name of secondary pilot number.

Command Default

No name is associated with the called voice hunt group.

Command Modes

Voice hunt-group configuration (config-voice-hunt-group)

Command History

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

Usage Guidelines

In Cisco Unified SRST 9.5, when the secondary pilot name is not explicitly configured, the primary pilot name is applicable to both pilot numbers.

Note Use quotes (") when input strings have spaces in between.

Examples

The following example configures the primary pilot name for both the primary and the secondary pilot numbers:

```
name SALES
```

The following example configures different names for the primary and secondary pilot numbers:

```
name SALES secondary SALES-SECONDARY
```

The following example associates a two-word name for the primary pilot number and a one-word name for the secondary pilot number:

```
name "CUSTOMER SERVICE" secondary CS
```

The following example associates a one-word name for the primary pilot number and a two-word name for the secondary pilot number:

```
name FINANCE secondary "INTERNAL ACCOUNTING"
```

The following example associates two-word names for the primary and secondary pilot numbers:

```
name "INTERNAL CALLER" secondary "EXTERNAL CALLER"
```

When incoming call A reaches voice hunt group B and lands on final C, extension C does not show the name of the forwarder because the voice hunt group is not configured to display the name. To display the name of the forwarder and the final number, two separate names are required for the primary and secondary pilots.

The following example shows how the primary and secondary pilot names are configured in voice hunt-group configuration mode:

```
voice hunt-group 24 parallel
   final 097
   list 885,886,124,154
   timeout 20
   pilot 021 secondary 621
   name SALES secondary SALES-SECONDARY
```

The following is a sample output of the **show voice hunt-group** command when the primary and secondary pilot names are configured in voice hunt-group configuration mode:

```
show voice hunt-group 1
Group 1
    type: parallel
    pilot number: 1000, peer-tag 2147483647
    secondary number: 2000, peer-tag 2147483646
    pilot name: SALES
    secondary name: SALES-SECONDARY
    list of numbers: 2004,2005
```

Related Commands

Command	Description
voice hunt-group	Enters voice hunt-group configuration mode and creates a hunt group for phones in a Cisco Unified CME system.
show voice hunt-group	Displays configuration information associated with one or all voice hunt groups in a Cisco Unified CME system.

number (voice register pool)

To indicate the E.164 phone numbers that the registrar permits to handle the Register message from a Cisco Unified SIP IP phone, use the **number** command in voice register pool configuration mode. To disable number registration, use the **no** form of this command.

number *tag* {*number-pattern* [**preference** *value*] [**huntstop**] | **dn** *dn-tag*}
no number *tag*

Syntax Description

tag	Telephone number when there are multiple **number** commands. Range is 1 to 114.
number-pattern	Phone numbers (including wild cards and patterns) that are permitted by the registrar to handle the Register message from the Cisco Unified SIP IP phone.
preference *value*	(Optional) Defines the number list preference order. Range is 0 to 10. The highest preference is 0. There is no default.
huntstop	(Optional) Stops hunting when the dial peer is busy.
dn *dn-tag*	Identifies the directory number tag for this phone number as defined by the **voice register dn** command. Range is 1 to 288.

Command Default

Cisco Unified SIP IP phones cannot register in Cisco Unified CME.

Command Modes

Voice register pool configuration (config-register-pool)

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 CME and the **dn** keyword was added.
12.4(11)XJ	Cisco Unified CME 4.1 Cisco Unified SRST 4.1	This command was modified. The *number-pattern* argument and **preference** and **huntstop** keywords were removed from Cisco Unified CME.
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.
15.2(4)M	Cisco Unified CME 9.1 Cisco Unified SIP SRST 9.1	This command was modified to increase the valid value of the *tag* argument to 114.

Usage Guidelines

The **number** command indicates the phone numbers that are permitted by the registrar to handle the Register message from the Cisco Unified SIP IP phone.

In Cisco Unified SRST, the keywords and arguments of this command allow for more explicit setting of user preferences regarding what number patterns should match the voice register pool.

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

Examples

The following example shows three directory numbers assigned to Cisco Unified SIP IP phone 1 in Cisco Unified CME:

```
!
voice register pool  1
 id mac 0017.E033.0284
 type 7961
 number 1 dn 10
 number 2 dn 12
 number 3 dn 13
 codec g711ulaw
!
```

The following example shows directory numbers 10, 12, and 13 assigned to phone numbers 1, 2, and 55 of Cisco Unified SIP IP phone 2:

```
voice register pool 2
id mac 0017.E033.0284
type 7961
number 1 dn 10
number 2 dn 12
number 55 dn 13
codec g711ulaw
```

The following example shows a telephone number pattern set to 95... in Cisco Unified SRST. This means all five-digit numbers beginning with 95 are permitted by the registrar to handle the Register message.

```
voice register pool 3
 id network 10.2.161.0 mask 255.255.255.0
 number 1 95... preference 1
 cor incoming call95 1 95011
```

Related Commands

Command	Description
id (voice register pool)	Explicitly identifies a locally available, individual Cisco Unified SIP IP phone or, when running Cisco Unified SIP SRST, a set of Cisco Unified SIP IP phones.
voice register dn	Enters voice register dn configuration mode to define an extension for a phone line, intercom line, voice-mail port, or a message-waiting indicator.

outbound-proxy system

To specify that all Cisco Unified Communications Manager Express (Cisco Unified CME) line-side phones connected to a Cisco IOS voice gateway use the global settings for forwarding Session Initiation Protocol (SIP) messages to an outbound proxy, use the **outbound-proxy system** command in voice register global configuration mode. To disable the SIP outbound proxy function for Cisco Unified CME line-side SIP phones, use the **no** form of this command.

outbound-proxy system
no outbound-proxy system

Syntax Description

This command has no arguments or keywords.

Command Default

The SIP outbound proxy function for all SIP line-side phones in Cisco Unified CME is enabled and behavior is determined by the global setting on the Cisco IOS gateway.

Command Modes

Voice register global configuration (config-register-global)

Command History

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

Usage Guidelines

If global configuration for outbound proxy is enabled on the Cisco IOS voice gateway and Cisco Unified CME receives a call, Cisco Unified CME forwards all SIP messages to the outbound proxy causing incoming calls to line-side SIP phones to fail. This is the default behavior.

To avoid these failed calls, use the **no** form of this command in voice register global configuration mode to override global outbound proxy settings for the gateway and disable the outbound proxy function for all line-side SIP phones connected to Cisco Unified CME.

To configure outbound proxy settings for an individual dial peer on the gateway, use the **voice-class sip outbound-proxy** command in dial peer voice configuration mode .

Examples

The following example shows how to disable the global outbound proxy feature for all line-side SIP phones on a Cisco Unified CME:

```
Router> enable
Router# configure
 terminal
Router(config)# voice register global
Router(config-register-global)# no outbound-proxy
```

Related Commands

Command	Description
voice-class sip outbound-proxy	Configures SIP outbound proxy settings for an individual dial peer that override global settings for the Cisco IOS voice gateway.

overlap-signal

To configure overlap dialing in SCCP or SIP IP phones, use the overlap-signal command in ephone, ephone-template, telephony-service, voice register pool, voice register global, or voice register template configuration mode.

overlap-signal

Syntax Description

This command has no arguments or keywords.

Command Default

Overlap-signal is disabled.

Command Modes

Call-manager-fallback
Ephone configuration (config-ephone)
Ephone-template configuration (config-ephone-template)
Telephony-service configuration (config-telephony)
Voice register pool (config-register-pool)
Voice register global configuration (config-register-global)
Voice register template (config-register-template)

Command History

Cisco IOS Release	Cisco Product	Modification
15.1(3)T	Cisco Unified CME 8.5 Cisco Unified SRST 8.5	This command was introduced.

Usage Guidelines

SCCP IP phones

In SCCP IP phones, overlap dialing is enabled when the overlap signal command is configured in ephone, ephone-template, and telephony-service configurations modes.

SIP IP phones

In SIP IP Phones, overlap dialing is enabled when the overlap signal command is configured in voice register pool, voice register global, and voice register template configuration modes.

Cisco Unified SRST

In Cisco Unified SRST, overlap dialing is enabled on SCCP IP phones when overlap signal command is configured in call-manager-fallback configuration mode.

Examples

The following example shows overlap-signal enabled on SCCP phones:

```
Router# show running config
!
!
telephony-service
 max-ephones 25
 max-dn 15
 load 7906 SCCP11.8-5-3S.loads
 load 7911 SCCP11.8-5-3S.loads
 load 7921 CP7921G-1.3.3.LOADS
 load 7941 SCCP41.8-5-3S.loads
 load 7942 SCCP42.8-5-3S.loads
```

```
 load 7961 SCCP41.8-5-3S.loads
 load 7962 SCCP42.8-5-3S.loads
 max-conferences 12 gain -6
 web admin system name cisco password cisco
 transfer-system full-consult
 create cnf-files version-stamp Jan 01 2002 00:00:00
 overlap-signal
!
ephone-template  1
 button-layout 1 line
 button-layout 3-6 blf-speed-dial
!
ephone-template  9
 feature-button 1 Endcall
 feature-button 3 Mobility
!
!
ephone-template  10
 feature-button 1 Park
 feature-button 2 MeetMe
 feature-button 3 CallBack
 button-layout 1 line
 button-layout 2-4 speed-dial
 button-layout 5-6 blf-speed-dial
 overlap-signal
!
ephone  10
 device-security-mode none
 mac-address 02EA.EAEA.0010
 overlap-signal
```

!

The following example shows overlap-signal configured in voice register global and voice register pool 10:

```
Router#show running config
!
!
!
voice service voip
 ip address trusted list
  ipv4 20.20.20.1
 media flow-around
 allow-connections sip to sip
!
voice class media 10
 media flow-around
!
!
voice register global
 max-pool 10
 overlap-signal
!
voice register pool  5
 overlap-signal
!
!
!
```

The following example shows overlap-signal configured in call-manager-fallback mode:

```
Router# show run | sec call-manager
```

```
call-manager-fallback
 max-conferences 12 gain -6
 transfer-system full-consult
 overlap-signal
```

pattern direct (vm-integration)

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

pattern direct *tag1* {**CGN** | **CDN** | **FDN**} [*tag2* {**CGN** | **CDN** | **FDN**}] [*tag3* {**CGN** | **CDN** | **FDN**}] [**last-tag**]
no pattern direct *tag1* {**CGN** | **CDN** | **FDN**} [*tag2* {**CGN** | **CDN** | **FDN**}] [*tag3* {**CGN** | **CDN** | **FDN**}] [*last-tag*]

Syntax Description

tag1	Alphanumeric string fewer than four DTMF digits in length. The alphanumeric string consists of a combination of four letters (A,B,C, and D), two symbols (* and #), and ten digits (0 to 9). The tag numbers match the numbers defined in the voice-mail system's integration file, immediately preceding either the number of the calling party, the number of the called party, or a forwarding number.
tag2 and *tag3*	(Optional) See *tag1*.
last-tag	(Optional) See *tag1*. This tag indicates the end of the pattern.
CGN	Calling number (CGN) information is sent to the voice-mail system.
CDN	Called number (CDN) information is sent to the voice-mail system.
FDN	Forwarding number (FDN) information is sent to the voice-mail system.

Command Default

This feature is disabled.

Command Modes

Voice-mail integration 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 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(13)T	Cisco SRST 2.02	This command was introduced for Cisco SRST Version 2.02.

Usage Guidelines

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

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

Examples

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

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

Related Commands

Command	Description
pattern ext-to-ext busy (vm-integration)	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an internal extension attempts to connect to a busy extension and the call is forwarded to voice mail
pattern ext-to-ext no-answer (vm- integration)	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an internal extension fails to connect to an extension and the call is forwarded to voice mail.
pattern trunk-to-ext busy (vm-integration)	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an external trunk call reaches a busy extension and the call is forwarded to voice mail.
pattern trunk-to-ext no-answer (vm- integration)	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when an external trunk call reaches an unanswered extension and the call is forwarded to voice mail.
vm-integration	Enters voice-mail integration configuration mode and enables voice-mail integration with DTMF and analog voice-mail systems.

pattern ext-to-ext busy (vm-integration)

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

pattern ext-to-ext busy *tag1* {**CGN** | **CDN** | **FDN**} [*tag2* {**CGN** | **CDN** | **FDN**}] [*tag3* {**CGN** | **CDN** | **FDN**}] [*last-tag*]
no pattern ext-to-ext busy *tag1* {**CGN** | **CDN** | **FDN**} [*tag2* {**CGN** | **CDN** | **FDN**}] [*tag3* {**CGN** | **CDN** | **FDN**}] [*last-tag*]

Syntax Description

tag1	Alphanumeric string fewer than four DTMF digits in length. The alphanumeric string consists of a combination of four letters (A,B,C, and D), two symbols (* and #), and ten digits (0 to 9). The tag numbers match the numbers defined in the voice-mail system's integration file, immediately preceding either the number of the calling party, the number of the called party, or a forwarding number.
tag2 and *tag3*	(Optional) See *tag1*.
last-tag	(Optional) See *tag1*. This tag indicates the end of the pattern.
CGN	Calling number (CGN) information is sent to the voice-mail system.
CDN	Called number (CDN) information is sent to the voice-mail system.
FDN	Forwarding number (FDN) information is sent to the voice-mail system.

Command Default

This feature is disabled.

Command Modes

Voice-mail integration 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 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(13)T	Cisco SRST 2.02	This command was introduced for Cisco SRST Version 2.02.

Usage Guidelines

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

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

Examples

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

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

Related Commands

Command	Description
pattern direct (vm- integration)	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when the user presses the messages button on the phone.
pattern ext-to-ext no-answer (vm-integration)	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an internal extension fails to connect to an extension and the call is forwarded to voice mail.
pattern trunk-to-ext busy (vm-integration)	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an external trunk call reaches a busy extension and the call is forwarded to voice mail.
pattern trunk-to-ext no-answer (vm- integration)	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when an external trunk call reaches an unanswered extension and the call is forwarded to voice mail.
vm-integration	Enters voice-mail integration configuration mode and enables voice-mail integration with DTMF and analog voice-mail systems.

pattern ext-to-ext no-answer (vm-integration)

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

pattern ext-to-ext no-answer *tag1* {**CGN** | **CDN** | **FDN**} [*tag2* {**CGN** | **CDN** | **FDN**}] [*tag3* {**CGN** | **CDN** | **FDN**}] [*last-tag*]
no pattern ext-to-ext no-answer *tag1* {**CGN** | **CDN** | **FDN**} [*tag2* {**CGN** | **CDN** | **FDN**}] [*tag3* {**CGN** | **CDN** | **FDN**}] [*last-tag*]

Syntax Description

tag1	Alphanumeric string fewer than four DTMF digits in length. The alphanumeric string consists of a combination of four letters (A,B,C, and D), two symbols (* and #), and ten digits (0 to 9). The tag numbers match the numbers defined in the voice-mail system's integration file, immediately preceding either the number of the calling party, the number of the called party, or a forwarding number.
tag2 and *tag3*	(Optional) See *tag1*.
last-tag	(Optional) See *tag1*. This tag indicates the end of the pattern.
CGN	Calling number (CGN) information is sent to the voice-mail system.
CDN	Called number (CDN) information is sent to the voice-mail system.
FDN	Forwarding number (FDN) information is sent to the voice-mail system.

Command Default

This feature is disabled.

Command Modes

Voice-mail integration 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 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	2.01	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760 routers.
12.2(13)T	2.02	This command was introduced for Cisco SRST Version 2.02.

Usage Guidelines

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

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

Examples

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

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

Related Commands

Command	Description
pattern direct (vm-integration)	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when the user presses the messages button on the phone.
pattern ext-to-ext busy (vm-integration)	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an internal extension fails to connect to an extension and the call is forwarded to voice mail.
pattern trunk-to-ext busy (vm-integration)	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an external trunk call reaches a busy extension and the call is forwarded to voice mail.
pattern trunk-to-ext no-answer (vm-integration)	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when an external trunk call reaches an unanswered extension and the call is forwarded to voice mail.
vm-integration	Enters voice-mail integration configuration mode and enables voice-mail integration with DTMF and analog voice-mail systems.

pattern trunk-to-ext busy (vm-integration)

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

pattern trunk-to-ext busy *tag1* {**CGN** | **CDN** | **FDN**} [*tag2* {**CGN** | **CDN** | **FDN**}] [*tag3* {**CGN** | **CDN** | **FDN**}] [*last-tag*]
no pattern trunk-to-ext busy *tag1* {**CGN** | **CDN** | **FDN**} [*tag2* {**CGN** | **CDN** | **FDN**}] [*tag3* {**CGN** | **CDN** | **FDN**}] [*last-tag*]

Syntax Description

tag1	Alphanumeric string fewer than four DTMF digits in length. The alphanumeric string consists of a combination of four letters (A,B,C, and D), two symbols (* and #), and ten digits (0 to 9). The tag numbers match the numbers defined in the voice-mail system's integration file, immediately preceding either the number of the calling party, the number of the called party, or a forwarding number.
tag2 and *tag3*	(Optional) See *tag1*.
last-tag	(Optional) See *tag1*. This tag indicates the end of the pattern.
CGN	Calling number (CGN) information is sent to the voice-mail system.
CDN	Called number (CDN) information is sent to the voice-mail system.
FDN	Forwarding number (FDN) information is sent to the voice-mail system.

Command Default

This feature is disabled.

Command Modes

Voice-mail integration 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 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(13)T	Cisco SRST 2.02	This command was introduced for Cisco SRST Version 2.02.

Usage Guidelines

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

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

Examples

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

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

Related Commands

Command	Description
pattern direct (vm- integration)	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when the user presses the messages button on the phone.
pattern ext-to-ext busy (vm-integration)	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an internal extension attempts to connect to a busy extension and the call is forwarded to voice mail.
pattern ext-to-ext no-answer (vm- integration)	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an internal extension fails to connect to an extension and the call is forwarded to voice mail.
pattern trunk-to-ext no-answer (vm- integration)	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when an external trunk call reaches an unanswered extension and the call is forwarded to voice mail.
vm-integration	Enters voice-mail integration configuration mode and enables voice-mail integration with DTMF and analog voice-mail systems.

pattern trunk-to-ext no-answer (vm-integration)

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

pattern trunk-to-ext no-answer *tag1* {**CGN** | **CDN** | **FDN**} [*tag2* {**CGN** | **CDN** | **FDN**}] [*tag3* {**CGN** | **CDN** | **FDN**}] [*last-tag*]
no pattern trunk-to-ext no-answer *tag1* {**CGN** | **CDN** | **FDN**} [*tag2* {**CGN** | **CDN** | **FDN**}] [*tag3* {**CGN** | **CDN** | **FDN**}] [*last-tag*]

Syntax Description

tag1	Alphanumeric string fewer than four DTMF digits in length. The alphanumeric string consists of a combination of four letters (A,B,C, and D), two symbols (* and #), and ten digits (0 to 9). The tag numbers match the numbers defined in the voice-mail system's integration file, immediately preceding either the number of the calling party, the number of the called party, or a forwarding number.
tag2 and *tag3*	(Optional) See *tag1*.
last-tag	(Optional) See *tag1*. This tag indicates the end of the pattern.
CGN	Calling number (CGN) information is sent to the voice-mail system.
CDN	Called number (CDN) information is sent to the voice-mail system.
FDN	Forwarding number (FDN) information is sent to the voice-mail system.

Command Default

This feature is disabled.

Command Modes

Voice-mail integration 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 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(13)T	Cisco SRST 2.02	This command was introduced for Cisco SRST Version 2.02.

Usage Guidelines

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

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

Examples

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

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

Related Commands

Command	Description
pattern direct (vm-integration)	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when the user presses the messages button on the phone.
pattern ext-to-ext busy (vm-integration)	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an internal extension attempts to connect to a busy extension and the call is forwarded to voice mail.
pattern ext-to-ext no-answer (vm-integration)	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an internal extension fails to connect to an extension and the call is forwarded to voice mail.
pattern trunk-to-ext busy (vm-integration)	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an external trunk call reaches a busy extension and the call is forwarded to voice mail.
vm-integration	Enters voice-mail integration configuration mode and enables voice-mail integration with DTMF and analog voice-mail systems.

phoneload

To define the phone firmware support for a phone type, use the **phoneload** command in ephone-type configuration mode. To remove firmware support, use the **no** form of this command.

phoneload
no phoneload

Syntax Description

This command has no arguments or keywords.

Command Default

Phone type supports firmware configuration.

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 whether the phone type defined in the phone-type template supports firmware configuration using the **load** command.

Examples

The following example shows that support for phone firmware is disabled for the Nokia E61 phone type:

```
Router(config)# ephone-type E61
Router(config-ephone-type)# no phoneload
```

Related Commands

Command	Description
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.

phone-display

To enable a SIP/SCCP phone user to display hunt group related information using the Services button on the phone, use the **phone-display** command in voice hunt-group configuration mode. To reset to the default value, use the **no** form of this command.

phone-display
no phone-display

Syntax Description

This command has no arguments or keywords.

Command Default

By default, this command is enabled.

Command Modes

Voice register template configuration (config-hunt-group)

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 user to display hunt group information on the phone.

Example

The following example shows that voice hunt group display option is disabled for phone 7:

```
Router(config)# ephone 7
Router(config-ephone-type)# no phone-ui voice-hunt-groups
```

phone-mode only

To enable Jabber phone-only client suppor, use the **phone-mode only** command. To exit the configuration, use the **no** form of the command.

phone-mode *phone-only*

Syntax Description

This command has no arguments or keywords.

Command Default

Privileged EXEC mode

Command Modes

Voice register global

Voice register pool

Voice register 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

This command enables Jabber client support for MAC, iPhone, iPad and android for SCCP and SIP phones.

Example

The following example shows that phone-mode is enabled:

```
Router(config)# voice register pool
Router(config-telephony)# phone-mode phone-only
```

Related Commands

Command	Description
Voice register global	Enters voice register global configuration mode.
Voice register pool	Enters voice register pool configuration mode.
Voice register template	Enters voice register template configuration mode.

pickup (call-manager-fallback)

To enable the PickUp soft key on all Cisco IP phones, allowing an external Direct Inward Dialing (DID) call coming into one extension to be picked up from another extension during SRST, use the **pickup** command in call-manager-fallback configuration mode. To disable the PickUp soft key on all Cisco IP phones during SRST, use the **no** form of this command.

pickup *telephone-number*
no pickup *telephone-number*

Syntax Description

telephone-number	The telephone number to match an incoming called number.

Command Default

The PickUp soft key is disabled.

Command Modes

Call-manager-fallback configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(7)XL	Cisco SRST 3.1.1	This command was introduced.
12.3(11)T	Cisco SRST 3.2	This command was integrated into Cisco IOS Release 12.3(11)T.

Usage Guidelines

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

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

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

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

Examples

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

```
call-manager-fallback
 no huntstop
 alias 1 8005550100 to 5001
```

```
alias 2 8005550100 to 5002
alias 3 8005550100 to 5003
alias 4 8005550100 to 5004
pickup 8005550100
```

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

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

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

preference (voice register pool)

To set the preference order for creating the VoIP dial peers created for a number associated with a voice pool, use the **preference** command in voice register pool configuration mode. To put the number in default preference order, use the **no** form of this command.

preference *preference-order*
no preference

Syntax Description

preference-order	Preference order for the extension or telephone number associated with a pool. Range is 0 to 10. Default is 0, which is the highest preference.

Command Default

0 (highest preference order)

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

When you create a voice register pool for a SIP phone or a group of SIP phones in a Cisco Unified CME or Cisco Unified Session Initiation Protocol (SIP) Survivable Remote Site Telephony (SRST) environemnt, you automatically create a virtual voice port and one to four virtual dial peers to be used by the number associated with that pool. The preference value is passed transparently to dial peers created for the number. The preference value allows you to control the selection of a desired dial peer when multiple dial peers are matched on the same destination pattern (extension or phone number) associated with the pool. In this way, the **preference** command can be used to establish a hunt strategy for incoming calls.

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

Examples

The following example shows how to set a preference of 2 for extension number 3000:

```
voice register pool 1
 number 3000
 preference 2
```

In the following example, extension number 1222 under voice register dn 4 has a higher preference than number 1222 under voice register pool 5.

```
voice register pool 4
 number 1222
 preference 0
!
!
voice register dn 5
 number 1222
 preference 1
```

Related Commands

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.
voice register pool	Enters voice register pool configuration mode for SIP phones.

proxy (voice register pool)

To autogenerate additional VoIP dial peers to reach the main proxy whenever a Cisco Session Initiation Protocol (SIP) IP phone registers with a SIP Survivable Remote Site Telephony (SRST) gateway, use the **proxy** command in voice register pool configuration mode. To disable a dial peer as a SIP proxy, use the **no** form of this command.

proxy *ip-address* [**preference** *value*] [**monitor probe** {**icmp-ping** | **rtr**} [*alternate-ip-address*]]
no proxy

Syntax Description

ip-address	IP address of the SIP proxy.
preference *value*	(Optional) Defines the preference of the proxy dial peers that are created. Range is from 0 to 10. The highest preference is 0. There is no default.
monitor probe	(Optional) Enables monitoring of proxy dial peers. • **icmp-ping**—Enables monitoring of proxy dial peers using ICMP ping. • **rtr**—Enables monitoring of proxy dial peers using RTR probes. • *alternate-ip-address*—(Optional) Enables monitoring of alternate IP addresses other than the proxy address. For example, to monitor a gateway front end to a SIP proxy.

Command Default

Proxy dial peer is disabled.

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.

Usage Guidelines

The **proxy** command autogenerates additional VoIP dial peers to reach the main proxy whenever a Cisco SIP IP phone registers with a Cisco Unified SIP SRST gateway. This autogeneration process enables all PSTN calls to be routed first to the main proxy before the backup dial peers for local Cisco SIP IP phones are tried.

Proxy dial peers can be monitored using ICMP ping or RTR probes, in case of WAN failure. If the Cisco Unified SIP SRST gateway loses probes to the main proxy, the proxy dial peers are temporarily set to an operational down state. Then the backup dial peers can be selected faster to lower the call setup time. In addition, the proxy dial peers can be monitored using an alternate location other than the main proxy to monitor the status of the WAN link.

Only one proxy address can be set per voice register pool.

For proxy monitoring to work, the **call fallback active** command must be configured.

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

Examples

The following partial sample output from the **show running-config** command shows that voice register pool 1 has defined 10.2.161.187 as the SIP proxy and that it is monitored by ICMP ping:

```
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
```

Related Commands

Command	Description
call fallback active	Enables a call request to fall back to alternate dial peers in case of network congestion.
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.

registrar server (SIP)

To enable SIP registrar functionality, use the **registrar server** command in SIP configuration mode. To disable SIP registrar functionality, use the **no** form of the command.

registrar server [**expires** [**max** *sec*] [**min** *sec*]]
no registrar server

Syntax Description

expires	(Optional) Sets the active time for an incoming registration.
max *sec*	(Optional) Maximum expires time for a registration, in seconds. The range is from 600 to 86400. The default is 3600.
min *sec*	(Optional) Minimum expires time for a registration, in seconds. The range is from 60 to 3600. The default is 60.

Command Default

Disabled

Command Modes

SIP 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

When this command is entered, the router accepts incoming SIP Register messages. If SIP Register message requests are for a shorter expiration time than what is set with this command, the SIP Register message expiration time is used.

This command is mandatory for Cisco Unified SIP SRST or Cisco Unified CME and must be entered before any **voice register pool** or **voice register global** commands are configured.

If the WAN is down and you reboot your Cisco Unified CME or Cisco Unified SIP SRST router, when the router reloads it will have no database of SIP phone registrations. The SIP phones will have to register again, which could take several minutes, because SIP phones do not use a keepalive functionality. To shorten the time before the phones re-register, the registration expiry can be adjusted with this command. The default expiry is 3600 seconds; an expiry of 600 seconds is recommended.

Examples

The following partial sample output from the **show running-config** command shows that SIP registrar functionality is set:

voice service voip

allow-connections sip-to-sip

sip

registrar server expires max 1200 min 300

Related Commands

Command	Description
sip	Enters SIP configuration mode from voice service VoIP configuration mode.
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.

reset (call-manager-fallback)

To reset Cisco IP phones, use the **reset** command in call-manager-fallback configuration mode.

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

Syntax Description

all	All Cisco IP phones.
seconds	Time interval, in seconds, that passes between each Cisco IP phone resetting. The range is from 0 to 60.
mac-address *mac-address*	MAC address of a particular Cisco IP phone.

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; 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

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

Examples

The following example resets all Cisco IP phones in 8-second intervals:

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

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

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

Related Commands

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

secondary-dialtone (call-manager-fallback)

To enable a secondary dial tone when a Cisco Unified IP phone user dials a defined PSTN access prefix, use the **secondary-dialtone** command in call-manager-fallback configuration mode. To disable the secondary dial tone, use the **no** form of this command.

secondary-dialtone *digit-string*
no secondary-dialtone *digit-string*

Syntax Description

digit-string	The number of the access prefix.

Command Default

Secondary dial tone is disabled.

Command Modes

Call-manager-fallback configuration

Command History

Cisco IOS Release	Cisco Unified 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 Unified IOS Release 12.3(4)T.

Usage Guidelines

The secondary dial tone is turned off when the next number after the *digit-string* is pressed. For example, if 8 were the *digit-string* and a person were dialing 8 555-0100, the secondary dial tone would be turned off when the number 5 is pressed.

The tone value for the secondary dial is the skinny DtOutsideDialtone.

Examples

The following enables a secondary dial tone when a Cisco Unified IP phone users enters the number nine to get an outside line:

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# secondary-dialtone 9
```

Related Commands

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

security

To define whether a phone type supports security features, use the **security** command in ephone-type configuration mode. To disable security support, use the **no** form of this command.

security
no security

Syntax Description

This command has no arguments or keywords.

Command Default

Enabled (phone type supports security features).

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 whether security features are supported by the type of phone being added with a phone-type template.

Examples

The following example shows that support for security is disabled for the Nokia E61 when creating the ephone-type template:

```
Router(config)# ephone-type E61
Router(config-ephone-type)# no security
```

Related Commands

Command	Description
device-id	Specifies the device ID for a phone type.
type	Assigns a phone type to an SCCP phone.

security-policy (voice register global)

To define the security level of SIP phones allowed to register, use the security-policy command in voice register global configuration mode. To return to the default, use the no form of this command.

Note The security-policy command only works with SRST. While it is possible to configure this command when in CME mode, TLS-based connections from Cisco Unified IP Phones will fail. This failure will occur even if using the "CME-as-SRST" failover model.

Cisco IOS Release 15.0(1)XA and later releases
security-policy secure
no security-policy secure

Syntax Description

secure	Requires SIP phones to use TLS for signaling transport. Non-secure SIP phones are blocked from registering. Valid for Cisco Unified SRST.

Command Default

All levels of phone security are permitted to register, also known as device-default mode.

Command Modes

Voice register global configuration (config-register-global)

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified SRST 8.0	This command was introduced.
Cisco IOS XE Dublin 17.10.1a	Unified Cisco SRST 14.3	Introduced support for YANG models.

Usage Guidelines

The secure keyword configures the SIP registration security policy so that only encrypted phones may register to the Cisco Unified SRST device in the event of a failover from the primary call control. When this keyword is configured, non-secure phones using TCP or UDP for signaling transport, as well as authenticated phones using TLS/TCP for signaling transport, will be blocked from registering.

Examples

The following example shows that only registration requests from encrypted SIP phones in a Cisco Unified SRST system are permitted:

```
Router(config)# voice register global
Router(config-register-global)# security-policy secure
```

Related Commands

Command	Description
crypto signaling	Identifies the trustpoint and encryption restrictions used during the TLS handshake.

show call-manager-fallback all

To display the detailed configuration of all Cisco IP phones, directory numbers, voice ports, and dial peers in your network during Cisco Unified CallManager fallback, use the **show call-manager-fallback all** command in privileged EXEC mode.

show call-manager-fallback all

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

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; 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.
12.3(4)T	Cisco SRST 3.0	The Version was added to output.

Examples

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

```
Router #show call-manager-fallback

CONFIG (Version=4.1(0))
=====================
Version 4.1(0)
For on-line documentation please see:
www.cisco.com/univercd/cc/td/doc/product/access/ip_ph/ip_ks/index.htm
ip source-address 0.0.0.0 port 2000
max-ephones 0
max-dn 0
max-conferences 8 gain -6
dspfarm units 0
dspfarm transcode sessions 0
huntstop
cnf-file location: system:
cnf-file option: PER-PHONE-TYPE
network-locale[0] US (This is the default network locale for this box)
```

```
network-locale[1] US
network-locale[2] US
network-locale[3] US
network-locale[4] US
user-locale[0] US (This is the default user locale for this box)
user-locale[1] US
user-locale[2] US
user-locale[3] US
user-locale[4] US
srst mode auto-provision is OFF
srst ephone template is 0
srst dn template is 0
srst dn line mode is single
time-format 12
date-format mm-dd-yy
timezone 0 Greenwich Standard Time
no transfer-pattern is configured, transfer is restricted to local SCCP phones only.
keepalive 30 auxiliary 30
timeout interdigit 10
timeout busy 10
timeout ringing 180
timeout ringin-callerid 8
caller-id name-only: enable
Limit number of DNs per phone:
7910: 36
7935: 36
7936: 36
7940: 36
7960: 36
7970: 36
Log (table parameters):
max-size: 150
retain-timer: 15
transfer-system full-consult
local directory service: enabled.
Extension-assigner tag-type ephone-tag.
========================================
```

Table 1: show call-manager-fallback all Field Descriptions, on page 41 describes the significant fields shown in the display.

Table 1: show call-manager-fallback all Field Descriptions

Field	Description
destination-pattern	Destination pattern (telephone number) configured for this dial peer.
dial-peer voice	Voice dial peer.
ephone-dn	Cisco IP phone directory number.
(no) huntstop	Whether or not huntstop is set.
ip source-address	IP address used by the Cisco IP phones to register with the router for service.
keepalive	Cisco IP phone keepalive period in seconds.
max-dn	Maximum directory numbers or virtual voice ports.
max-ephones	Maximum number of Cisco IP phones.

Field	Description
port	TCP port number used by the Cisco IP phones to communicate with the router.
station-id number	Number used for caller-ID purposes when calls are made using the line.
voice-port	(Virtual) voice port designator.
Version	SRST version number designation.

Related Commands

Command	Description
show call-manager- fallback dial-peer	Displays detailed configuration output for the dial peers in your network during Cisco Unified CallManager fallback.
show call-manager- fallback ephone-dn	Displays output for the Cisco IP phone directory numbers or virtual voice ports during Cisco Unified CallManager fallback.
show call-manager- fallback voice-port	Displays output for the voice ports while Cisco Unified CallManager is active.

show call-manager-fallback dial-peer

To display detailed configuration output for the dial peers in your network during Cisco Unified Communications Manager fallback, use the **show call-manager-fallback dial-peer** command in privileged EXEC mode.

show call-manager-fallback dial-peer

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

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; 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.

Examples

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

```
Router# show call-manager-fallback dial-peer
dial-peer voice 20046 pots
 destination-pattern 4444
 call-forward busy 5001
 call-forward noan 5001
 port 50/0/1
dial-peer voice 20047 pots
 destination-pattern 3333
 call-forward busy 5001
 call-forward noan 5001
 port 50/0/2
dial-peer voice 20048 pots
 destination-pattern 5555
 call-forward busy 5001
 call-forward noan 5001
 port 50/0/3
dial-peer voice 20049 pots
 preference 9
 destination-pattern 3...
```

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

Table 2: show call-manager-fallback dial-peer Field Descriptions, on page 44describes the significant fields shown in the display.

Table 2: show call-manager-fallback dial-peer Field Descriptions

Field	Description
call-forward busy	Indicates call forwarding when a Cisco IP phone is busy.
call-forward noan	Indicates call forwarding when no answer is received from a Cisco IP phone.
destination-pattern	Destination pattern (telephone number) configured for this dial peer.
dial-peer voice	Voice dial peer.
port	(Virtual) voice port designator.

Related Commands

Command	Description
show call-manager- fallback all	Displays the detailed configuration of all Cisco IP phones, directory numbers, voice ports, and dial peers in your network during Cisco Unified Communications Manager fallback.
show call-manager- fallback ephone-dn	Displays output for the Cisco IP phone directory numbers or virtual voice ports during Cisco Unified Communications Manager fallback.
show call-manager- fallback voice-port	Displays output for the voice ports while Cisco Unified Communications Manager is active.

show call-manager-fallback ephone-dn

To display detailed configuration output for the Cisco IP phone directory numbers or virtual voice ports during Cisco Unified CallManager fallback, use the **show call-manager-fallback ephone-dn** command in privileged EXEC mode.

show call-manager-fallback ephone-dn

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

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; 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.
12.3(4)T	Cisco SRST 3.0	The Version, ephone-dn,vand voice-port was added to output.

Examples

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

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

CONFIG (Version=4.1(0))
=====================
Version 4.1(0)
For on-line documentation please see:
www.cisco.com/univercd/cc/td/doc/product/access/ip_ph/ip_ks/index.htm
ip source-address 0.0.0.0 port 2000
max-ephones 0
max-dn 0
max-conferences 8 gain -6
dspfarm units 0
dspfarm transcode sessions 0
huntstop
cnf-file location: system:
cnf-file option: PER-PHONE-TYPE
```

```
network-locale[0] US (This is the default network locale for this box)
network-locale[1] US
network-locale[2] US
network-locale[3] US
network-locale[4] US
user-locale[0] US (This is the default user locale for this box)
user-locale[1] US
user-locale[2] US
user-locale[3] US
user-locale[4] US
srst mode auto-provision is OFF
srst ephone template is 0
srst dn template is 0
srst dn line mode is single
time-format 12
date-format mm-dd-yy
timezone 0 Greenwich Standard Time
no transfer-pattern is configured, transfer is restricted to local SCCP phones only.
keepalive 30 auxiliary 30
timeout interdigit 10
timeout busy 10
timeout ringing 180
timeout ringin-callerid 8
caller-id name-only: enable
Limit number of DNs per phone:
7910: 36
7935: 36
7936: 36
7940: 36
7960: 36
7970: 36
Log (table parameters):
max-size: 150
retain-timer: 15
transfer-system full-consult
local directory service: enabled.
Extension-assigner tag-type ephone-tag.
========================================
```

Table 3: show call-manager-fallback ephone-dn Field Descriptions, on page 46 describes the significant fields shown in the display.

Table 3: show call-manager-fallback ephone-dn Field Descriptions

Field	Description
ephone-dn	Cisco IP phone directory number.
(no) huntstop	Whether or not huntstop is set.
number	Cisco IP phone number.
translate called	The configured translation rule. Can be called or calling, plus the tag number by which the rule set is referenced.

Related Commands

Command	Description
show call-manager- fallback all	Displays the detailed configuration of all Cisco IP phones, directory numbers, voice ports, and dial peers in your network during Cisco Unified CallManager fallback.
show call-manager- fallback dial-peer	Displays detailed configuration output for the dial peers in your network during Cisco Unified CallManager fallback.
show call-manager- fallback voice-port	Displays output for the voice ports while Cisco Unified CallManager is active.

show call-manager-fallback voice-port

To display detailed configuration output for the voice ports while Cisco Unified Communications Manager is active, use the **show call-manager-fallback voice-port** command in privileged EXEC mode.

show call-manager-fallback voice-port

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

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.

Examples

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

```
Router# show call-manager-fallback voice-port
voice-port 50/0/1
 station-id number 4444
 timeout ringing 8
 translate called 1
!
voice-port 50/0/2
 station-id number 3333
 timeout ringing 8
 translate called 1
!
voice-port 50/0/3
 station-id number 5555
 timeout ringing 8
 translate called 1
!
voice-port 50/0/4
 timeout ringing 8
 translate called 1
!
```

Table 4: show call-manager-fallback voice-port Field Descriptions, on page 49describes the significant fields shown in the display.

Table 4: show call-manager-fallback voice-port Field Descriptions

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

Related Commands

Command	Description
show call-manager- fallback all	Displays the detailed configuration of all Cisco IP phones, directory numbers, voice ports, and dial peers in your network during Cisco Unified Communications Manager fallback.
show call-manager- fallback dial-peer	Displays detailed configuration output for the dial peers in your network during Cisco Unified Communications Manager fallback.
show call-manager- fallback ephone-dn	Displays output for the Cisco IP phone directory numbers or virtual voice ports during Cisco Unified Communications Manager fallback.

show credentials

To display the credentials settings that have been configured for use during Cisco Unified CME phone authentication communications or secure Cisco Unified SRST fallback, use the **show credentials command in** privileged EXEC mode.

show credentials

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

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 displays the credentials settings on a Cisco Unified CME router that has been configured with a CTL provider to be used with Cisco Unified CME phone authentication.

Cisco Unified SRST

This command displays the credentials settings on the Cisco Unified SRST router that are supplied to Cisco Unified Communications Manager for use during secure SRST fallback.

Examples

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

```
Router# show credentials

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

Table 5: show credentials Field Descriptions, on page 50 describes the fields in the sample output.

Table 5: show credentials Field Descriptions

Field	Description
Credentials IP	Cisco Unified CME—IP address where the CTL provider is configured. Cisco Unified SRST—The specified IP address where certificates from Cisco Unified Communications Manager to the SRST router are received.
Credentials PORT	Cisco Unified CME—TCP port for credentials service communication. Default is 2444. Cisco Unified SRST—The port to which the SRST router connects to receive messages from the Cisco Unified IP phones. The port number is from 2000 to 9999. The default port number is 2445.

Field	Description
Trustpoint	Cisco Unified CME—CTL provider trustpoint label that will be used for TLS sessions with the CTL client. Cisco Unified SRST—The name of the trustpoint that is associated with the credentials service between the Cisco Unified Communications Manager client and the SRST router.

Related Commands

Command	Description
credentials	Enters credentials configuration mode to configure a Cisco Unified CME CTL provider certificate or a Cisco Unified 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 a Cisco Unified 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.
trustpoint (credentials)	Specifies the name of the trustpoint to be associated with a Cisco Unified CME CTL provider certificate or with a Cisco Unified SRST router certificate.

show ephone

To display information about registered Cisco Unified IP phones, use the **show ephone** command in privileged EXEC mode.

show ephone [{*mac-addressphone-type*}]

Syntax Description

mac-address	(Optional) Displays information for the phone with the specified MAC address.
phone-type	(Optional) Displays information for phones of the specified phone type. Valid types are as follows: • **7902**—Cisco Unified IP Phone 7902G. • **7905**—Cisco Unified IP Phone 7905G. • **7906**—Cisco Unified IP Phone 7905G. • **7910**—Cisco Unified IP Phone 7910 and 7910G. • **7911**—Cisco Unified IP Phone 7911G. • **7912—**Cisco Unified IP Phone 7912G. • **7914—**Cisco Unified IP Phone 7914 Expansion Module. • **7920—**Cisco Unified Wireless IP Phone 7920. • **7921**—Cisco Unified Wireless IP Phone 7921. • **7931—**Cisco Unified Wireless IP Phone 7931G. • **7935—**Cisco Unified IP Conference Station 7935. • **7936—**Cisco Unified IP Conference Station 7936. • **7940—**Cisco Unified IP Phones 7940 and 7940G. • **7941**—Cisco Unified IP Phone 7941G. • **7941GE**—Cisco Unified IP Phone 7941G-GE. • **7942**—Cisco Unified IP Phone 7942. • **7945**—Cisco Unified IP Phone 7945. • **7960—**Cisco Unified IP Phones 7960 and 7960G**.** • **7961**—Cisco Unified IP Phone 7961G. • **7961GE**—Cisco Unified IP Phone 7961G-GE. • **7962**—Cisco Unified IP Phone 7962. • **7965**—Cisco Unified IP Phone 7965. • **7970**—Cisco Unified IP Phone 7970G. • **7971**—Cisco Unified IP Phone 7971G-GE. • **7975**—Cisco Unified IP Phone 7975 • **7985**—Cisco Unified IP Phone 7985. • **ata—**Cisco ATA-186 or Cisco ATA-188.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco ITS 1.0 Cisco SRST 1.0	This command was introduced on the Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(2)XT	Cisco ITS 2.0 Cisco SRST 2.0	This command was implemented on the Cisco 1750 and Cisco 1751.
12.2(8)T	Cisco ITS 2.0 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.
12.2(8)T1	Cisco ITS 2.0 Cisco SRST 2.0	This command was implemented on the Cisco 2600XM and Cisco 2691.
12.2(11)T	Cisco ITS 2.01 Cisco SRST 2.01	The **ata** keyword was added and this command was implemented on the Cisco 1760.
12.2(11)YT	Cisco ITS 2.1 Cisco SRST 2.1	The **7914** keyword was added.
12.2(15)ZJ	Cisco CME 3.0 Cisco SRST 3.0	The **7902**, **7905**, and **7912** keywords were added.
12.3(7)T	Cisco CME 3.1 Cisco SRST 3.1	The **7920** and **7936** keywords were added.
12.3(11)XL	Cisco CME 3.2.1 Cisco SRST 3.2.1	The **7970** keyword was added.
12.3(14)T	Cisco CME 3.3 Cisco SRST 3.3	The **7971** keyword was added, and this command was integrated into Cisco IOS Release 12.3(14)T.
12.4(4)XC	Cisco Unified CME 4.0 Cisco Unified SRST 4.0	The **7911**, **7941**, **7941GE**, **7961**, and **7961GE** keywords were added.
12.4(9)T	Cisco Unified CME 4.0 Cisco Unified SRST 4.0	The **7911**, **7941**, **7941GE**, **7961**, and **7961GE** keywords were integrated into Cisco IOS Release 12.4(9)T.
12.4(6)XE	Cisco Unified CME 4.0(2)	The **7931** keyword was added for Cisco Unified CME.
12.4(4)XC4	Cisco Unified CME 4.0(3)	The **7931** keyword was added for Cisco Unified CME.
12.4(11)T	Cisco Unified CME 4.0(3)	The **7931** keyword for Cisco Unified CME was integrated in Cisco IOS Release 12.4(11)T.
12.4(11)XJ2	Cisco Unified CME 4.1 Cisco Unified SRST 4.1	The **7921** and **7985** keywords were introduced.
12.4(15)T1	Cisco Unified CME 4.1(1) Cisco Unified SRST 4.1(1)	The **7942**, **7945**, **7962**, **7965**, and **7975** keywords were introduced.
12.4(11)XW3	Cisco Unified CME 4.2 Cisco Unified SRST 4.2	The **7942**, **7945**, **7962**, **7965**, and **7975** keywords were introduced.

Cisco IOS Release	Cisco Product	Modification
12.4(15)XY	Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1)	Emergency response location (ERL) information displays in the output.
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.

Examples

Significant fields in the output from this command are described in Table 6: show ephone Field Descriptions, on page 55 .

The following sample output shows general information for registered phones:

```
Router# show ephone
ephone-1 Mac:0003.E3E7.F627 TCP socket:[2] activeLine:0 REGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
IP:10.0.0.2 51671 Telecaster 7940 keepalive 28 max_line 2
button 1: dn 1 number 4444
ephone-2 Mac:0030.94C3.F43A TCP socket:[1] activeLine:0 REGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
IP:10.0.0.3 50094 Telecaster 7960 keepalive 28 max_line 6
button 1: dn 3 number 5555
ephone-3 Mac:0003.6B40.99DA TCP socket:[3] activeLine:0 REGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
IP:10.2.168.200 51879 Telecaster 7960 keepalive 28 max_line 6
button 1: dn 2 number 3333
```

The following sample output shows general information for the phone with the MAC address 0003.E3E7.F627:

```
Router# show ephone 0003.E3E7.F627
ephone-1 Mac:0003.E3E7.F627 TCP socket:[2] activeLine:0 REGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
IP:10.0.0.2 51671 Telecaster 7940 keepalive 28 max_line 2
button 1: dn 1 number 4444
Active Call on DN 1:3001 10.0.0.51 31808 to 10.2.159.100 22708
Tx Pkts 452 bytes 41584 Rx Pkts 452 bytes 41584 Lost 0
Jitter 0 Latency 0
```

The following sample output shows information for a phone that has two Cisco Unified IP Phone 7914 Expansion Modules attached. The output shows this module as a subsidiary type in addition to the main **7960** type for the phone itself. Subtype 3 means that one Cisco Unified IP Phone 7914 Expansion Module is attached to the main Cisco Unified IP Phone 7960 or 7960G, and subtype 4 means that two are attached.

```
Router# show ephone 7914
ephone-2 Mac:0007.0EA6.39F8 TCP socket:[2] activeLine:0 REGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
IP:10.2.205.206 49278 Telecaster 7960  sub=4 keepalive 2723 max_line 34
button 1: dn 21 number 60021 CH1 IDLE
button 2: dn 22 number 60022 CH1 IDLE
button 7: dn 11 number 60011 CH1 IDLE      monitor-ring
button 8: dn 12 number 60012 CH1 IDLE      monitor-ring
button 9: dn 13 number 60013 CH1 IDLE      monitor-ring
button 10: dn 14 number 60014 CH1 IDLE      monitor-ring
button 11: dn 15 number 60015 CH1 IDLE      monitor-ring
button 12: dn 16 number 60016 CH1 IDLE      monitor-ring
button 13: dn 17 number 60017 CH1 IDLE      monitor-ring
```

```
button 14: dn 18 number 60018 CH1 IDLE      monitor-ring
button 15: dn 19 number 60019 CH1 IDLE      monitor-ring
button 16: dn 20 number 60020 CH1 IDLE      monitor-ring
button 17: dn 39 number 60039 CH1 IDLE      CH2 IDLE      monitor-ring
button 18: dn 40 number 60040 CH1 IDLE      CH2 IDLE      monitor-ring
button 19: dn 23 number 60023 CH1 IDLE      monitor-ring
button 20: dn 24 number 60024 CH1 IDLE      monitor-ring
button 21: dn 25 number 60025 CH1 IDLE      monitor-ring
button 22: dn 26 number 60026 CH1 IDLE      monitor-ring
button 23: dn 27 number 60027 CH1 IDLE      monitor-ring
button 24: dn 28 number 60028 CH1 IDLE      monitor-ring
button 25: dn 29 number 60029 CH1 IDLE      monitor-ring
button 26: dn 30 number 60030 CH1 IDLE      monitor-ring
button 27: dn 31 number 60031 CH1 IDLE      CH2 IDLE      monitor-ring
button 28: dn 32 number 60032 CH1 IDLE      CH2 IDLE      monitor-ring
button 29: dn 33 number 60033 CH1 IDLE      CH2 IDLE      monitor-ring
button 30: dn 34 number 60034 CH1 IDLE      CH2 IDLE      monitor-ring
button 31: dn 35 number 60035 CH1 IDLE      CH2 IDLE      monitor-ring
button 32: dn 36 number 60036 CH1 IDLE      CH2 IDLE      monitor-ring
button 33: dn 37 number 60037 CH1 IDLE      CH2 IDLE      monitor-ring
button 34: dn 38 number 60038 CH1 IDLE      CH2 IDLE      monitor-ring
```

The following sample output shows a phone that has a paging-dn and has received a page:

```
Router# show ephone 7910
ephone-2 Mac:0087.0E76.B93C TCP socket:[4] activeLine:0 REGISTERED
mediaActive:1 offhook:0 ringing:0 reset:0 reset_sent:0 paging 1 debug:0
IP:10.50.50.20 49231 Telecaster 7910  keepalive 112 max_line 2 dual-line
button 1:dn 3  number 95021 CH1 IDLE
paging-dn 25
```

Table 6: show ephone Field Descriptions, on page 55 describes significant fields in the output.

Table 6: show ephone Field Descriptions

Field	Description
Active Call	An active call is in progress.
activeLine	Line (button) on the phone that is in use. Zero indicates that no line is in use.
auto-dial *number*	Intercom extension that automatically dials *number*.
button *number*: dn *number*	Phone button number and the extension (ephone-dn) dn-tag number associated with that button.
bytes	Total number of voice data bytes sent or received by the phone.
Called Dn, Calling Dn	Ephone-dn tag numbers of the called and calling ephone-dn. Set to -1 if the call is not to or from an ephone-dn, or if there is no active call.
cfa *number*	Call-forward-all to *number* is enabled for this extension.
CH1 CH2	Status of channel 1 and, if this is a dual-line ephone-dn, the status of channel 2.
debug	1 indicates that debug for the phone is enabled. 0 indicates that debug is disabled.
DnD	Do Not Disturb is set on this phone.

Field	Description
DP tag	Not used.
ephone-*number*	Unique sequence number used to identify this phone during configuration (phone-tag).
IP	Assigned IP address of the Cisco Unified IP phone.
Jitter	Amount of variation (in milliseconds) of the time interval between voice packets received by the Cisco Unified IP phone.
keepalive	Number of keepalive messages received from the Cisco Unified IP phone by the router.
Latency	Estimated playout delay for voice packets received by the Cisco Unified IP phone.
line *number*	Button number on an IP phone. Line 1 is the button nearest the top of the phone.
Lost	Number of voice packets lost, as calculated by the Cisco Unified IP phone, on the basis of examining voice packet time-stamp and sequence numbers during playout.
Mac	MAC address.
Max Conferences	Maximum number of allowable conference calls and number of active conference calls.
max_line *number*	Maximum number of line buttons that can be configured on this phone.
mediaActive	1 indicates that an active conversation is in progress. 0 indicates that no conversation is ongoing.
monitor-ring	This button is set up as a monitor button.
number	Telephone or extension number associated with the Cisco Unified IP phone button and its dn-tag.
offhook	1 indicates that the phone is off-hook. 0 indicates that the phone is on-hook.
overlay	This button contains an overlay set. Use **show ephone overlay** to display the contents of overlay sets.
paging	1 indicates that the phone has received an audio page. 0 indicates that the phone has not received an audio page.
paging-dn	Ephone-dn that is dedicated for receiving audio pages on this phone. The paging-dn number is the number of the paging set to which this phone belongs.
Password	Authentication string that the phone user types when logging in to the web-based Cisco Unified CME GUI.
Port	Port used for TAPI transmissions.

Field	Description
REGISTERED	The Cisco Unified IP phone is active and registered. Alternative states are UNREGISTERED (indicating that the connection to the Cisco Unified IP phone was closed in a normal manner) and DECEASED (indicating that the connection to the Cisco Unified IP phone was closed because of a keepalive timeout).
reset	Pending reset.
reset_sent	Request for reset has been sent to the Cisco Unified IP phone.
ringing	1 indicates that the phone is ringing. 0 indicates that the phone is not ringing.
Rx Pkts	Number of received voice packets.
silent-ring	Silent ring has been set on this button and extension.
socket	TCP socket number used to connect to IP phone.
speed dial *speed-tag:digit-string label-text*	This button is a speed-dial button, assigned to the speed-dial sequence number *speed-tag*. It dials *digit-string* and displays the text *label-text* next to the button.
sub=3, sub=4	Subtype 3 means that one Cisco Unified IP Phone 7914 Expansion Module is attached to the main Cisco Unified IP Phones 7960 and 7960G, and subtype 4 means that two are attached.
Tag *number*	Dn-tag number, the unique sequence number that identifies an ephone-dn during configuration, followed by the type of ephone-dn it is.
TAPI Client IP Address	IP address of the PC running the TAPI client.
TCP socket	TCP socket number used to communicate with the Cisco Unified IP phone. This can be correlated with the output of other debug and show commands.
Telecaster *model-number*	Type and model of the Cisco Unified IP phone. This information is received from the phone during its registration with the router.
Tx Pkts	Number of transmitted voice packets.
Username	Username that the phone user types when logging in to the web-based Cisco Unified CME GUI.

Related Commands

Command	Description
show ephone-dn	Displays information about Cisco Unified IP phone extensions (ephone-dns).
show ephone login	Displays the login states of all local ephones.
show telephony-service all	Displays systemwide status and information for a Cisco Unified CME system.

show ephone cfa

To display status and information on the registered phones that have call-forward-all set on one or more of their extensions (ephone-dns), use the **show ephone cfa** command in privileged EXEC mode.

show ephone cfa

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

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

Examples

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

```
Router# show ephone cfa
ephone-1 Mac:0007.0EA6.353A TCP socket:[2] activeLine:0 REGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
IP:1.2.205.205 52491 Telecaster 7960  keepalive 14 max_line 6
button 1: dn 11 number 60011 cfa 60022 CH1 IDLE
button 2: dn 17 number 60017 cfa 60021 CH1 IDLE
```

The **show ephone** command describes significant fields in this output.

Related Commands

Command	Description
show ephone	Displays statistical information about registered Cisco IP phones.

show ephone dn

To display phone information for specified dn-tag or for all dn-tags, use the **show ephone dn** command in privileged EXEC mode.

show ephone dn [*dn-tag*]

Syntax Description

dn-tag	(Optional) Unique sequence number that is used during configuration to identify a particular extension (ephone-dn).

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

Use this command to identify the phone on which a particular dn-tag has been assigned.

Examples

The following is sample output for the two appearances of DN 5:

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

The **show ephone** command describes significant fields in this output.

Related Commands

Command	Description
show ephone	Displays statistical information about registered Cisco IP phones.

show ephone dnd

To display information on the registered phones that have "do not disturb" set on one or more of their extensions (ephone-dns), use the **show ephone dnd** command in privileged EXEC mode.

show ephone dnd

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

This command does not apply to Cisco Unified SRST.

Examples

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

```
Router# show ephone dnd
ephone-1 Mac:0007.0EA6.353A TCP socket:[1] activeLine:0 REGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
IP:1.2.205.205 52486 Telecaster 7960  keepalive 2729 max_line 6 DnD
button 1: dn 11 number 60011 CH1 IDLE
```

The **show ephone** command describes significant fields in this output.

Related Commands

Command	Description
show ephone	Displays statistical information about registered Cisco IP phones.

show ephone login

To display the login states of all local IP phones, use the **show ephone login** command in privileged EXEC mode.

show ephone login

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

The **show ephone login** command displays whether an ephone has a personal identification number (PIN) and whether its owner has logged in.

Examples

The following is sample output from the **show ephone login** command. It shows that a PIN is enabled for ephone 1 and that its owner has not logged in. The other phones do not have PINs associated with them.

```
Router# show ephone login
ephone 1        Pin enabled:TRUE        Logged-in:FALSE
ephone 2        Pin enabled:FALSE
ephone 3        Pin enabled:FALSE
ephone 4        Pin enabled:FALSE
ephone 5        Pin enabled:FALSE
ephone 6        Pin enabled:FALSE
ephone 7        Pin enabled:FALSE
ephone 8        Pin enabled:FALSE
ephone 9        Pin enabled:FALSE
```

Table 7: show ephone login Field Descriptions, on page 61 describes significant fields in this output.

Table 7: show ephone login Field Descriptions

Field	Description
ephone *phone-tag*	Phone identified with its unique phone-tag sequence number.
Pin enabled	TRUE indicates that a PIN has been defined for this phone. FALSE indicates that no PIN has been defined for this phone.
Logged-in	TRUE indicates that a phone user is currently logged in on this phone. FALSE indicates that no phone user is currently logged in on this phone.

Related Commands

Command	Description
login (telephony-service)	Defines when users of IP phones in a Cisco Unified CME system are logged out automatically.
pin	Sets set a personal identification number (PIN) for an IP phone in a Cisco Unified CME system.
show ephone	Displays statistical information about registered Cisco IP phones.

show ephone moh

To display information about moh files in use, use the **show ephone moh** command in global configuration mode.

show ephone moh

Syntax Description

This command has no arguments or keywords

Command Modes

Global Configuration mode.

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

Use the show ephone moh to display information about the different MOH group configured. The following examples displays different MOH group configured.

Examples

```
Router #show ephone moh
Skinny Music On Hold Status (moh-group 1)
Active MOH clients 0 (max 830), Media Clients 0
File flash:/minuet.au (not cached) type AU Media_Payload_G711Ulaw64k 160 bytes
Moh multicast 239.10.16.6 port 2000
Skinny Music On Hold Status (moh-group 2)
Active MOH clients 0 (max 830), Media Clients 0
File flash:/audio/hello.au type AU Media_Payload_G711Ulaw64k  160 bytes
Moh multicast on 239.10.16.6 port 2000 via 0.0.0.0
Skinny Music On Hold Status (moh-group 3)
Active MOH clients 0 (max 830), Media Clients 0
File flash:/bells.au type AU Media_Payload_G711Ulaw64k  160 bytes
Moh multicast on 239.10.16.5 port 2000 via 0.0.0.0
Skinny Music On Hold Status (moh-group 4)
Active MOH clients 0 (max 830), Media Clients 0
File flash:/3003.au type AU Media_Payload_G711Ulaw64k  160 bytes
Moh multicast on 239.10.16.7 port 2000 via 0.0.0.0
Skinny Music On Hold Status (moh-group 5)
Active MOH clients 0 (max 830), Media Clients 0
File flash:/4004.au type AU Media_Payload_G711Ulaw64k  160 bytes
Moh multicast on 239.10.16.8 port 2000 via 0.0.0.0
```

Related Commands

Command	Description
show ephone-dn	Displays MOH group information for a phone directory number.
show ephone summary	Displays the information about the MOH files in use
show voice moh-group statistics	Displays the MOH subsystem statistics information

show ephone offhook

To display information and packet counts for the phones that are currently off hook, use the **show ephone offhook** command in privileged EXEC mode.

show ephone offhook

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

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

Examples

The following sample output is displayed when no phone is off hook:

```
Router# show ephone offhook
No ephone in specified type/condition.
```

The following sample output displays information for a phone that is off hook:

```
Router# show ephone offhook
ephone-5 Mac:000A.8A2C.8C6E TCP socket:[20] activeLine:1 REGISTERED
mediaActive:0 offhook:1 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
IP:10.22.84.71 51228 Telecaster 7960  keepalive 43218 max_line 6
button 1:dn 9  number 59943 CH1 SIEZE     silent-ring
button 2:dn 10 number 59943 CH1 IDLE
button 3:dn 42 number A4400  auto dial A4500 CH1 IDLE
button 4:dn 96 number 69943  auto dial 95259943 CH1 IDLE
button 5:dn 75 number 49943  auto dial 49943 CH1 IDLE
speed dial 1:57514 marketing
Active Call on DN 9 chan 1 :59943 0.0.0.0 0 to 0.0.0.0 2000 via 172.30.151.1
G711Ulaw64k  160 bytes vad
Tx Pkts 0 bytes 0 Rx Pkts 0 bytes 0 Lost 0
Jitter 0 Latency 0 callingDn -1 calledDn -1
Username:user1 Password:newuser
```

The following sample output displays information for a phone that has just completed a call:

```
Router# show ephone offhook
ephone-5 Mac:000A.8A2C.8C6E TCP socket:[20] activeLine:1 REGISTERED
mediaActive:1 offhook:1 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
IP:10.22.84.71 51228 Telecaster 7960  keepalive 43224 max_line 6
button 1:dn 9  number 59943 CH1 CONNECTED silent-ring
button 2:dn 10 number 59943 CH1 IDLE
button 3:dn 42 number A4400  auto dial A4500 CH1 IDLE
button 4:dn 96 number 69943  auto dial 95259943 CH1 IDLE
button 5:dn 75 number 49943  auto dial 49943 CH1 IDLE
speed dial 1:57514 marketing
```

```
Active Call on DN 9 chan 1 :59943 10.23.84.71 22926 to 172.30.131.129 2000 via 172.30.151.1
G711Ulaw64k  160 bytes no vad
Tx Pkts 0 bytes 0 Rx Pkts 0 bytes 0 Lost 0
Jitter 0 Latency 0 callingDn -1 calledDn -1 (media path callID 19288 srcCallID 1
9289)
Username:user1 Password:newuser
```

The **show ephone** command describes significant fields in this output.

Related Commands

Command	Description
show ephone	Displays statistical information about registered Cisco IP phones.

show ephone overlay

To display information for the registered phones that have overlay ephone-dns associated with them, use the **show ephone overlay** in privileged EXEC mode.

show ephone overlay

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

This command does not apply to Cisco Unified SRST.

Examples

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

```
Router# show ephone overlay
ephone-1 Mac:0007.0EA6.353A TCP socket:[1] activeLine:0 REGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
IP:10.2.225.205 52486 Telecaster 7960  keepalive 2771 max_line 6
button 1: dn 11 number 60011 CH1 IDLE      overlay
button 2: dn 17 number 60017 CH1 IDLE      overlay
button 3: dn 24 number 60024 CH1 IDLE      overlay
button 4: dn 30 number 60030 CH1 IDLE      overlay
button 5: dn 36 number 60036 CH1 IDLE      CH2 IDLE      overlay
button 6: dn 39 number 60039 CH1 IDLE      CH2 IDLE      overlay
overlay 1: 11(60011) 12(60012) 13(60013) 14(60014) 15(60015) 16(60016)
overlay 2: 17(60017) 18(60018) 19(60019) 20(60020) 21(60021) 22(60022)
overlay 3: 23(60023) 24(60024) 25(60025) 26(60026) 27(60027) 28(60028)
overlay 4: 29(60029) 30(60030) 31(60031) 32(60032) 33(60033) 34(60034)
overlay 5: 35(60035) 36(60036) 37(60037)
overlay 6: 38(60038) 39(60039) 40(60040)
```

The **show ephone** command describes significant fields in this output. Table 8: show ephone overlay Field Descriptions, on page 66 describes a field that is not in that table.

Table 8: show ephone overlay Field Descriptions

Field	Description
overlay *number*	Displays the contents of an overlay set, including each dn-tag and its associated extension number.

Related Commands

Command	Description
show ephone	Displays statistical information about registered Cisco IP phones.

show ephone phone-load

To display information about the phone firmware that is loaded on registered phones, use the **show ephone phone-load** command in privileged EXEC mode.

show ephone phone-load

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

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

Examples

The following is sample output that displays the phone firmware versions for all phones in the system:

```
Router# show ephone phone-load
DeviceName        CurrentPhoneload       PreviousPhoneload      LastReset
======================================================================
SEP0002B9AFC49F   3.2(2.14)              3.2(2.14)              TCP-timeout
SEP003094C2D0B0   3.2(2.14)              3.2(2.14)              TCP-timeout
SEP000C30F03707   3.2(2.14)              3.2(2.14)              TCP-timeout
SEP003094C2999F   3.2(2.14)              3.2(2.14)              TCP-timeout
SEP000A8A2C8C6E   3.2(2.14)              3.2(2.14)              Initialized
SEP0002B9AFBB4D   3.2(2.14)              3.2(2.14)              TCP-timeout
SEP00075078627F   3.2(2.14)              3.2(2.14)              TCP-timeout
SEP0002FD659E59   3.2(2.14)              3.2(2.14)              TCP-timeout
SEP00024BCCD626   3.2(2.14)                                     CM-closed-TCP
SEP0008215F88C1   3.2(2.14)              3.2(2.14)              TCP-timeout
SEP000C30F0390C   3.2(2.14)              3.2(2.14)              TCP-timeout
SEP003094C30143   3.2(2.14)              3.2(2.14)              TCP-timeout
```

Table 9: show ephone phone-load Field Descriptions, on page 68describes significant fields in this output.

Table 9: show ephone phone-load Field Descriptions

Field	Description
DeviceName	Device name.
CurrentPhoneLoad	Current phone firmware version.
PreviousPhoneLoad	Phone firmware version before last phone load.
LastReset	Reason for last reset of phone.

Related Commands

Command	Description
show ephone	Displays statistical information about registered Cisco IP phones.

show ephone registered

To display the status of registered phones, use the **show ephone registered** command in privileged EXEC mode.

show ephone registered

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

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

Examples

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

```
Router# show ephone registered
ephone-2 Mac:000A.8A5C.5961 TCP socket:[1] activeLine:0 REGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
IP:192.168.0.50 50349 Telecaster 7940  keepalive 23738 max_line 2
button 1: dn 2  number 91450 CH1 IDLE      CH2 IDLE
button 2: dn 0   --
button 3: dn 0   --
button 4: dn 0   --
button 5: dn 0   --
button 6: dn 0   --
```

The **show ephone** command describes significant fields in this output.

Related Commands

Command	Description
show ephone	Displays statistical information about registered Cisco IP phones.

show ephone remote

To display nonlocal phones (phones with no Address Resolution Protocol [ARP] entry), use the **show ephone remote** command in privileged EXEC mode.

show ephone remote

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

Phones without ARP entries are suspected not to be on the LAN. Use the **show ephone remote** command to identify phones without ARP entries that might have operational issues.

Examples

The following is sample output that identifies ephone 2 as not having an ARP entry:

```
Router# show ephone remote
ephone-2 Mac:0185.047C.993E TCP socket:[4] activeLine:0 REGISTERED
mediaActive:1 offhook:0 ringing:0 reset:0 reset_sent:0 paging 1 debug:0
IP:10.50.50.20 49231 Telecaster 7910  keepalive 112 max_line 2 dual-line
button 1:dn 3  number 95021 CH1 IDLE
paging-dn 25
```

The **show ephone** command describes significant fields in this output.

Related Commands

Command	Description
show ephone	Displays statistical information about registered Cisco IP phones.

show ephone ringing

To display information on phones that are ringing, use the **show ephone ringing** command in privileged EXEC mode.

show ephone ringing

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

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

Examples

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

```
Router# show ephone ringing
ephone-1 Mac:0005.5E37.8090 TCP socket:[1] activeLine:0 REGISTERED
mediaActive:0 offhook:0 ringing:1 reset:0 reset_sent:0 paging 0 debug:0
IP:10.50.50.10 49329 Telecaster 7960  keepalive 17602 max_line 6
button 1:dn 1  number 95011 CH1 RINGING   CH2 IDLE
button 2:dn 2  number 95012 CH1 IDLE
```

The **show ephone** command describes significant fields in this output.

Related Commands

Command	Description
show ephone	Displays statistical information about registered Cisco IP phones.

show ephone summary

To display brief information about Cisco IP phones, use the **show ephone summary** command in privileged EXEC mode.

show ephone summary

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC (#)

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco CME 1.0 Cisco SRST 1.0	This command was introduced.
12.2(8)T	Cisco CME 2.0 Cisco SRST 2.0	This command was integrated into Cisco IOS Release 12.2(8)T .
15.0(1)XA	Cisco Unified CME 8.0 Cisco Unified SRST 8.0	This command was modified. The output was enhanced to show IPv6 or IPv4 addresses configured on ephones.
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.
15.1(2)T	Cisco Unified CME 8.1 Cisco Unified SRST 8.1	This command was modified. The output was enhanced to show voice-class stun-usage information.

Examples

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

```
Router# show ephone summary
hairpin_block:
ephone-1[0] Mac:FCAC.3BAE.0000 TCP socket:[17] activeLine:0 whisperLine:0 REGISTERED
mediaActive:0 whisper_mediaActive:0 startMedia:0 offhook:0 ringing:0 reset:0 reset_sent:0
debug:0  primary_dn: 1*
IP:10.2.1.0 * SCCP Gateway (AN)  keepalive 2966   music 0  1:1
port 0/0/0
voice-class stun is enabled
ephone-2[1] Mac:FCAC.3BAE.0001 TCP socket:[18] activeLine:0 whisperLine:0 REGISTERED
mediaActive:0 whisper_mediaActive:0 startMedia:0 offhook:0 ringing:0 reset:0 reset_sent:0
debug:0  primary_dn: 2*
IP:10.2.1.5 * SCCP Gateway (AN)  keepalive 2966   music 0  1:2
port 0/0/1
voice-class stun is enabled
ephone-4 Mac:0030.94C3.F43A TCP socket:[-1] activeLine:0 REGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 debug:0
IP:10.2.1.1 Telecaster 7960  keepalive 59
Max 48, Registered 1, Unregistered 0, Deceased 0, Sockets 1
Max Conferences 4 with 0 active (4 allowed)
Skinny Music On Hold Status
```

```
Active MOH clients 0 (max 72), Media Clients 0
No MOH file loaded
```

The **show ephone** command describes significant fields in this output.

Command	Description
show ephone	Displays statistical information about registered Cisco IP phones.

show ephone tapiclients

To display status of ephone Telephony Application Programming Interface (TAPI) clients, use the **show ephone tapiclients** command in privileged EXEC mode.

show ephone tapiclients

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

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

Examples

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

```
Router# show ephone tapiclients
ephone-4 Mac:0007.0EA6.39F8 TCP socket:[2] activeLine:0 REGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
IP:192.168.1.18 50291 Telecaster 7960  sub=3 keepalive 728 max_line 20
button 1:dn 6  number 1004 CH1 IDLE      CH2 IDLE
button 2:dn 1  number 1000 CH1 IDLE      shared
button 3:dn 2  number 1000 CH1 IDLE      shared
button 7:dn 3  number 1001 CH1 IDLE      CH2 IDLE      monitor-ring shared
button 8:dn 4  number 1002 CH1 IDLE      CH2 IDLE      monitor-ring shared
button 9:dn 5  number 1003 CH1 IDLE      CH2 IDLE      monitor-ring
button 10:dn 91 number A00  auto dial A01 CH1 IDLE
speed dial 1:2000 PAGE-STAFF
speed dial 2:2001 HUNT-STAFF
paging-dn 90
Username:userB Password:ge30qe
Tapi client information
Username:userB status:REGISTERED Socket :[5]
 Tapi Client IP address: 192.168.1.5  Port:2295
```

The **show ephone** command describes significant fields in this output.

Related Commands

Command	Description
show ephone	Displays statistical information about registered Cisco IP phones.

show ephone telephone-number

To display information for the phone associated with a specified number, use the **show ephone telephone-number** command in privileged EXEC mode.

show ephone telephone-number *number*

Syntax Description

number	Telephone number that is associated with an ephone.

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

Use this command to find the phone on which a particular telephone number appears.

Examples

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

```
Router# show ephone telephone-number 91400
DP tag: 0, primary
Tag 1, Normal or Intercom dn
  ephone 1, mac-address 000A.0E51.19F0, line 1
```

The **show ephone** command describes significant fields in this output.

Related Commands

Command	Description
show ephone	Displays statistical information about registered Cisco IP phones.

show ephone unregistered

To display information about unregistered phones, use the **show ephone unregistered** command in privileged EXEC mode.

show ephone unregistered

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

There are two ways that an ephone can become unregistered. The first way is when an ephone is listed in the running configuration but no physical device has been registered for that ephone. The second way is when an unknown device was registered at some time after the last router reboot but has since unregistered.

Examples

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

```
Router# show ephone unregistered
ephone-1 Mac:0007.0E81.10F0 TCP socket:[-1] activeLine:0 UNREGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
IP:0.0.0.0 0 Unknown 0  keepalive 0 max_line 0
```

The **show ephone** command describes significant fields in this output.

Related Commands

Command	Description
show ephone	Displays statistical information about registered Cisco IP phones.

show ephone-dn

To display status and information for a Cisco IP phone destination number or for extensions (ephone-dns) in a Cisco Unified Communications Manager Express (Cisco Unified CME) or Cisco Unified Survivable Remote Site Telephony (SRST) environment, use the **show ephone-dn** command in privileged EXEC mode.

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

Syntax Description

dn-tag	(Optional) For Cisco Unified CME, a unique sequence number that is used during configuration to identify a particular extension (ephone-dn). (Optional) For Cisco Unified SRST, a destination number tag. The destination number can be from 1 to 288.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco CME 1.0 Cisco SRST 1.0	This command was introduced on the Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(2)XT	Cisco CME 2.0 Cisco SRST 2.0	This command was implemented on the Cisco 1750 and Cisco 1751.
12.2(8)T	Cisco CME 2.0 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.
12.2(8)T1	Cisco CME 2.0 Cisco SRST 2.0	This command was implemented on the Cisco 2600XM and Cisco 2691.
12.2(11)T	Cisco CME 2.01 Cisco SRST 2.01	This command was implemented on the Cisco 1760.
12.3(4)T	Cisco CME 3.0 Cisco SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Examples

Cisco Unified CME

The following Cisco Unified CME sample output displays status and information for all ephone-dns:

```
Router# show ephone-dn
50/0/1 CH1 DOWN
EFXS 50/0/1 Slot is 50, Sub-unit is 0, Port is 1
 Type of VoicePort is EFXS
 Operation State is UP
 Administrative State is UP
 No Interface Down Failure
 Description is not set
 Noise Regeneration is enabled
```

```
 Non Linear Processing is enabled
 Non Linear Mute is disabled
 Non Linear Threshold is -21 dB
 Music On Hold Threshold is Set to -38 dBm
 In Gain is Set to 0 dB
 Out Attenuation is Set to 0 dB
 Echo Cancellation is enabled
 Echo Cancellation NLP mute is disabled
 Echo Cancellation NLP threshold is -21 dB
 Echo Cancel Coverage is set to 8 ms
 Playout-delay Mode is set to adaptive
 Playout-delay Nominal is set to 60 ms
 Playout-delay Maximum is set to 200 ms
 Playout-delay Minimum mode is set to default, value 40 ms
 Playout-delay Fax is set to 300 ms
 Connection Mode is normal
 Connection Number is not set
 Initial Time Out is set to 10 s
 Interdigit Time Out is set to 10 s
 Call Disconnect Time Out is set to 60 s
 Ringing Time Out is set to 180 s
 Wait Release Time Out is set to 30 s
 Companding Type is u-law
 Region Tone is set for US
 Station name None, Station number 91400
 Caller ID Info Follows:
 Standard BELLCORE
 Translation profile (Incoming):
 Translation profile (Outgoing):
 Digit Duration Timing is set to 100 ms
50/0/2 CH1 IDLE      CH2 IDLE
EFXS 50/0/2 Slot is 50, Sub-unit is 0, Port is 2
 Type of VoicePort is EFXS
 Operation State is DORMANT
 Administrative State is UP
 No Interface Down Failure
 Description is not set
 Noise Regeneration is enabled
 Non Linear Processing is enabled
 Non Linear Mute is disabled
 Non Linear Threshold is -21 dB
 Music On Hold Threshold is Set to -38 dBm
 In Gain is Set to 0 dB
 Out Attenuation is Set to 0 dB
 Echo Cancellation is enabled
 Echo Cancellation NLP mute is disabled
 Echo Cancellation NLP threshold is -21 dB
 Echo Cancel Coverage is set to 8 ms
 Playout-delay Mode is set to adaptive
 Playout-delay Nominal is set to 60 ms
 Playout-delay Maximum is set to 200 ms
 Playout-delay Minimum mode is set to default, value 40 ms
 Playout-delay Fax is set to 300 ms
 Connection Mode is normal
 Connection Number is not set
 Initial Time Out is set to 10 s
 Interdigit Time Out is set to 10 s
 Call Disconnect Time Out is set to 60 s
 Ringing Time Out is set to 180 s
 Wait Release Time Out is set to 30 s
 Companding Type is u-law
 Region Tone is set for US
 Station name None, Station number 91450
 Caller ID Info Follows:
```

```
 Standard BELLCORE
 Translation profile (Incoming):
 Translation profile (Outgoing):
 Digit Duration Timing is set to 100 ms
```

Cisco Unified SRST

The following SRST sample output displays status and information for all ephone-dns:

```
Router# show ephone-dn 7
50/0/7 INVALID
EFXS 50/0/7 Slot is 50, Sub-unit is 0, Port is 7
 Type of VoicePort is EFXS
 Operation State is UP
 Administrative State is UP
 No Interface Down Failure
 Description is not set
 Noise Regeneration is enabled
 Non Linear Processing is enabled
 Non Linear Mute is disabled
 Non Linear Threshold is -21 dB
 Music On Hold Threshold is Set to -38 dBm
 In Gain is Set to 0 dB
 Out Attenuation is Set to 0 dB
 Echo Cancellation is enabled
 Echo Cancellation NLP mute is disabled
 Echo Cancellation NLP threshold is -21 dB
 Echo Cancel Coverage is set to 8 ms
 Playout-delay Mode is set to default
 Playout-delay Nominal is set to 60 ms
 Playout-delay Maximum is set to 200 ms
 Playout-delay Minimum mode is set to default, value 4 ms
 Playout-delay Fax is set to 300 ms
 Connection Mode is normal
 Connection Number is not set
 Initial Time Out is set to 10 s
 Interdigit Time Out is set to 10 s
 Call Disconnect Time Out is set to 60 s
 Ringing Time Out is set to 8 s
 Wait Release Time Out is set to 30 s
 Companding Type is u-law
 Region Tone is set for US
 Station name None, Station number None
 Caller ID Info Follows:
 Standard BELLCORE
 Voice card specific Info Follows:
 Digit Duration Timing is set to 100 ms
```

Table 10: show ephone-dn Field Descriptions, on page 80 describes significant fields in the output from this command.

Table 10: show ephone-dn Field Descriptions

Field	Description
Administrative State	Administrative (configured) state of the voice port.

Field	Description
alert	The number of calls that were disconnected by the far-end device when the local IP phone was in the call alerting state (for example, because the far-end phone rang but was not answered and the far-end system decided to drop the call rather than let the phone ring for too long).
answered (incoming)	The number of incoming calls that were actually answered (the phone goes off hook when ringing).
answered (outgoing)	The number of outgoing call attempts that were answered by the far end.
busy	The number of outgoing call attempts that got a busy response.
Call Disconnect Time Out	Not applicable to the Cisco IP phone.
called, calling	Extension numbers of called and calling parties.
Caller ID Info Follows	Information about the caller ID.
Call Ref	A unique per-call identifier used by the SCCP protocol. The Call Ref values are assigned sequentially within the Cisco CME–SCCP interface, so this value also indicates the total number of SCCP calls since the router was last rebooted.
chan	Channel number of an ephone-dn.
CODEC	Codec type.
Companding Type	Not applicable to the Cisco IP phone.
connect	The number of calls that were disconnected by the far-end device when the local IP phone was in the call connected state.
Connection Mode	Not applicable to the Cisco IP phone.
Connection Number	Not applicable to the Cisco IP phone.
Description	Not applicable to the Cisco IP phone.
Digit Duration Timing	Not applicable to the Cisco IP phone.
DN STATE	Ephone-dn tag number and state of the phone line associated with an extension.
Echo Cancellation...	Not applicable to the Cisco IP phone.
Echo Cancel Coverage	Not applicable to the Cisco IP phone.
EFXS	Voice port type.
Far-end disconnect at...	See connect, alert, hold, and ring.
Final Jitter	The final voice packet receive jitter reported by the IP phone at the end of the call.

Field	Description
hold	The number of calls that were disconnected by the far-end device when the local IP phone was in the call hold state (for example, if the caller was left on hold for too long and got tired of waiting).
incoming	The number of incoming calls presented (the phone rings).
In Gain	Not applicable to the Cisco IP phone.
Initial Time Out	Amount of time the system waits for an initial input digit from the caller.
Interdigit Time Out	Amount of time the system waits for a subsequent input digit from the caller.
Last 64 far-end disconnect cause codes	See the Mappings of PSTN Cause Codes to SIP Event table for a list of public switch telephone network (PSTN) cause codes that can be sent as an ISDN cause information element (IE) and the corresponding Session Interface Protocol (SIP) event.
Latency	The final voice packet receive latency reported by the IP phone at the end of the call.
Lost	Number of lost packets.
Music On Hold Threshold	Not applicable to the Cisco IP phone.
No Interface Down Failure	State of the interface.
Noise Regeneration	Not applicable to the Cisco IP phone.
Non Linear...	Not applicable to the Cisco IP phone.
Operation State	Operational state of the voice port.
Out Attenuation	Not applicable to the Cisco IP phone.
outgoing	The number of outgoing call attempts.
Playout-delay Maximum	Not applicable to the Cisco IP phone.
Playout-delay...	Not applicable to the Cisco IP phone.
Port	Port number for the interface associated with the voice interface card.
Region Tone	Not applicable to the Cisco IP phone.
ring	The number of calls that were disconnected by the far-end device when the local IP phone was in the ringing state (for example, if the call was not answered and the caller hung up).
Ringing Time Out	Duration, in seconds, for which ringing is to continue if a call is not answered. Set with the **timeouts ringing** command.
Rx Pkts, bytes	Number of packets and bytes received during the current or last call.

Field	Description
Signal Level to phone, peak	For G.711 calls only, this parameter indicates the most recent voice signal level in the voice IP packets sent from the router to the IP phone. This parameter is valid only for VoIP or PSTN G.711 calls to the IP phones. This parameter is not valid for calls between local IP phones, or calls that use codecs other than G.711. The peak field indicates the peak signal level seen during the entire call.
Slot	Slot used in the voice interface card for this port.
Station name	Station name.
Station number	Station number.
Sub-unit	Subunit used in the voice interface card for this port.
Tx Pkts, bytes	Number of packets and bytes transmitted during the current call or last call.
Type of VoicePort	Voice port type.
VAD	Voice activity detection.
Voice card specific info	Information specific to the voice card.
VPM STATE	State indication for the VPM software component.
VTSP STATE	State indication for the VTSP software component.
Wait Release Time Out	Time that a voice port stays in the call-failure state while the router sends a busy tone, reorder tone, or out-of-service tone to the port.

The following table lists the PSTN cause codes that can be sent as an ISDN cause information element (IE) and the corresponding SIP event for each. These are the far-end disconnect cause codes listed in the output for the **show ephone-dn statistics** command.

Table 11: Mappings of PSTN Cause Codes to SIP Events

PSTN Cause Code	Description	SIP Event
1	Unallocated number	410 Gone
3	No route to destination	404 Not found
16	Normal call clearing	BYE
17	User busy	486 Busy here
18	No user responding	480 Temporarily unavailable
19	No answer from the user	
21	Call rejected	603 Decline

PSTN Cause Code	Description	SIP Event
22	Number changed	302 Moved temporarily
27	Destination out of order	404 Not found
28	Address incomplete	484 Address incomplete
29	Facility rejected	501 Not implemented
31	Normal unspecified	404 Not found
34	No circuit available	503 Service unavailable
38	Network out of order	
41	Temporary failure	
42	Switching equipment congestion	
44	Requested channel not available	
47	Resource unavailable	
55	Incoming class barred within CUG	603 Decline
57	Bearer capability not authorized	501 Not implemented
58	Bearer capability not presently available	
63	Service or option unavailable	503 Service unavailable
65	Bearer cap not implemented	501 Not implemented
79	Service or option not implemented	
87	User not member of CUG	603 Decline
88	Incompatible destination	400 Bad Request
95	Invalid message	
102	Recover on timer expiry	408 Request timeout
111	Protocol error	400 Bad request
127	Interworking unspecified	500 Internal server error
Any code other than those listed above	500 Internal server error	

Related Commands

Command	Description
show ephone-dn callback	Displays information about pending callbacks in a Cisco Unified CME or a Cisco Unified SRST environment.

Command	Description
show ephone-dn loopback	Displays information about loopback ephone-dns that have been created in a Cisco Unified CME or a Cisco Unified SRST environment.
show ephone-dn statistics	Displays display call statistics for a Cisco IP destination or for extensions (ephone-dns) in a Cisco Unified CME or a Cisco Unified SRST environment.
show ephone-dn summary	Displays brief information about Cisco IP phone destination numbers or for extensions (ephone-dns) in a Cisco Unified CME or a Cisco Unified SRST environment.

show ephone-dn callback

To display information about pending callbacks in a Cisco Unified Communications Manager Express (Cisco Unified CME) or a Cisco Unified Survivable Remote Site Telephony (Cisco Unified SRST) environment, use the **show ephone-dn callback** command in privileged EXEC mode.

show ephone-dn callback

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

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

Examples

The following sample output shows a callback placed by ephone-dn 1 against ephone-dn 3. Ephone-dn 3 has its channel 1 on hold and has just seized dial tone on its channel 2.

```
Router# show ephone-dn callback
DN 3 (95021) CallBack pending to DN 1 (95021) for ephone-1 age 7 seconds
State for DN 3 is CH1 HOLD      CH2 SIEZE
```

The following sample output shows a callback placed by ephone-dn 1 against ephone-dn 3. Ephone-dn 3 has a call in progress on channel 1.

```
Router# show ephone-dn callback
DN 3 (95021) CallBack pending to DN 1 (95021) for ephone-1 age 8 seconds
State for DN 3 is CH1 CONNECTED
```

Significant fields in the output from this command are described in Table 12: show ephone-dn callback Field Descriptions, on page 86.

Table 12: show ephone-dn callback Field Descriptions

Field	Description
DN 3 (95021) CallBack pending to DN 1 (95021)	Callback originator is the extension with the dn-tag 1 (in this example), and the callback has been placed on the extension with the dn-tag 3 and the number 95021.
age	Number of seconds since the callback was placed.
State for DN 3 is CH1... CH2...	Call states for channel 1 and channel 2, if any, of the extension that the callback is for.

Related Commands

Command	Description
show ephone-dn	Displays status and information for a Cisco IP phone destination number or for extensions (ephone-dns) in a Cisco Unified CME or a Cisco Unified SRST environment.

show ephone-dn loopback

To display information about loopback ephone-dns that have been created in a Cisco Unified Communications Manager Express (Cisco Unified CME) or a Cisco Unified Survivable Remote Site Telephony (Cisco Unified SRST) environment, use the **show ephone-dn loopback** command in privileged EXEC mode.

show ephone-dn loopback

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco CME 1.0 Cisco SRST 1.0	This command was introduced on the Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(2)XT	Cisco CME 2.0 Cisco SRST 2.0	This command was implemented on the Cisco 1750 and Cisco 1751.
12.2(8)T	Cisco CME 2.0 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.
12.2(8)T1	Cisco CME 2.0 Cisco SRST 2.0	This command was implemented on the Cisco 2600XM and Cisco 2691.
12.2(11)T	Cisco CME 2.01 Cisco SRST 2.01	This command was implemented on the Cisco 1760.

Examples

The following example displays information for a loopback using ephone-dn 21 and ephone-dn 22:

```
Router# show ephone-dn loopback
LOOPBACK DN status (min 21, max 22):
DN 21 51...  Loopback to DN 22 CH1 IDLE
CallingDn -1 CalledDn -1 Called   Calling   G711Ulaw64k
Strip NONE, Forward 2, prefix 10 retry 10 Media 0.0.0.0 0
callID 0 srcCallID 0 ssrc 0 vector 0
DN 22 11...  Loopback to DN 21 CH1 IDLE
CallingDn -1 CalledDn -1 Called   Calling   G711Ulaw64k
Strip NONE, Forward 2, prefix 50 retry 10 Media 0.0.0.0 0
callID 0 srcCallID 0 ssrc 0 vector 0
```

Significant fields in the output from this command are described in Table 13: show ephone-dn loopback Field Descriptions, on page 88, in alphabetical order.

Table 13: show ephone-dn loopback Field Descriptions

Field	Description
Called, Calling	Called number and calling number when there is a call present.

Field	Description
CalledDn, CallingDn	Ephone-dn tag numbers of the called and calling ephone-dn. Set to -1 if the call is not to or from an ephone-dn, or if there is no active call.
callID	Internal call reference. This usage is the same as in other Cisco IOS voice gateway commands.
DN	Ephone-dn tag (sequence number).
Forward	Number of digits in the original called number to forward to the other ephone-dn in the loopback-dn pair.
G711...	G711Ulaw64k indicates G.711 codec, mu-law, 64000-bit stream. G711alaw64k indicates G.711 codec, A-law, 64000-bit stream.
Loopback to ...	Indicates the opposite ephone-dn in the loopback pair and the status of that ephone-dn.
Media	IP destination address, if any, for any voice packets that are passing through the loopback DN.
min, max	Lowest and highest dn-tag numbers of ephone-dns that are configured as loopback-dns.
prefix	Digit string to add to the beginning of forwarded called numbers.
retry	Number of seconds to wait before retrying the loopback target when is it busy.
srcCallID	Internal call reference for the destination.
ssrc	Real-time transport protocol (RTP) synchronization source (SSRC) of the most recent RTP packet.
Strip	Number of leading digits to strip before forwarding to the other extension in the loopback-dn pair.
vector	The following values describe the media path for voice packets that pass through the loopback-dn: • 0—No media path or not a loopback-dn path (inactive). • 1—Normal path. Loopback-dn has identified the final media destination as a local IP phone. The media IP address field shows a valid, non-zero value. • 2—Hairpin. Media packets are routed back through paired loopback-dns. The final destination is not known. For example, this can be a VoIP-to-VoIP call path by a loopback-dn. • 3—Hairpin. The final destination is an ephone-dn in a special mode such as paging. • 4—Loopback-dn chain has been detected, in which two loopback-dn pairs have been connected together. • 5—Loopback-dn chain has been detected in which more than two loopback-dn pairs are connected in series.

Related Commands

Command	Description
loopback-dn	Creates a virtual loopback voice port (loopback-dn) to establish a demarcation point for VoIP voice calls and supplementary services.
show ephone-dn	Displays status and information for a Cisco IP phone destination number or for extensions (ephone-dns) in a Cisco Unified CME or a Cisco Unified SRST environment.

show ephone-dn statistics

To display call statistics for a Cisco IP destination or for extensions (ephone-dns) in a Cisco Unified Communications Manager Express (Cisco Unified CME) or a Cisco Unified Survivable Remote Site Telephony (Cisco Unified SRST) environment, use the **show ephone-dn** command in privileged EXEC mode.

show ephone-dn [*dn-tag*] **statistics**

Syntax Description

dn-tag	(Optional) Unique sequence number that is used during configuration to identify a particular extension (ephone-dn).
statistics	Displays voice quality statistics on calls for a specified extension or for all extensions.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ1	Cisco CME 3.0 Cisco SRST 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0 Cisco SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Examples

The following sample output displays statistics for all extensions (ephone-dns) in a Cisco Unified CME system. There are two ephone-dns (DN1 and DN3) in this example.

```
Router# show ephone-dn statistics
Total Calls 103
Stats may appear to be inconsistent for conference or shared line cases
DN 1 chan 1 incoming 36 answered 21 outgoing 60 answered 30 busy 6
Far-end disconnect at:connect 29 alert 18 hold 7 ring 15
Last 64 far-end disconnect cause codes
17 17 17 17 17 17 16 16 16 16 16 16 16 16 16 16
16 16 16 16 65 16 65 65 65 65 16 65 65 65 16 16
16 16 16 16 16 16 16 16 16 16 16 16 16 65 47 65
47 47 16 16 16 16 16 16 16 16 16 16 16 16 16 16
local phone on-hook
DN 1 chan 1 (95011) voice quality statistics for last call
Call Ref 103 called 91500 calling 95011
Total Tx Pkts 0 bytes 0 Rx Pkts 0 bytes 0 Lost 0
Final Jitter 30 Latency 0 Lost 0
Signal Level to phone 0 (-78 dB) peak 0 (-78 dB)
Packets counted by router 0
DN 1 chan 2 incoming 0 answered 0 outgoing 1 answered 0 busy 0
Far-end disconnect at:connect 0 alert 0 hold 0 ring 0
Last 64 far-end disconnect cause codes
0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
local phone on-hook
DN 1 chan 2 (95011) voice quality statistics for last call
Call Ref 86 called  calling
```

```
Total Tx Pkts 0 bytes 0 Rx Pkts 0 bytes 0 Lost 0
Final Jitter 0 Latency 0 Lost 0
Signal Level to phone 0 (-78 dB) peak 0 (-78 dB)
Packets counted by router 0
DN 3 chan 1 incoming 0 answered 0 outgoing 1 answered 1 busy 0
Far-end disconnect at:connect 0 alert 0 hold 0 ring 0
Last 64 far-end disconnect cause codes
0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
DN 3 chan 1 (95021) voice quality statistics for current call
Call Ref 102 called 94011 calling 95021
Current Tx Pkts 241 bytes 3133 Rx Pkts 3304 bytes 515023 Lost 0
Jitter 30 Latency 0
Worst Jitter 30 Worst Latency 0
Signal Level to phone 201 (-39 dB) peak 5628 (-12 dB)
Packets counted by router 3305
```

The following sample output displays voice quality statistics for the ephone-dn with dn-tag 2:

```
Router# show ephone-dn 2 statistics
DN 2 chan 1 incoming 0 answered 0 outgoing 2 answered 0 busy 0
Far-end disconnect at: connect 0 alert 0 hold 0 ring 0
Last 64 far-end disconnect cause codes
28 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
local phone on-hook
DN 2 chan 1 (91450) voice quality statistics for last call
Call Ref 2 called  calling
Total Tx Pkts 0 bytes 0 Rx Pkts 0 bytes 0 Lost 0
Final Jitter 0 Latency 0 Lost 0
Signal Level to phone 0 (-78 dB) peak 0 (-78 dB)
Packets counted by router 0
```

The **show ephone-dn** command describes significant fields in the output from this command.

Related Commands

Command	Description
show ephone-dn	Displays status and information for a Cisco IP phone destination number or for extensions (ephone-dns) in a Cisco Unified CME or a Cisco Unified SRST environment.

show ephone-dn summary

To display brief information about Cisco IP phone destination numbers or for extensions (ephone-dns) in a Cisco Unified Communications Manager Express (Cisco Unified CME) or a Cisco Unified Survivable Remote Site Telephony (Cisco Unified SRST) environment, use the **show ephone-dn summary** command in privileged EXEC mode.

show ephone-dn summary

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco CME 1.0 Cisco SRST 1.0	This command was introduced on the Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(2)XT	Cisco CME 2.0 Cisco SRST 2.0	This command was implemented on the Cisco 1750 and Cisco 1751.
12.2(8)T	Cisco CME 2.0 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.
12.2(8)T1	Cisco CME 2.0 Cisco SRST 2.0	This command was implemented on the Cisco 2600XM and Cisco 2691.
12.2(11)T	Cisco CME 2.01 Cisco SRST 2.01	This command was implemented on the Cisco 1760.

Examples

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

```
Router# show ephone-dn summary
PORT     DN STATE    CODEC    VAD VTSP STATE             VPM STATE
======== ========== ======== === ===================== =========
50/0/1   DOWN        -          -  -                    EFXS_ONHOOK

50/0/2   DOWN        -          -  -                    EFXS_ONHOOK

50/0/3   DOWN        -          -  -                    EFXS_ONHOOK

50/0/4   INVALID     -          -  -                    EFXS_INIT

50/0/5   INVALID     -          -  -                    EFXS_INIT

50/0/6   INVALID     -          -  -                    EFXS_INIT
```

Table 14: show ephone-dn summary Field Descriptions, on page 94 describes significant fields in the output from this command.

Table 14: show ephone-dn summary Field Descriptions

Field	Description
CODEC	Type of codec.
DN STATE	Status of the ephone-dn.
EFXS	Voice port type.
PORT	Port number (virtual) for this interface. The number that follows the last slash in the port number is the ephone-dn tag. For example, if the port number is 50/0/1, the dn-tag is 1.
VAD	Voice activity detection status.
VPM STATE	State indication for the voice port module (VPM) software component.
VTSP STATE	State indication for the voice telephony service provider (VTSP) software component.

Related Commands

Command	Description
show ephone-dn	Displays status and information for a Cisco IP phone destination number or for extensions (ephone-dns) in a Cisco Unified CME or a Cisco Unified SRST environment.

show sip-ua status registrar

To display all the SIP endpoints that are currently registered with the contact address, use the **show sip-ua status registrar** command in privileged EXEC mode.

show sip-ua status registrar

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

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.
15.0(1)XA	Cisco SIP SRST 8.0	This command was updated to display the signaling transport protocol.

Examples

The following is a sample output from this command:

```
Router# show sip-ua status registrar
Line            destination       expires(sec)   contact
transport       call-id
                peer
============    ===============   ============   ===============
3029991         10.2.30.108       388            10.2.30.108
TLS             00120014-4ae40064-f1a3e9fe-8d301072@10.2.30.1
              40004
3029993         10.2.30.103       382            10.2.30.103
TCP             001bd433-1c840052-655cd596-4e992eed@10.2.30.1
                40011
3029982         10.2.30.106        406           10.2.30.106
UDP             001d452c-dbba0056-0481d321-1f3f848d@10.2.30.1
                40001
```

Table 15: show sip-ua status registrar Field Descriptions, on page 95 describes the significant fields shown in this output.

Table 15: show sip-ua status registrar Field Descriptions

Field	Description
call-id	A unique ID assigned for each call.
contact	The contact IP address provided by the Cisco SIP IP phone.
destination	The destination IP address.
expires (sec)	The amount of time, in seconds, until registration expires.
Line	The phone number that maintains registration of SIP devices.

Field	Description
peer	When an SIP IP phone registers, an associated VoIP dial peer is automatically generated. This dial peer contains general information on how to contact the phone. The information includes the directory number or numbers associated with the phone and the IP address and protocol of the phone.

Related Commands

Command	Description
registrar server	Enables SIP registrar functionality.

show sip-ua connections tcp tls detail

To display the status, port details, and negotiated ciphers for SIP OAuth.

Note The Conn-Id that is suffixed with * is the client connections using SIP OAuth port.

show sip-ua connections tcp tls detail

This command has no arguments or keywords.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
Cisco IOS XE Cupertino 17.8.1a	The command output is updated to display the status, port details, and negotiated ciphers for SIP OAuth.
Cisco IOS XE 16.10.1	The command output for **show sip-ua connections tcp tls detail** was updated to display the Cipher and the Curve-Size.

```
Router#show sip-ua connections tcp tls detail
Total active connections      : 4
No. of send failures          : 0
No. of remote closures        : 8
No. of conn. failures         : 0
No. of inactive conn. ageouts : 0
TLS client handshake failures : 0
TLS server handshake failures : 0

---------Printing Detailed Connection Report---------
Note:
 ** Tuples with no matching socket entry
    - Do 'clear sip <tcp[tls]/udp> conn t ipv4:<addr>:<port>'
      to overcome this error condition
 ++ Tuples with mismatched address/port entry
    - Do 'clear sip <tcp[tls]/udp> conn t ipv4:<addr>:<port> id <connid>'
      to overcome this error condition
 * Connections with SIP OAuth ports

Remote-Agent:10.5.10.200, Connections-Count:0

Remote-Agent:10.5.10.201, Connections-Count:0

Remote-Agent:10.5.10.202, Connections-Count:0

Remote-Agent:10.5.10.212, Connections-Count:1
  Remote-Port Conn-Id Conn-State  WriteQ-Size Local-Address TLS-Version Cipher
             Curve
  =========== ======= =========== =========== ============= ===========
==================================== =====
        52248      27 Established           0             -      TLSv1.2
ECDHE-RSA-AES256-GCM-SHA384 P-256

Remote-Agent:10.5.10.213, Connections-Count:1
```

```
  Remote-Port Conn-Id Conn-State  WriteQ-Size Local-Address TLS-Version Cipher
             Curve
  =========== ======= =========== =========== ============= ===========
============================== =====
        50901     28* Established           0             -      TLSv1.2
ECDHE-RSA-AES256-GCM-SHA384 P-256

Remote-Agent:10.5.10.209, Connections-Count:1
  Remote-Port Conn-Id Conn-State  WriteQ-Size Local-Address TLS-Version Cipher
             Curve
  =========== ======= =========== =========== ============= ===========
============================== =====
        51402     29* Established           0             -      TLSv1.2
ECDHE-RSA-AES256-GCM-SHA384 P-256

Remote-Agent:10.5.10.204, Connections-Count:1
  Remote-Port Conn-Id Conn-State  WriteQ-Size Local-Address TLS-Version Cipher
             Curve
  =========== ======= =========== =========== ============= ===========
============================== =====
        50757     30* Established           0             -      TLSv1.2
ECDHE-RSA-AES256-GCM-SHA384 P-256

Remote-Agent:10.5.10.218, Connections-Count:0


-------------- SIP Transport Layer Listen Sockets ---------------
  Conn-Id               Local-Address
 ===========    =============================
   0            [0.0.0.0]:5061:
   2            [0.0.0.0]:5090:
```

show voice emergency

To display the IP address, subnet mask, and ELIN for each emergency response location, use the **show voice emergency** command in user EXEC or privileged EXEC mode.

show voice emergency

Syntax Description

This command has no arguments or keywords.

Command Default

No default behavior or values

Command Modes

User EXEC (>)
Privileged EXEC (#)

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 displays the IP address, subnet mask, and ELIN for each emergency response location.

Examples

The following example shows a sample output which includes IP mask and ELIN information for each ERL:

```
EEMERGENCY RESPONSE LOCATIONS
ERL                  | ELIN 1      | ELIN2        | SUBNET 1         | SUBNET 2
1                    | 6045550101 |              | 10.0.0.0         | 255.0.0.0
2                    | 6045550102 | 6045550106 | 192.168.0.0      | 255.255.0.0
3                    |             | 6045550107 | 172.16.0.0       | 255.255.0.0
4                    | 6045550103 |              | 192.168.0.0      | 255.255.0.0
5                    | 6045550105 |              | 209.165.200.224 | 255.0.0.0
6 6045550198        |             | 6045550109 | 209.165.201.0    | 255.255.255.224
```

Related Commands

Command	Description
voice emergency response location	Creates a tag for identifying an ERL for E911 services.

show voice emergency addresses

To display address information for each emergency response location, use the **show emergency addresses** command in user EXEC or privileged EXEC mode.

show voice emergency addresses

Syntax Description

This command has no arguments or keywords.

Command Default

No default behavior or values

Command Modes

User EXEC (>)
Privileged EXEC (#)

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 displays the physical address of each emergency response location.

Examples

The following example shows a sample output which includes physical address information for the ERL:

```
Router# show voice emergency addresses
3850 Zanker Rd, San Jose,604,5550101
225 W Tasman Dr, San Jose,604,5550102
275 W Tasman Dr, San Jose,604,5550103
518 Bellew Dr,Milpitas,604,5550104
400 Tasman Dr,San Jose,604,5550105
    3675 Cisco Way,San Jose,604,5550106
```

Related Commands

Command	Description
address	Specifies a comma separated text entry (up to 250 characters) of an ERL's civic address.
show voice emergency all	Displays all emergency response location information.
voice emergency response location	Creates a tag for identifying an ERL for E911 services.

show voice emergency all

To display all emergency response location information, use the **show voice emergency all** command in user EXEC or privileged EXEC mode.

show voice emergency all

Syntax Description

This command has no arguments or keywords.

Command Default

No default behavior or values

Command Modes

User EXEC (>)
Privileged EXEC (#)

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 displays all information configured for each emergency response location.

Examples

The following example shows a sample output, displaying all ERL-related information for ERL 1 and 3.

```
VOICE EMERGENCY RESPONSE SETTINGS
   Callback Number: 6045550103
   Emergency Line ID Number: 6045550155
   Expiry: 2 minutes
   Logging Enabled
EMERGENCY RESPONSE LOCATION 1
   Name: Cisco Systems 1
   Address: 3850 Zanker Rd, San Jose,elin.1.3,elin.4.10
   IP Address 1: 209.165.200.226 IP mask 1: 255.255.255.254
   IP Address 2: 209.165.202.129 IP mask 2: 255.255.0.0
   Emergency Line ID 1: 6045550180
   Emergency Line ID 2:
   Last Caller:  6045550188 [Jan 30 2007 16:05.52 PM]
   Next ELIN For Emergency Call: 6045550166
EMERGENCY RESPONSE LOCATION 3
   Name: Cisco Systems 3
   Address: 225 W Tasman Dr, San Jose,elin.1.3,elin.4.10
   IP Address 1: 209.165.202.133 IP mask 1: 255.255.0.0
   IP Address 2: 209.165.202.130 IP mask 2: 255.0.0.0
   Emergency Line ID 1:
   Emergency Line ID 2: 6045550150
   Last Caller:
   Next ELIN For Emergency Call: 6045550151
```

Related Commands

Command	Description
address	Specifies a comma separated text entry (up to 250 characters) of an ERL's civic address.
elin	Specifies a PSTN number that will replace the caller's extension.
name	Specifies a string (up to 32-characters) used internally to identify or describe the emergency response location.
subnet	Defines which IP phones are part of this ERL.
voice emergency response location	Creates a tag for identifying an ERL for the E911 services.

show voice emergency callers

To display a list of 911 calls made over the last three hours, use the **show emergency callers** command in privileged EXEC mode.

show voice emergency callers

Syntax Description

This command has no arguments or keywords.

Command Default

No list of 911 calls is displayed.

Command Modes

Privileged EXEC (#)

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.
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 displays a list of all 911 calls made in the past three hours. The list shows the originating number, the ELIN used, and the time the call was placed.

Examples

The following example shows a sample output, which includes the originating number, the ELIN used, and the time the call was placed:

```
router# show voice emergency callers
EMERGENCY CALLS CALL BACK TABLE
ELIN                      | CALLER                       | TIME
6045550181                | 8155550151                  | Oct 12 2006 04:05:21
6045550182                | 8155550152                  | Oct 12 2006 04:05:21
```

Related Commands

Command	Description
voice emergency response location	Creates a tag for identifying an ERL for the enhanced 911 service.

show voice emergency zone

To display each emergency response zone's list of locations in the order of priority, use the **show voice emergency zone** command in user EXEC or privileged EXEC mode.

show voice emergency zone

Syntax Description

This command has no arguments or keywords.

Command Default

No default behavior or values

Command Modes

User EXEC (>)
Privileged EXEC (#)

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 displays a list of the locations, in priority order, of all configured emergency response zones.

Examples

The following example shows a sample output which displays the ERL locations for emergency response zones 90 and 100.

```
EMERGENCY RESPONSE ZONES
 zone 90
    location 4
    location 5
    location 6
    location 7
    location 2147483647
 zone 100
    location 1 priority 1
    location 2 priority 2
    location 3 priority 3
```

Related Commands

Command	Description
location	Identifies locations within an emergency response zone.
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.

show voice moh-group

To display information about voice moh-groups, use the **show voice moh-group** command in in privileged EXEC mode.

show voice moh-group

Syntax Description

This command has no arguments or keywords

Command Modes

Privileged EXEC (#)

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.

Examples

The following sample output shows general information about voice moh-groups in Cisco Unifiede CME or Cisco Unified SRST.

```
Router# show voice moh-group
voice moh-group 1
 description this moh group is for sales
 moh hello.au
 multicast moh 239.1.1.1 port 16386 route 239.1.1.3 239.1.1.3
 extension-range 1000 to 1999
 extension-range 2000 to 2999
 extension-range 3000 to 3999
extension-range 20000 to 22000
 extension-range A1000 to A1999
voice moh-group 2
 description (not configured)
 moh minuet.au
 multicast moh 239.23.4.10 port 2000
 extension-range 7000 to 7999
 extension-range 8000 to 8999
voice moh-group 3
 description This is for marketing
 moh happy.au
 multicast moh 239.15.10.1 port 3000
 extension-range 9000 to 9999
voice moh-group 4
 description (not configured)
moh sun.au
 multicast moh 239.16.12.1 port 4000
 extension-range 10000 to 19999
voice moh-group 5
 description (not configured)
 moh flower.wav
 multicast moh 239.12.1.2 port 5000
 extension-range ABCD to DECF
 extension-range 0012 to 0024
```

```
 extension-range 0934 to 0964
=== Total of 5 voice moh-groups ===
e
```

Examples

Command	Description
showcall-manager-fallback all	**Displays the detailed configuration of all Cisco IP phones, directory numbers, voice ports, and dial peers in your network during Cisco Unified Communications Manager fallback.**
show ephone summary	Displays the information about the MOH files in use
show voice moh-group statistics	Displays the MOH subsystem statistics information

show voice moh-group statistics

To display the MOH subsystem statistics information, use the **show voice moh-group** command in privileged EXEC mode.

show voice moh-group statistics

Syntax Description

This command has no arguments or keywords

Command Modes

Privileged EXEC (#)

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.

Examples

In the following example, the MOH Group Streaming Interval Timing Statistics shows the media packet counts during streaming intervals.

Each packet counter is of 32 bit size and holds a count limit of 4294967296 intervals. This means that with 20 milliseconds packet interval (for G.711), the counters restart from 0 any time after 2.72 years (2 years 8 months). You must use the e clear voice moh-group statistics once in every two years to reset the packet counters.

MOH Group Packet Transmission Timing Statistics shows the maximum and minimum amount of time (in microseconds) taken by the MOH groups to send out media packets.

The MOH Group Loopback Interval Timing Statistics is available when loopback interface is configured as part of the multicast MOH routes in Cisco Unified SRST . These counts are loopback packet counts within certain streaming timing intervals.

```
router# show voice moh-group statistics
MOH Group Streaming Interval Timing Statistics:
Grp#  ~19 msec     20~39      40~59      60~99     100~199   200+ msec
==== ========== ========== ========== ========== ========== ==========
 0:       25835   17559966      45148          0          0          1
 1:       19766   17572103      39079          0          0          1
 2:       32374   17546886      51687          0          0          1
 3:       27976   17555681      47289          0          0          1
 4:       34346   17542940      53659          0          0          1
 5:       14971   17581689      34284          0          0          1
MOH Group Packet Transmission Timing Statistics:
Grp#  max(usec)  min(usec)
==== ========== ==========
 0:          97          7.
 1:          95          7.
 2:          97          7.
 3:          96          7.
 4:          94          7.
 5:          67          7.
```

```
 MOH Group Loopback Interval Timing Statistics:
 loopback event array: svc_index=1542, free_index=1549, max_q_depth=31
 Grp#  ~19 msec     20~39      40~59      60~99     100~199   200+ msec
 ==== ========== ========== ========== ========== ========== ==========
 0:     8918821    8721527      10023          0          1          1
 1:     9007373    8635813       7184          0          1          1
 2:     8864760    8772851      12758          0          1          1
 3:     8924447    8715457      10464          0          1          1
 4:     8858393    8778957      13017          0          1          1
 5:     9005511    8639936       4919          0          1          1
Statistics collect time: 4 days 2 hours 5 minutes 39 seconds.
```

Related Commands

Command	Description
show ephone-dn	Displays MOH group information for a phone directory number.
show ephone summary	Displays the information about the MOH files in use
show voice moh-group	Displays the MOH groups configured

show voice register all

To display all Cisco Unified Session Initiation Protocol (SIP) Survivable Remote Site Telephony (SRST) or Cisco Unified Communications Manager Express (Cisco Unified CME) configurations and register information, use the **show voice register all** command in privileged EXEC mode.

show voice register all

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

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.
15.0(1)XA	Cisco SIP SRST 8.0	This command was updated to display the signaling transport protocol.
15.1(2)T	Cisco Unified CME 8.1 Cisco Unified SIP SRST 8.1	This command was modified. The output display was modified.

Examples

Cisco Unified SIP SRST

The following is an example of show voice register all command:

```
Router# show voice register all
VOICE REGISTER GLOBAL
=====================
CONFIG [Version=8.1]
========================
  Version 8.1
  Mode is srst
  Max-pool is 10
  Max-dn is 10
  Outbound-proxy is enabled and will use global configured value
  Security Policy: DEVICE-DEFAULT
  timeout interdigit 10
  network-locale[0] US    (This is the default network locale for this box)
  network-locale[1] US
  network-locale[2] US
  network-locale[3] US
  network-locale[4] US
  user-locale[0] US    (This is the default user locale for this box)
  user-locale[1] US
```

```
  user-locale[2] US
  user-locale[3] US
  user-locale[4] US   Active registrations  : 0
  Total SIP phones registered: 0
  Total Registration Statistics
    Registration requests  : 0
    Registration success   : 0
    Registration failed    : 0
    unRegister requests    : 0
    unRegister success     : 0
    unRegister failed      : 0
    Attempts to register
           after last unregister : 0
    Last register request time   :
    Last unregister request time :
    Register success time        :
    Unregister success time      :
VOICE REGISTER DN
=================
Dn Tag 1
Config:
  Number is 45111
  Preference is 0
  Huntstop is disabled
  Pool 1    has this DN configured for line 1
Dn Tag 2
Config:
  Number is 45112
  Preference is 0
  Huntstop is disabled
  Pool 2    has this DN configured for line 1
Dn Tag 3
Config:
  Number is 45113
  Preference is 0
  Huntstop is disabled
  Pool 3    has this DN configured for line 1, 2
Dn Tag 4
Config:
Dn Tag 7
Config:
  Number is 451110
  Preference is 0
  Huntstop is disabled
  Pool 1    has this DN configured for line 4
Dn Tag 8
Config:
  Pool 1    has this DN configured for line 3
VOICE REGISTER POOL
===================
 Pool Tag 1
Config:
  Mac address is 001B.535C.D410
  Number list 1 : DN 1
  Number list 3 : DN 8
  Number list 4 : DN 7
  Proxy Ip address is 0.0.0.0
  DTMF Relay is disabled
  kpml signal is disabled
  Lpcor Type is none
  Reason for unregistered state:
         No registration request since last reboot/unregister
Dialpeers created:
Statistics:
```

```
  Active registrations  : 0
  Total SIP phones registered: 0
  Total Registration Statistics
    Registration requests  : 0
    Registration success   : 0
    Registration failed    : 0
    unRegister requests    : 0
    unRegister success     : 0
    unRegister failed      : 0
    Attempts to register
           after last unregister : 0
    Last register request time   :
    Last unregister request time :
    Register success time        :
    Unregister success time      :
 Pool Tag 2
Config:
  Mac address is 0015.C68E.6D13
  Number list 1 : DN 2
  Proxy Ip address is 0.0.0.0
  DTMF Relay is disabled
  kpml signal is disabled
  Lpcor Type is none
  Reason for unregistered state:
         No registration request since last reboot/unregister
Dialpeers created:
Statistics:
  Active registrations  : 0
  Total SIP phones registered: 0
  Total Registration Statistics
    Registration requests  : 0
    Registration success   : 0
    Registration failed    : 0
    unRegister requests    : 0
    unRegister success     : 0
    unRegister failed      : 0
    Attempts to register
           after last unregister : 0
    Last register request time   :
    Last unregister request time :
    Register success time        :
    Unregister success time      :
 Pool Tag 3
Config:
  Mac address is 0021.5553.8998
  Number list 1 : DN 3
  Number list 2 : DN 3
  Proxy Ip address is 0.0.0.0
  DTMF Relay is disabled
  kpml signal is enabled
  Lpcor Type is none
  Reason for unregistered state:
         No registration request since last reboot/unregister
Dialpeers created:
Statistics:
  Active registrations  : 0
  Total SIP phones registered: 0
  Total Registration Statistics
    Registration requests  : 0
    Registration success   : 0
    Registration failed    : 0
    unRegister requests    : 0
    unRegister success     : 0
    unRegister failed      : 0
```

```
    Attempts to register
           after last unregister : 0
    Last register request time   :
    Last unregister request time :
    Register success time        :
    Unregister success time      :
```

Cisco Unified CME

The following is an example of show voice register all command :

```
Router# show voice register all
1) show voice register all
VOICE REGISTER GLOBAL
=====================
CONFIG [Version=8.1]
========================
  Version 8.1
  Mode is cme
  Max-pool is 10
  Max-dn is 10
  Outbound-proxy is enabled and will use global configured value
  Security Policy: DEVICE-DEFAULT
  Source-address is 8.3.3.5 port 5060
  Time-format is 12
  Date-format is M/D/Y
  Time-zone is 5
  Hold-alert is disabled
  Mwi stutter is disabled
  Mwi registration for full E.164 is disabled
  Forwarding local is enabled
  Privacy is enabled
  Privacy-on-hold is disabled
  Dst auto adjust is enabled
    start at Apr week 1 day Sun time 02:00
    stop  at Oct week 8 day Sun time 02:00
  Max redirect number is 5
  IP QoS DSCP:
    ef (the MS 6 bits, 46, in ToS, 0xB8) for media
    cs3 (the MS 6 bits, 24, in ToS, 0x60) for signal
    af41 (the MS 6 bits, 34, in ToS, 0x88) for video
    default (the MS 6 bits, 0, in ToS, 0x0) for service
  Telnet Level: 0
  Tftp path is flash:
  Generate text file is disabled
  Tftp files are created, current syncinfo 0001140473454008
  OS79XX.TXT is not created
  timeout interdigit 10
  network-locale[0] US    (This is the default network locale for this box)
  network-locale[1] US
  network-locale[2] US
  network-locale[3] US
  network-locale[4] US
  user-locale[0] US    (This is the default user locale for this box)
  user-locale[1] US
  user-locale[2] US
  user-locale[3] US
  user-locale[4] US   Active registrations  : 0
  Total SIP phones registered: 0
  Total Registration Statistics
    Registration requests  : 0
```

```
    Registration success   : 0
    Registration failed    : 0
    unRegister requests    : 0
    unRegister success     : 0
    unRegister failed      : 0
    Attempts to register
           after last unregister : 0
    Last register request time   :
    Last unregister request time :
    Register success time        :
    Unregister success time      :
VOICE REGISTER DN
=================
Dn Tag 1
Config:
  Number is 45111
  Preference is 0
  Huntstop is disabled
  Auto answer is disabled
  Pool 1    has this DN configured for line 1
Dn Tag 2
Config:
  Number is 45112
  Preference is 0
  Huntstop is disabled
  Auto answer is disabled
  call-forward b2bua noan 999 timeout 8
  after-hour exempt
  Pool 2    has this DN configured for line 1
  Pool 7    has this DN configured for line 1
Dn Tag 3
Config:
  Number is 45113
  Preference is 0
  Huntstop is disabled
  Auto answer is disabled
  call-forward b2bua all 87687
  Pool 3    has this DN configured for line 1, 2
Dn Tag 4
Config:
  Auto answer is disabled
Dn Tag 7
Config:
  Number is 451110
  Preference is 0
  Huntstop is disabled
  Auto answer is disabled
  after-hour exempt
  Pool 1    has this DN configured for line 4
Dn Tag 8
Config:
  Auto answer is disabled
  call-forward b2bua all 678
  after-hour exempt
  Pool 1    has this DN configured for line 3
VOICE REGISTER TEMPLATE
=======================
Temp Tag 1
Config:
  Attended Transfer is enabled
  Blind Transfer is enabled
  Semi-attended Transfer is enabled
  Conference is enabled
  Caller-ID block is disabled
```

```
  DnD control is enabled
  Anonymous call block is disabled
  Dialplan Tag is 1
  softkey connected  Confrn
  Lpcor type none
  Pool 4 has this template configured
VOICE REGISTER DIALPLAN
=======================
Dialplan Tag 1
Config:
  Type is 7905-7912
  Template 1 has this dialplan configured
  Pool 4 has this dialplan configured
VOICE REGISTER POOL
===================
 Pool Tag 1
Config:
  Mac address is 001B.535C.D410
  Type is 7960
  Number list 1 : DN 1
  Number list 3 : DN 8
  Number list 4 : DN 7
  Proxy Ip address is 0.0.0.0
  DTMF Relay is disabled
  Call Waiting is enabled
  DnD is disabled
  Busy trigger per button value is 0
  call-forward phone all is 4566
  call-forward b2bua all 4555
  keep-conference is enabled
  Lpcor Type is none
  Transport type is udp
  service-control mechanism is not supported
  Privacy feature is not configured.
  Privacy button is disabled
  Reason for unregistered state:
         No registration request since last reboot/unregister
Dialpeers created:
Statistics:
  Active registrations  : 0
  Total SIP phones registered: 0
  Total Registration Statistics
    Registration requests  : 0
    Registration success   : 0
    Registration failed    : 0
    unRegister requests    : 0
    unRegister success     : 0
    unRegister failed      : 0
    Attempts to register
           after last unregister : 0
    Last register request time   :
    Last unregister request time :
    Register success time        :
    Unregister success time      :
 Pool Tag 2
Config:
  Mac address is 0015.C68E.6D13
  Type is 7960
  Number list 1 : DN 2
  Proxy Ip address is 0.0.0.0
  DTMF Relay is disabled
  Call Waiting is enabled
  DnD is disabled
  Busy trigger per button value is 0
```

```
  call-forward phone noan is 9886, timeout 98
  keep-conference is enabled
  username pool2 password lab
  Lpcor Type is none
  Transport type is udp
  service-control mechanism is not supported
  Privacy feature is not configured.
  Privacy button is disabled
  Reason for unregistered state:
         No registration request since last reboot/unregister
Dialpeers created:
Statistics:
  Active registrations  : 0

  Total SIP phones registered: 0
  Total Registration Statistics
    Registration requests  : 0
    Registration success   : 0
    Registration failed    : 0
    unRegister requests    : 0
    unRegister success     : 0
    unRegister failed      : 0
    Attempts to register
           after last unregister : 0
    Last register request time   :
    Last unregister request time :
    Register success time        :
    Unregister success time      :
 Pool Tag 3
Config:
  Mac address is 0021.5553.8998
  Type is 7975
  Number list 1 : DN 3
  Number list 2 : DN 3
  Proxy Ip address is 0.0.0.0
  DTMF Relay is disabled
  Call Waiting is enabled
  DnD is enabled
  Busy trigger per button value is 0
  call-forward phone all is 45112
  call-forward b2bua all 45111
  after-hour exempt
  keep-conference is enabled
  kpml signal is enabled
  Lpcor Type is none
  Transport type is udp
  service-control mechanism is not supported
  Privacy feature is not configured.
  Privacy button is disabled
  Reason for unregistered state:
         No registration request since last reboot/unregister
Dialpeers created:
Statistics:
  Active registrations  : 0
  Total SIP phones registered: 0
  Total Registration Statistics
    Registration requests  : 0
    Registration success   : 0
    Registration failed    : 0
    unRegister requests    : 0
    unRegister success     : 0
    unRegister failed      : 0
    Attempts to register
           after last unregister : 0
```

```
    Last register request time   :
    Last unregister request time :
    Register success time        :
    Unregister success time      :
 Pool Tag 4
Config:
  Mac address is 8989.9867.8769
  Proxy Ip address is 0.0.0.0
  DTMF Relay is disabled
  Call Waiting is enabled
  DnD is disabled
  Busy trigger per button value is 0
  keep-conference is enabled
  template is 1
  Lpcor Type is none
  Transport type is udp
  service-control mechanism is not supported
  Privacy feature is not configured.
  Privacy button is disabled
  Reason for unregistered state:
         No registration request since last reboot/unregister
Dialpeers created:
Statistics:
  Active registrations  : 0
  Total SIP phones registered: 0
  Total Registration Statistics
    Registration requests  : 0
    Registration success   : 0
    Registration failed    : 0
    unRegister requests    : 0
    unRegister success     : 0
    unRegister failed      : 0
    Attempts to register
           after last unregister : 0
    Last register request time   :
    Last unregister request time :
    Register success time        :
    Unregister success time      :
 Pool Tag 7
Config:
  Mac address is 0018.BAC8.D2B1
  Number list 1 : DN 2
  Proxy Ip address is 0.0.0.0
  DTMF Relay is disabled
  Call Waiting is enabled
  DnD is disabled
  Busy trigger per button value is 0
  keep-conference is enabled
  Lpcor Type is none
  Transport type is udp
  service-control mechanism is not supported
  Privacy feature is not configured.
  Privacy button is disabled
  Reason for unregistered state:
         No registration request since last reboot/unregister
Dialpeers created:
Statistics:
  Active registrations  : 0
  Total SIP phones registered: 0
  Total Registration Statistics
    Registration requests  : 0
    Registration success   : 0
    Registration failed    : 0
    unRegister requests    : 0
```

```
unRegister success     : 0
unRegister failed      : 0
Attempts to register
       after last unregister : 0
Last register request time   :
Last unregister request time :
Register success time        :
Unregister success time      :
```

Table 16: show voice register all Field Descriptions, on page 117 describes the significant fields shown in this output.

Table 16: show voice register all Field Descriptions

Field	Description
Pool Tag	Used with the **all** and **pool** keywords. Shows the assigned tag number of the current pool.
Config	Used with the **all** and **pool** keywords. Shows the voice register pool.
Network address and Mask	Used with the **all** and **pool** keywords. Shows network address and mask information if the **id** command is configured.
Number list, Pattern, and Preference	Used with the **all** and **pool** keywords. Shows the **number** command configuration.
Proxy IP address	Used with the **all** and **pool** keywords. Shows the **proxy** command configuration.
Default preference	Used with the **all** and **pool** keywords. Shows the default preference value of this pool.
Incoming called number	Used with the **all** and **pool** keywords. Shows the **incoming called-number** command configuration.
Translate outgoing called tag	Used with the **all** and **pool** keywords. Shows the **translate-outgoing** command configuration.
Class of Restriction List Tag	Used with the **all** and **pool** keywords. Shows the COR tag.
Incoming corlist name	Used with the **all** and **pool** keywords. Shows the **cor** command configuration.
Application	Used with the **all** and **pool** keywords. Shows the **application** command configuration for this pool.
Dialpeers created:	Used with the **all** and **pool** keywords. What follows is a list of all dial peers created and their contents. Dial-peer contents differ per application and are not described here.
Statistics	Used with the **all**, **pool**, and **statistics** keywords. Shows the registration statistics for this pool.
Active registrations	Used with the **all**, **pool**, and **statistics** keywords. Shows the current active registrations.

Field	Description
Total Registration Statistics	Used with the **all**, **pool**, and **statistics** keywords. Shows the total registration statistics for this pool.
Registration requests	Used with the **all**, **pool**, and **statistics** keywords. Shows the incoming registration requests.
Registration success	Used with the **all**, **pool**, and **statistics** keywords. Shows the successful registrations.
Registration failed	Used with the **all**, **pool**, and **statistics** keywords. Shows the failed registrations.
unRegister requests	Used with the **all**, **pool**, and **statistics** keywords. Shows the incoming unregister/registration expire requests.
unRegister success	Used with the **all**, **pool**, and **statistics** keywords. Reports the number of successful unregisters.
unRegister failed	Used with the **all**, **pool**, and **statistics** keywords. Reports the number of failed unregisters.

Related Commands

Command	Description
show sip-ua status registrar	Displays all the SIP endpoints currently registered with the contact address.
show voice register dial-peers	Displays details of all dynamically created VoIP dial peers associated with the Cisco Unified SIP SRST or Cisco Unified CME register event
show voice register pool	Displays all configuration information associated with a particular voice register pool.

show voice register dial-peers

To display details of all dynamically created VoIP dial peers associated with the Cisco Unified Session Initiation Protocol (SIP) Survivable Remote Site Telephony (SRST) or Cisco Unified CallManager Express (Cisco Unified CME) register event, use the **show voice register dial-peers** command in privileged EXEC mode.

show voice register dial-peers [**pool** **tag**]

Syntax Description

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

Command Modes

Privileged EXEC

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 CME.
15.1(2)T	Cisco Unified CME 8.1 Cisco Unified SIP SRST 8.1	This command was modified. Pool tag keyword-argument was added. Command output display was also modified to display dial-peers specific to a pool.

Usage Guidelines

Use this command to display the dial-peers associated with a pool. To display the dynamic dial-peers associated with a specific pool, use the pool keyword followed by the pool tag. When using the pool keyword, you must specify the pool tag.

Examples

Cisco Unified CME adn Cisco Unified SIP SRST

The following is a sample output from this command displaying all dial-peers:

```
Router#show voice register dial-peers
Dial-peers for Pool 1
dial-peer voice 40001 voip
 destination-pattern 45111
 session target ipv4:8.3.3.111:5060
 session protocol sipv2
 call-fwd-all         4555
 after-hours-exempt   FALSE
dial-peer voice 40002 voip
  destination-pattern 45113
  session target ipv4:8.33.33.111:5060
  session protocol sipv2
  after-hours-exempt   FALSE
```

```
Dial-peers for Pool 2
 dial-peer voice 40003 voip
 destination-pattern 45112
 session target ipv4:8.33.33.112:5060
 session protocol sipv2
 call-fwd-noan-timeou 8
 call-fwd-noan        999
 after-hours-exempt   TRUE
```

Cisco Unified CME and Cisco Unified SRST

The following is a sample output from this command displaying all statistical information related to pool 1:

```
Router# show voice register dial-peers pool 1
Dial-peers for Pool 1:
dial-peer voice 40004 voip
 destination-pattern 1000
 redirect ip2ip
 session target ipv4:9.13.18.40:19633
 session protocol sipv2
 dtmf-relay rtp-nte sip-notify
 digit collect kpml
 codec  g711ulaw bytes 160
  after-hours-exempt   FALSE
dial-peer voice 40001 voip
 destination-pattern 2000
 redirect ip2ip
 session target ipv4:9.13.18.40:19634
 session protocol sipv2
 dtmf-relay rtp-nte sip-notify
 digit collect kpml
 codec  g711ulaw bytes 160
```

after-hours-exempt FALSE

Related Commands

Command	Description
show sip-ua status registrar	Displays all the SIP endpoints currently registered with the contact address.
show voice register all	Displays all Cisco Unified SIP SRST and Cisco Unified CME configurations and register information.
show voice register pool	Displays all configuration information associated with a particular voice register pool.

show voice register dn

To display all configuration information associated with a specific voice register dn, use the **show voice register dn** command in privileged EXEC mode.

show voice register dn {*tag* | **all**}

Syntax Description

tag	Tag number of the voice register dn for which to display information. Range is 1 to 750.
all	(Optional) Displays configuration information associated with all voice register dns defined in a system.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Version	Modification
12.4(4)T	Cisco CME 3.4 Cisco SIP SRST 3.4	This command was introduced.
15.1(2)T	Cisco CME 8.1 Cisco SIP SRST 8.1	This command was modified.The display output now shows pools that have DNs configured under them. All keyword was added to show configuration information for all voice register dns defined in system.

Usage Guidelines

In Cisco Unified CME 8.1 and Cisco Unified SIP SRST 8.1, the show voice register dn command displays the pools that have the DNs configured under them. When used with all keyword, the show voice register dn command displays configuration information for all the DNs defined in a system.

Examples

Cisco Unified SIP CME

The following is a sample output from this command:

```
Router# show voice register dn 1
Dn Tag 1
Config:
  Number is 11
  Preference is 10
  Huntstop is enabled
  Translation-profile incoming saaa
  Allow watch is enabled
  Pool 1    has this DN configured for line 1
```

Cisco Unified SIP SRST

The following is a sample output from this command:

```
Router# show voice register dn 2
Dn Tag 1
Config:
```

```
  Number is 11
  Preference is 10
  Huntstop is enabled
  Translation-profile incoming saaa
  Allow watch is enabled
  Pool 1    has this DN configured for line 1
```

Cisco Unified SIP SRST

The following is a sample output from this command displaying information for all the dns:

```
Dn Tag 1
Config:
  Number is 11
  Preference is 10
  Huntstop is enabled
  Translation-profile incoming saaa
  Allow watch is enabled
  Pool 1    has this DN configured for line 1
Dn Tag 2
Config:
  Number is 12
  Preference is 1
  Huntstop is enabled
  Allow watch is enabled
  Pool 2    has this DN configured for line 1, 2
```

Cisco Unified SIP CME

The following is a sample output from this command displaying information for all the dns:

```
Router# show voice register dn all
Dn Tag 1
Config:
  Number is 45111
  Preference is 0
  Huntstop is disabled
  Auto answer is disabled
Dn Tag 2
Config:
  Number is 45112
  Preference is 0
  Huntstop is disabled
  Auto answer is disabled
  call-forward b2bua noan 999 timeout 8
  after-hour exempt
  Pool 2    has this DN configured for line 1
  Pool 7    has this DN configured for line 1
Dn Tag 3
Config:
  Number is 45113
  Preference is 0
  Huntstop is disabled
  Auto answer is disabled
  call-forward b2bua all 87687
  Preference is 0
  Huntstop is disabled
  Auto answer is disabled
  call-forward b2bua all 87687
```

```
  Pool 1    has this DN configured for line 1
  Pool 3    has this DN configured for line 1, 2
Dn Tag 4
Config:
  Auto answer is disabled
Dn Tag 7
Config:
  Number is 451110
  Preference is 0
  Huntstop is disabled
  Auto answer is disabled
  after-hour exempt
  Pool 1    has this DN configured for line 4
Dn Tag 8
Config:
  Auto answer is disabled
  call-forward b2bua all 678
  after-hour exempt
  Pool 1    has this DN configured for line 3
```

contains descriptions of significant fields shown in this output, listed in alphabetical order.

Table 17: show voice register dn Field Descriptions

Field	Description
Auto answer	Status of auto-answer feature defined with the **auto-answer** command.
Config	List of configuration options defined for this voice register dn.
Dn Tag	Tag number of the requested voice register dn.
Huntstop	Status of huntstop behavior defined with the **huntstop** command.
Number	Telephone or extension number set with the **number** command in voice register dn configuration mode.
Preference	Preference order set with the **preference** command in voice register dn configuration mode.

Related Commands

Command	Description
show voice register pool	Displays all configuration information associated with a particular voice register pool.
show voice register dn all	Displays information associated with all the dns configured in a system.
voice register dn	Enters voice register dn configuration mode to define an extension for a SIP phone line.

show voice register global

To display all global configuration parameters associated with SIP phones, use the **show voice register global** command in privileged EXEC mode.

show voice register global

Syntax Description

This command has no arguments or keywords.

Command Default

Privileged EXEC

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.
15.0(1)XA	Cisco SIP SRST 8.0	This command was updated to display the signaling transport protocol.
15.1(2)T	Cisco Unified CME 8.1 Cisco Unified SIP SRST 8.1	This command was modified.The output display now includes global statistics.
Cisco IOS XE Amsterdam 17.2.1r	Unified SRST 12.8	This command was modified to display VRF information for Cisco 4000 Series Integrated Services Routers.

Cisco Unified CME

The following is sample output from this command:

```
Router# show voice register global
CONFIG [Version=8.1]
========================
  Version 8.1
  Mode is cme
  Max-pool is 10
  Max-dn is 10
  Outbound-proxy is enabled and will use global configured value
  Security Policy: DEVICE-DEFAULT
  Source-address is 8.3.3.5 port 5060
  Time-format is 12
  Date-format is M/D/Y
  Time-zone is 5
  Hold-alert is disabled
  Mwi stutter is disabled
  Mwi registration for full E.164 is disabled
  Forwarding local is enabled
  Privacy is enabled
  Privacy-on-hold is disabled
  Dst auto adjust is enabled
    start at Apr week 1 day Sun time 02:00
    stop  at Oct week 8 day Sun time 02:00
  Max redirect number is 5
  IP QoS DSCP:
```

```
    ef (the MS 6 bits, 46, in ToS, 0xB8) for media
    cs3 (the MS 6 bits, 24, in ToS, 0x60) for signal
    af41 (the MS 6 bits, 34, in ToS, 0x88) for video
    default (the MS 6 bits, 0, in ToS, 0x0) for service
  Telnet Level: 0
  Tftp path is flash:
  Generate text file is disabled
  Tftp files are created, current syncinfo 0001140473454008
  OS79XX.TXT is not created
  timeout interdigit 10
  network-locale[0] US    (This is the default network locale for this box)
  network-locale[1] US
  network-locale[2] US
  network-locale[3] US
  network-locale[4] US
  user-locale[0] US    (This is the default user locale for this box)
  user-locale[1] US
  user-locale[2] US
  user-locale[3] US
  user-locale[4] US   Active registrations  : 0
  Total SIP phones registered: 0
  Total Registration Statistics
    Registration requests  : 0
    Registration success   : 0
    Registration failed    : 0
    unRegister requests    : 0
    unRegister success     : 0
    unRegister failed      : 0
    Attempts to register
           after last unregister : 0
    Last register request time   :
    Last unregister request time :
    Register success time        :
    Unregister success time      :
```

VRF for Unified SRST 12.8

```
Router# show voice register global
CONFIG [Version=12.8]
========================
  Version 12.8
 Mode is srst
 Max-pool is 50
 Max-dn is 50
 VRF vrf1
 Outbound-proxy is enabled and will use global configured value
 Security Policy: DEVICE-DEFAULT
 Allow-hash-in-dn is disabled
 Forced Authorization Code Refer is enabled
 timeout interdigit 10
 timeout transfer recall 0
 network-locale[0] US  (This is the default network locale for this box)
 network-locale[1] US
 network-locale[2] US
 network-locale[3] US
 network-locale[4] US
 user-locale[0] US  (This is the default user locale for this box)
 user-locale[1] US
 user-locale[2] US
 user-locale[3] US
 user-locale[4] US
```

```
 MWI unsolicited notify is disabled
 Active registrations : 0

 Total SIP phones registered: 0
 Total Registration Statistics
  Registration requests : 0
  Registration success  : 0
  Registration failed  : 0
  unRegister requests  : 0
  unRegister success   : 0
  unRegister failed   : 0
  Auto-Register requests : 0
  Attempts to register
     after last unregister : 0
  Last register request time  :
  Last unregister request time :
  Register success time    :
  Unregister success time   :
```

Cisco Unified SIP SRST

```
Router# show voice register global
CONFIG [Version=8.1]
========================
  Version 8.1
  Mode is srst
  Max-pool is 10
  Max-dn is 10
  Outbound-proxy is enabled and will use global configured value
  Security Policy: DEVICE-DEFAULT
  timeout interdigit 10
  network-locale[0] US    (This is the default network locale for this box)
  network-locale[1] US
  network-locale[2] US
  network-locale[3] US
  network-locale[4] US
  user-locale[0] US    (This is the default user locale for this box)
  user-locale[1] US
  user-locale[2] US
  user-locale[3] US
  user-locale[4] US   Active registrations  : 0
  Total SIP phones registered: 0
  Total Registration Statistics
    Registration requests  : 0
    Registration success   : 0
    Registration failed    : 0
    unRegister requests    : 0
    unRegister success     : 0
    unRegister failed      : 0
    Attempts to register
           after last unregister : 0
    Last register request time   :
    Last unregister request time :
    Register success time        :
    Unregister success time      :
```

Table 18: show voice register global Field Descriptions, on page 127contains descriptions of significant fields shown in this output, listed in alphabetical order.

Table 18: show voice register global Field Descriptions

Field	Description
Date-format	Value of **date-format** command.
DST auto adjust	Setting of **dst auto-adjust** command.
Forwarding local	Setting of **forwarding local** command.
Generate text file	Setting of **text file** command.
Hold-alert	Setting of **hold-alert** command.
Load	Value of **load** command.
Max-dn	Reports the maximum number of SIP voice register directory numbers (dns) supported by the Cisco Unified SIP CME or Cisco Unified SIP SRST router as configured with the **max-dn** command. The maximum possible number is platform-dependent.
Max-pool	Reports the maximum number of SIP voice register pools supported by the Cisco Unified SIP SRST or Cisco Unified CME router as configured with the **max-pool** command. The maximum possible number is platform-dependent.
Max redirect number	Maximum number of redirects set with the **max-redirect** command.
Mode	Reports the mode as configured with the **mode** command. Value can be either Cisco Unified CME or Cisco Unified SIP SRST.
MWI registration	Setting of **mwi** command.
MWI stutter	Setting of **mwi stutter** command.
Time-format	Value of **time-format** command.
Time-zone	Number of the timezone selected with the **timezone** command.
TFTP path	Directory location of provisioning files for SIP phones that is specified with the **tftp-path** command.
Version	Reports the Cisco Unified SIP SRST or Cisco Unified CME version number.
VRF	Displays information on the associated VRF ID.

Related Commands

Command	Description
show sip-ua status registrar	Displays all the SIP endpoints currently registered with the contact address.
show voice register all	Displays all Cisco Unified SIP SRST and Cisco Unified CME configurations and register information.
show voice register dial-peers	Displays details of all dynamically created VoIP dial peers associated with the Cisco Unified SIP SRST or Cisco Unified CME register event.

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 CME or Cisco Unified SIP SRST environment.

show voice register pool

To display all configuration information associated with a specific voice register pool, use the **show voice register pool** command in privileged EXEC mode.

show voice register pool {*pool-tag* | **all**} [**brief**]

Syntax Description

pool-tag	Tag number of the voice register pool for which information is displayed. Range is 1 to 262. **Note** The maximum number of pools is version and platform dependent.
all	Displays the information of all the voice register pools.
brief	(Optional) Displays brief information of all voice register pools.

Command Modes

Privileged EXEC (#)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco SIP SRST	This command was introduced.
12.3(4)T	Cisco SIP SRST	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 CME.
12.4(15)XY	Cisco Unified CME 4.2(1) Cisco Unified SIP SRST 4.2(1)	This command was modified to include emergency response location information in the output display.
12.4(20)T	Cisco Unified CME 7.0 Cisco Unified SIP SRST 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.
15.0(1)XA	Cisco Unified CME 8.0	This command was modified to include logical partitioning class of restriction (LPCOR) information in the output display.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.
15.1(2)T	Cisco Unified CME 8.1	This command was modified. The **all** and **brief** keywords were added. Voice-class stun-usage information is displayed in the output.
15.2(2)T	Cisco Unified CME 9.0	This command was modified to include conference admin, conference add mode, and conference drop mode in the output display.

Cisco IOS Release	Cisco Product	Modification
15.2(4)M	Cisco Unified CME 9.1 Cisco Unified SIP SRST 9.1	This command was modified to include Key Expansion Module (KEM) data in the output display.
Cisco IOS XE Everest 16.6.1	Unified SRST 12.0	This command was modified to include the IPv6 address in the output display for Unified SRST.
Cisco IOS XE Amsterdam 17.2.1r	Unified SRST 12.8	This command was modified to include VRF information for Cisco 4000 Series Integrated Services Routers: • **show voice register pool** *all*—Information for all the available configured pools. • **show voice register pool** *pool-tag*—Information specific to a specific configured pool.
Cisco IOS XE Amsterdam 17.2.1r	Unified SRST 12.8	Introduced support for YANG models.

Examples

Cisco Unified CME

The following is a sample output of the **show voice register pool** command, displaying information for voice register pool 33 in Cisco Unified CME:

```
Router# show voice register pool 33

Pool Tag 33
Config:
 Mac address is 0009.B7F7.532E
 Type is 7960
 Number list 1 : DN 1
 Number list 2 : DN 2
 Number list 3 : DN 3
 Proxy Ip address is 0.0.0.0
 DTMF Relay is disabled
 Call Waiting is enabled
 DnD is disabled
 Busy trigger per button value is 0
 keep-conference is enabled
 template is 1
 Emergency response location 3
 Lpcor Type is local
 Lpcor Incoming is sip_group
 Lpcor Outgoing is sip_group

 Transport type is udp
 service-control mechanism is not supported
 Privacy feature is not configured.
 Privacy button is disabled

Dialpeers created:

Statistics:
 Active registrations : 0
```

```
 Total SIP phones registered: 0
 Total Registration Statistics
  Registration requests : 0
  Registration success : 0
  Registration failed : 0
  unRegister requests : 0
  unRegister success : 0
  unRegister failed : 0
```

The following is a sample output of the **show voice register pool** command. The output shows that a meet-me hardware conference administrator has been assigned, the conference creator or any of the participants can add a new participant, and the conference creator can terminate the active video hardware conference by hanging up.

```
Router# show voice register pool 15
 Pool Tag 15
Config:
  Mac address is 1C17.D340.81F0
  Type is 9951
  Number list 1 : DN 15
  Proxy Ip address is 0.0.0.0
  Current Phone load version is Cisco-CP9951/9.0.1
  DTMF Relay is enabled, sip-notify
  Call Waiting is enabled
  DnD is disabled
  Video is enabled
  Camera is enabled
  Busy trigger per button value is 0
  feature-button 5 DnD
  feature-button 6 MeetMe
  keep-conference is enabled
  registration expires timer max is 86400 and min is 60
  template is 1
  kpml signal is enabled
  Lpcor Type is none
  Transport type is udp
  service-control mechanism is supported
  registration Call ID is 1c17d340-81f00002-6c48fe8e-03013c10@1.5.40.105
  Registration method: per line
  Privacy feature is not configured.
  Privacy button is disabled
  active primary line is: 3915
  contact IP address: 1.5.40.105 port 5060
  Phone SIS Version:  5.0.0
  GW SIS Version:  1.0.0
  conference admin: yes
  conference add mode: all
  conference drop mode: creator
  paging-dn: config 0 [multicast]  effective 0 [multicast]
...
```

The following is an example of a partial output of the **show voice register pool all** command, showing KEM data with the phone type information:

```
Router# show voice register pool all
Pool Tag 5
Config:
Mac address is B4A4.E328.4698
Type is 9971 addon 1 CKEM
Number list 1 : DN 2
```

```
Number list 2 : DN 3
Proxy Ip address is 0.0.0.0
DTMF Relay is disabled
Call Waiting is enabled
DnD is disabled
Video is enabled
Camera is enabled
Busy trigger per button value is 0
keep-conference is enabled
registration expires timer max is 200 and min is 60
kpml signal is enabled
Lpcor Type is none
```

The following is a sample output of the **show voice register pool all** command, showing the three KEMs configured with phone type 9971:

```
Router# show voice register pool all
Pool Tag 4
Config:
Mac address is B4A4.E328.4698
Type is 9971 addon 1 CKEM 2 CKEM 3 CKEM
Number list 1 : DN 4
Number list 2 : DN 5
Number list 3 : DN 9
```

Cisco Unified SIP SRST

The following is a sample output of the **show voice register pool** command, displaying all information for voice register pool 1 in Cisco Unified SIP SRST:

```
Router# show voice register pool 1

Pool Tag 1
Config:
 Network address is 192.168.0.0, Mask is 255.255.0.0
 Number list 1 : Pattern is 50.., Preference is 2
 Proxy Ip address is 0.0.0.0
 Default preference is 2
 Incoming called number is
 Translate outgoing called tag is 1
 Class of Restriction List Tag: default
 Incoming corlist name is allowall
 Application is default.new

Dialpeers created:

dial-peer voice 40007 voip
 application default.new
 corlist incoming allowall
 preference 2
 incoming called-number 5001
 destination-pattern 5001
 redirect ip2ip
 session target ipv4:192.168.0.3
 session protocol sipv2
 translate-outgoing called 1
 voice-class codec 1

Statistics:
 Active registrations : 2
```

```
 Total Registration Statistics
  Registration requests : 48
  Registration success : 48
  Registration failed : 0
  unRegister requests : 46
  unRegister success : 46
  unRegister failed : 0

Emergency response location 6
```

VRF for Unified SRST 12.8

The following is a sample output of the **show voice register pool** *all* and **show voice register pool** *pool-tag* command for Unified SRST 12.8:

```
Router# show voice register pool all

Pool Tag 1
Config:
 Mac address is 9C57.ADF5.C191
 Number list 1 : DN 1
 Proxy Ip address is 0.0.0.0
 DTMF Relay is enabled, rtp-nte
 kpml signal is enabled
 Lpcor Type is none

 Reason for unregistered state:
   No registration request since last reboot/unregister

 paging-dn: config 0 [multicast] effective 0 [multicast]

VRF:
  vrf1

Dialpeers created:

Statistics:
 Active registrations : 0

 Total SIP phones registered: 0
 Total Registration Statistics
  Registration requests : 0
  Registration success  : 0
  Registration failed  : 0
  unRegister requests  : 0
  unRegister success   : 0
  unRegister failed   : 0
  Auto-Register requests : 0
  Attempts to register
     after last unregister : 0
  Last register request time  :
  Last unregister request time :
  Register success time    :
  Unregister success time   :


Pool Tag 2
Config:
 Mac address is 9C57.ADF5.C192
 Number list 1 : DN 2
```

```
 Proxy Ip address is 0.0.0.0
 DTMF Relay is enabled, rtp-nte
 kpml signal is enabled
 Lpcor Type is none

 Reason for unregistered state:
   No registration request since last reboot/unregister

 paging-dn: config 0 [multicast] effective 0 [multicast]

VRF:
  vrf1

Dialpeers created:

Statistics:
 Active registrations : 0

 Total SIP phones registered: 0
 Total Registration Statistics
  Registration requests : 0
  Registration success  : 0
  Registration failed  : 0
  unRegister requests  : 0
  unRegister success   : 0
  unRegister failed   : 0
  Auto-Register requests : 0
  Attempts to register
     after last unregister : 0
  Last register request time  :
  Last unregister request time :
  Register success time    :
  Unregister success time   :



Router# show voice register pool 1
 Pool Tag 1
Config:
 Mac address is 9C57.ADF5.C191
 Number list 1 : DN 1
 Proxy Ip address is 0.0.0.0
 DTMF Relay is enabled, rtp-nte
 kpml signal is enabled
 Lpcor Type is none

 Reason for unregistered state:
   No registration request since last reboot/unregister

 paging-dn: config 0 [multicast] effective 0 [multicast]

VRF:
  vrf1

Dialpeers created:

Statistics:
 Active registrations : 0

 Total SIP phones registered: 0
 Total Registration Statistics
  Registration requests : 0
  Registration success  : 0
  Registration failed  : 0
  unRegister requests  : 0
```

```
  unRegister success   : 0
  unRegister failed   : 0
  Auto-Register requests : 0
  Attempts to register
     after last unregister : 0
  Last register request time  :
  Last unregister request time :
  Register success time    :
  Unregister success time   :
```

The following is a sample output of the **show voice register pool brief** command, showing an IPv6 source address configured on a Cisco SIP IP Phone:

```
Router# show voice register pool brief
Pool ID                        IP Address                 Ln DN  Number     State
==== ======================== ========================== == === ========== ============
1    8.0.0.0                                                                UNREGISTERED
2    2001:420:54FF:13::312:0   2001:420:54FF:13::312:1           10001$     REGISTERED
```

Voice class stun usage

The following is a sample output of the **show voice register pool** command, displaying voice-class stun-usage information for voice register pool 51:

```
Router# show voice register pool 51
 Pool Tag 51
Config:
  Mac address is 0011.209F.5D60
  Type is 7960
  Number list 1 : DN 51
  Proxy Ip address is 0.0.0.0
  Current Phone load version is Cisco-SIPGateway/IOS-12.x
  DTMF Relay is disabled
  Call Waiting is enabled
  DnD is disabled
  Busy trigger per button value is 0
  keep-conference is enabled
  template is 10
  Lpcor Type is none

Transport type is udp
  service-control mechanism is not supported
  registration Call ID is 2BA38EE3-17D311DB-800BCD81-A9AD11F0
  Privacy feature is not configured.
  Privacy button is disabled
  active primary line is: 16263646
  contact IP address: 192.168.0.87 port 5060
  Reason for unregistered state:
         No registration request since last reboot/unregister
  voice-class stun-usage is enabled. tag is 1
Dialpeers created:
Dial-peers for Pool 51:
Statistics:
  Active registrations  : 0
  Total SIP phones registered: 0
  Total Registration Statistics
    Registration requests  : 2
    Registration success   : 2
```

```
Registration failed    : 0
unRegister requests    : 2
unRegister success     : 2
unRegister failed      : 0
Attempts to register
       after last unregister : 0
Last register request time   : 13:43:27.839 IST Tue Apr 20 2010
```

Table 19: show voice register pool Field Descriptions (Continued), on page 136 contains descriptions of significant fields shown in the Cisco Unified CME and Cisco Unified SIP SRST output, listed in alphabetical order.

Table 19: show voice register pool Field Descriptions (Continued)

Field	Description
Active registrations	Shows the current active registrations.
Application	Shows the **application** command configuration for this pool.
Call Waiting	Shows the **call-waiting** command configuration.
Class of Restriction List Tag	Shows the COR tag.
Conference add mode	Shows the current setting of the hardware conference privilege for adding participants.
Conference admin	Shows whether the Cisco Unified SIP IP phone is assigned as the hardware conference administrator or not.
Conference drop mode	Shows who can terminate an active ad-hoc hardware conference by hanging up.
Config	Shows the voice register pool.
Default preference	Shows the default preference value of this pool.
Dialpeers created	Lists all the dial peers created and their contents. Dial-peer contents differ for each application and are not described here.
DnD	Shows the setting of the **dnd-control** command.
DTMF Relay	Shows the setting of the **dtmf-relay** command.
Emergency response location	Shows the ephone’s emergency response location to which an emergency response team is dispatched when an emergency call is made.
Incoming called number	Shows the **incoming called-number** command configuration.
Incoming corlist name	Shows the **cor** command configuration.
keep-conference	Shows the status of the **keep-conference** command.
Lpcor Incoming	Shows the setting of the **lpcor incoming** command.
Lpcor Outgoing	Shows the setting of the **lpcor outgoing** command.

Field	Description
Lpcor Type	Shows the setting of the **lpcor type** command.
Mac address	Shows the MAC address of the Cisco Unified SIP IP phone as defined by the **id** command.
Network address and Mask	Shows network address and mask information when the **id** command is configured.
Number list, Pattern, and Preference	Shows the **number** command configuration.
Pool Tag	Shows the assigned tag number of the current pool.
Proxy IP address	Shows the **proxy** command configuration; that is, the IP address of the external SIP server.
Registration failed	Shows the failed registrations.
Registration requests	Shows the incoming registration requests.
Registration success	Shows the successful registrations.
Statistics	Shows the registration statistics for this pool.
Template	Shows the template-tag number for the template applied to the Cisco Unified SIP IP phone.
Total Registration Statistics	Shows the total registration statistics for this pool.
Translate outgoing called tag	Shows the **translate-outgoing** command configuration.
Type	Shows the phone type identified for the Cisco Unified SIP IP phone using the **type** command.
unRegister failed	Reports the number of failed unregisters.
unRegister requests	Shows the incoming unregister/registration expiry requests.
unRegister success	Reports the number of successful unregisters.
Username Password	Shows the values within the authentication credential.
VRF	Displays information on the associated VRF ID.

Related Commands

Command	Description
application (voice register pool)	Selects the session-level application for the dial peer associated with an individual Cisco Unified SIP IP phone in a Cisco Unified CME environment or for a group of phones in a Cisco Unified SIP SRST environment.
call-waiting (voice register pool)	Enables the call-waiting option on a SIP phone.

Command	Description
cor (voice register pool)	Configures a class of restriction on the VoIP dial peers associated with directory numbers.
dnd-control (voice register template)	Enables the Do-Not-Disturb (DND) soft key on SIP phones.
dtmf-relay (voice register pool)	Specifies the list of dual-tone multifrequency (DTMF) relay methods that can be used to relay DTMF audio tones between SIP endpoints.
id (voice register pool)	Explicitly identifies a locally available, individual Cisco Unified SIP IP phone or, when running Cisco Unified SIP SRST, a set of Cisco Unified SIP IP phones.
incoming called-number (dial peer)	Specifies a digit string that can be matched by an incoming call to associate the call with a dial peer.
keep-conference (voice register pool)	Allows IP phone conference initiators to exit from conference calls and keep the remaining parties connected.
lpcor incoming	Associates an incoming call with a logical partitioning class of restriction (LPCOR) resource-group policy.
lpcor outgoing	Associates an outgoing call with an LPCOR resource-group policy.
lpcor type	Specifies the LPCOR type for an IP phone.
number (voice register pool)	Indicates the E.164 phone numbers that the registrar permits to handle the Register message from a Cisco Unified SIP IP phone.
proxy (voice register pool)	Autogenerates additional VoIP dial peers to reach the main proxy whenever a Cisco Unified SIP IP phone registers with a Cisco Unified SIP SRST gateway.
show sip-ua status registrar	Displays all the Cisco Unified SIP IP phones registered with the contact address.
show voice register all	Displays all Cisco Unified SIP SRST and Cisco Unified CME configurations and register information.
show voice register dial-peer	Displays details of all dynamically created VoIP dial peers associated with the Cisco Unified CME or Cisco Unified SIP SRST register event.
translate-outgoing (voice register pool)	Allows an explicit setting of translation rules on the VoIP dial peer to modify a phone number dialed by any Cisco Unified IP phone user.
type (voice register pool)	Defines a phone type for a SIP phone.
voice register pool	Enters voice register pool configuration mode for Cisco Unified SIP IP phones.

show voice register pool attempted-registrations

To display the details of phones that attempt to register with Cisco Unified CME or Cisco Unified SRST and fail, use the **show voice register pool attempted-registrations** command in privileged EXEC mode.

show voice register pool attempted-registrations

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Version	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 display the details of the phones that attempt to register with Cisco Unified CME or Cisco Unified SRST and fail. If the phone registers successfully after some time, the attempted registration entry will still show up in the attempted-registration table. Use the clear voice register attempted-registrations command to remove the entry from the attempted registration table.

Examples

Cisco Unified CME and Cisco Unified SRST

The following is a sample output from this command displaying information for show voice register pool attempted-registrations:

```
Router# show voice register pool attempted-registrations
Phones that have attempted registrations and have failed:
 MAC address: 001b.535c.d410
 IP address : 8.3.3.111
 Attempts   : 5
 Time of first attempt : *10:49:51.542 UTC Wed Oct 14 2009
 Time of latest attempt: *10:50:00.886 UTC Wed Oct 14 2009
 Reason for failure    :
         No pool match for the registration request
 MAC address: 0015.c68e.6d13
 IP address : 8.33.33.112
 Attempts   : 4
 Time of first attempt : *10:49:53.418 UTC Wed Oct 14 2009
 Time of latest attempt: *10:50:00.434 UTC Wed Oct 14 2009
 Reason for failure    :
         No pool match for the registration request
 MAC address: 0009.43E9.0B35
 IP address : 9.13.40.83
 Attempts   : 1
 Time of first attempt : *10:49:57.866 UTC Wed Oct 14 2009
 Time of latest attempt: *10:49:57.866 UTC Wed Oct 14 2009
 Reason for failure    :
         No pool match for the registration request
```

The following is a sample output from this command displaying information for show voice register pool attempted-registrations when none of the phones fail:

```
Router# show voice register pool attempted-registrations
 Phones that have attempted registrations and have failed: NONE
```

Related Commands

Command	Description
attempted-registrations size	Allows to set the size of the table that stores information related to SIP phones that attempt to register and fail.
clear voice register attempted-registrations	Clears entries from the attempted-registration table.

show voice register pool connected

To display the details of SIP phones that are in connected state, use the **show voice register pool connected** command in privileged EXEC mode.

show voice register pool connected [**brief**]

Syntax Description

brief	(Optional) Displays brief details of SIP phones that are in connected state.

Command Modes

Privileged EXEC

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 display the details of the phone that are currently in connected state (in conversation). The output for show voice register pool connected command shows details of both calls originating from the SIP phones and calls made towards SIP phones. When used with brief keyword, the show voice register pool connected command displays a brief detail of phones in connected state.

Cisco Unified CME and Cisco Unified SRST

The following is sample output from this command displaying all statistical information:

```
Router# show voice register pool connected
Outbound calls from SIP line phones:
Pool tag: 1
==============
 MAC Address    : 001B.535C.D410
 Contact IP     : 8.3.3.111
 Phone Number   : 45111
 Remote Number  : 45112
Call 2
SIP Call ID                : 001b535c-d4100010-79612b5a-336b0db5@8.3.3.111
   State of the call       : STATE_ACTIVE (7)
   Substate of the call    : SUBSTATE_NONE (0)
   Calling Number          : 45111
   Called Number           : 45112
   Bit Flags               : 0xC0401C 0x100 0x4
   CC Call ID              : 7
   Source IP Address (Sig ): 8.3.3.5
   Destn SIP Req Addr:Port : [8.3.3.111]:5060
   Destn SIP Resp Addr:Port: [8.3.3.111]:50076
   Destination Name        : 8.3.3.111
   Number of Media Streams : 1
   Number of Active Streams: 1
   RTP Fork Object         : 0x0
   Media Mode              : flow-through
   Media Stream 1
     State of the stream      : STREAM_ACTIVE
     Stream Call ID           : 7
     Stream Type              : voice-only (0)
     Stream Media Addr Type   : 1
     Negotiated Codec         : g729r8 (20 bytes)
```

```
      Codec Payload Type       : 18
      Negotiated Dtmf-relay    : inband-voice
      Dtmf-relay Payload Type  : 0
      QoS ID                   : -1
      Local QoS Strength       : BestEffort
      Negotiated QoS Strength  : BestEffort
      Negotiated QoS Direction : None
      Local QoS Status         : None
      Media Source IP Addr:Port: [8.3.3.5]:17580
      Media Dest IP Addr:Port  : [8.3.3.111]:26298
Options-Ping    ENABLED:NO    ACTIVE:NO
Inbound calls to SIP line phones:

Pool tag: 2
==============
 MAC Address    : 0015.C68E.6D13
 Contact IP     : 8.33.33.112
 Phone Number   : 45112
 Remote Number  : 45111
Call 3
SIP Call ID                : 4DA52F97-ADA311DE-8019803A-FF3E4CBC@8.3.3.5
   State of the call       : STATE_ACTIVE (7)
   Substate of the call    : SUBSTATE_NONE (0)
   Calling Number          : 45111
   Called Number           : 45112
   Bit Flags               : 0xC04018 0x100 0x80
   CC Call ID              : 8
   Source IP Address (Sig ): 8.3.3.5
   Destn SIP Req Addr:Port : [8.33.33.112]:5060
   Destn SIP Resp Addr:Port: [8.33.33.112]:5060
   Destination Name        : 8.33.33.112
   Number of Media Streams : 1
   Number of Active Streams: 1
   RTP Fork Object         : 0x0
   Media Mode              : flow-through
   Media Stream 1
     State of the stream      : STREAM_ACTIVE
     Stream Call ID           : 8
     Stream Type              : voice-only (0)
     Stream Media Addr Type   : 1
     Negotiated Codec         : g729r8 (20 bytes)
     Codec Payload Type       : 18
     Negotiated Dtmf-relay    : inband-voice
     Dtmf-relay Payload Type  : 0
     QoS ID                   : -1
     Local QoS Strength       : BestEffort
     Negotiated QoS Strength  : BestEffort
     Negotiated QoS Direction : None
     Local QoS Status         : None
     Media Source IP Addr:Port: [8.3.3.5]:16384
     Media Dest IP Addr:Port  : [8.33.33.112]:30040
```

The following is sample output from this command displaying brief statistical information:

```
Router# show voice register pool connected brief
Pool IP Address      Number               Remote Number
==== =============== ==================== ====================
1    8.3.3.111       45111                45112
Inbound calls to SIP line phones:
Pool IP Address      Number               Remote Number
==== =============== ==================== ====================
2    8.33.33.112     45112                45111
```

Related Commands

Command	Description
show sip-ua calls	Displays active user agent client (UAC) and user agent server (UAS) information on SIP calls
show voice register all	Displays all Cisco Unified SIP SRST and Cisco Unified CME configurations and register information.
show voice register pool	Displays all configuration information associated with a particular voice register pool.

show voice register pool ip

To display the details of a SIP phone with a specific IP address, use the **show voice register pool ip** command in privileged EXEC mode.

show voice register pool ip ip-address

Syntax Description

ip-address	IPv4 address of the SIP phone .

Command Modes

Privileged EXEC

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 display the details of a phone with a specific IP-address. When the pool ID is configured as a mac address or an IP address the registered pools contain the IP address information. The pool information is displayed if the IP addresses match.

When the pool ID is IP and the pool is unregistered, IP address configured under pool is compared with the input IP. When the pool ID is network contact, the IP address of each phone that is registered is compared with the input IP address.

Examples

Cisco Unified CME and Cisco Unified SRST

The following is sample output from this command displaying all statistical information:

```
Router# show voice register pool ip 8.3.3.111
Pool ID             IP Address      Ln DN  Number               State
==== =============== =============== == === ==================== ============
1    001B.535C.D410  8.3.3.111       1  1   45111                REGISTERED
                                     4  7   451110               UNREGISTERED
```

Table 20: show voice register pool ip Field Descriptions, on page 144 contains descriptions of significant fields shown in this output, listed in alphabetical order.

Table 20: show voice register pool ip Field Descriptions

Field	Description
DN	Voice register DN tag of the line.
ID	Phone identification (ID) address.
IP Address	IP address of the SIP phone.
LN	Line number of the telephone number.
Number	Number of the phones that have a mac address.

Field	Description
Pool	Tag ID of the pool.
State	Registration state of the line.

Related Commands

Command	Description
show voice register all	Displays all Cisco Unified SIP SRST and Cisco Unified CME configurations and register information.
show voice register pool	Displays all configuration information associated with a particular voice register pool.

show voice register pool mac

To display the details of voice register pool associated with a specific phone type, use the **show voice register pool mac** command in privileged EXEC mode.

show voice register pool mac H.H.H

Syntax Description

H.H.H	MAC address of the SIP phone attempting to register.

Command Modes

Privileged EXEC

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 display the details of the phone with the mac address H.H.H. The command displays only the pools that are configured with an ID as mac.

Examples

Cisco Unified CME and Cisco Unified SRST

The following is sample output from this command displaying all statistical information:

```
Router# show voice register pool mac 001B.535C.D410
Pool ID              IP Address      Ln  DN  Number                State
==== =============== =============== == === ==================== ============
1    001B.535C.D410  8.3.3.111       1   1   45111                REGISTERED
                                     4   7   451110               UNREGISTERED
```

Table 21: show voice register pool mac Field Descriptions, on page 146 contains descriptions of significant fields shown in this output, listed in alphabetical order.

Table 21: show voice register pool mac Field Descriptions

Field	Description
DN	Voice register DN tag of the line.
ID	Phone identification (ID) address.
IP Address	IP address of the SIP phone.
LN	Line number of the telephone number.
Number	Number of the phones that have a mac address.
Pool	Tag ID of the pool.
State	Registration state of the line.

Related Commands

Command	Description
show voice register all	Displays all Cisco Unified SIP SRST and Cisco Unified CME configurations and register information.
show voice register pool	Displays all configuration information associated with a particular voice register pool.

show voice register pool network

To display the details of a phone with a specific network address, use the **show voice register pool network** command in privileged EXEC mode.

show voice register pool network *network-address*

Syntax Description

network-address	Network address of the SIP phone.

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

Use this command to display the details of pools that have network ID configured and whose network address matches the specific network address provided by the user.

Examples

The following is sample output from this command displaying all statistical information:

```
Router# show voice register pool network 78.89.0.0
Pool ID              IP Address      Ln DN  Number               State
==== =============== =============== == === ==================== ============
7    78.89.0.0                       1  1   6576                 UNREGISTERED
```

Table 22: show voice register pool network Field Descriptions, on page 148 contains descriptions of significant fields shown in this output, listed in alphabetical order.

Table 22: show voice register pool network Field Descriptions

Field	Description
DN	Directory number of the phone.
ID	Phone identification (ID) address.
IP Address	IP address and port number of the phones
LN	Line number of the phone.
Number	Number of the phone that have network address.
Pool	Shows the current pool.
State	Registration state.

Related Commands

Command	Description
show voice register all	Displays all Cisco Unified SIP SRST and Cisco Unified CME configurations and register information.
show voice register pool	Displays all configuration information associated with a particular voice register pool.
show voice register pool ip	Displays the voice register pool details of a phone with a specific IP address.

show voice register pool on-hold

To display the details of phones that are currently on-hold, use the **show voice register pool oh-hold** command in privileged EXEC mode.

show voice register pool on-hold [**brief**]

Syntax Description

brief	(Optional) Displays brief details of SIP phones that are currently on-hold.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Version	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 display the details of the phone that are currently on-hold. The show voice register pool on-hold command output also displays a field to show if the hold was a locally initiated hold (initiated on the phone) or if the hold was initiated on the remote end. When used with brief keyword, the show voice register pool on-hold command displays a brief information of the phones that are currently put on hold by the remote caller or have put the remote caller on hold. The “Hold-Origin” field specifies the type of the hold, which can be either remote or local. Local indicates that the call is placed on hold by the local phone and remote indicates that call is placed on hold by the remote phone. In case of double-hold, the hold origin will display the value “Local and Remote”.

Examples

Cisco Unified CME and Cisco Unified SRST

The following is a sample output from this command displaying information for phones ringing in a voice register pool:

```
Router# show voice register pool on-hold brief
Outbound calls from SIP line phones:
Pool IP Address      Number               Remote Number        Hold Origin
==== =============== ==================== ==================== ==============
1    8.3.3.111       45111                45112                Remote & Local
Inbound calls to SIP line phones:
Pool IP Address      Number               Remote Number        Hold Origin
==== =============== ==================== ==================== ==============
2    8.33.33.112     45112                45111                Remote & Local
```

Cisco Unified CME and Cisco Unified SRST

The following is a sample output from this command displaying information for phones on-hold:

```
Router# show voice register pool on-hold
Outbound calls from SIP line phones:
```

```
Pool tag: 1
==============
 MAC Address    : 001B.535C.D410
 Contact IP     : 8.3.3.111
 Phone Number   : 45111
 Remote Number  : 45112
 Local Hold     : CALL HOLD Pressed on SIP Phone
Call 4
SIP Call ID                : 001b535c-d4100010-79612b5a-336b0db5@8.3.3.111
   State of the call       : STATE_ACTIVE (7)
   Substate of the call    : SUBSTATE_NONE (0)
   Calling Number          : 45111
   Called Number           : 45112
   Bit Flags               : 0xC0401C 0x10100 0x4
   CC Call ID              : 7
   Source IP Address (Sig ): 8.3.3.5
   Destn SIP Req Addr:Port : [8.3.3.111]:5060
   Destn SIP Resp Addr:Port: [8.3.3.111]:50076
   Destination Name        : 8.3.3.111
   Number of Media Streams : 1
   Number of Active Streams: 1
   RTP Fork Object         : 0x0
   Media Mode              : flow-through
   Media Stream 1
     State of the stream      : STREAM_ACTIVE
     Stream Call ID           : 7
     Stream Type              : voice-only (0)
     Stream Media Addr Type   : 1
     Negotiated Codec         : g729r8 (20 bytes)
     Codec Payload Type       : 18
     Negotiated Dtmf-relay    : inband-voice
     Dtmf-relay Payload Type  : 0
     QoS ID                   : -1
     Local QoS Strength       : BestEffort
     Negotiated QoS Strength  : BestEffort
     Negotiated QoS Direction : None
     Local QoS Status         : None
     Media Source IP Addr:Port: [8.3.3.5]:17580
     Media Dest IP Addr:Port  : [8.3.3.111]:26298
Options-Ping    ENABLED:NO    ACTIVE:NO
Inbound calls to SIP line phones:
Pool tag: 2
==============
 MAC Address    : 0015.C68E.6D13
 Contact IP     : 8.33.33.112
 Phone Number   : 45112
 Remote Number  : 45111
 Remote Hold    : SIP Phone has received CALL HOLD
Call 5
SIP Call ID                : 4DA52F97-ADA311DE-8019803A-FF3E4CBC@8.3.3.5
   State of the call       : STATE_ACTIVE (7)
   Substate of the call    : SUBSTATE_NONE (0)
   Calling Number          : 45111
   Called Number           : 45112
   Bit Flags               : 0xC04018 0x4100 0x80
   CC Call ID              : 8
   Source IP Address (Sig ): 8.3.3.5
   Destn SIP Req Addr:Port : [8.33.33.112]:5060
   Destn SIP Resp Addr:Port: [8.33.33.112]:5060
   Destination Name        : 8.33.33.112
   Number of Media Streams : 1
   Number of Active Streams: 1
   RTP Fork Object         : 0x0
   Media Mode              : flow-through
```

```
   Media Stream 1
     State of the stream      : STREAM_ACTIVE
     Stream Call ID           : 8
     Stream Type              : voice-only (0)
     Stream Media Addr Type   : 1
     Negotiated Codec         : g729r8 (20 bytes)
     Codec Payload Type       : 18
     Negotiated Dtmf-relay    : inband-voice
     Dtmf-relay Payload Type  : 0
     QoS ID                   : -1
     Local QoS Strength       : BestEffort
     Negotiated QoS Strength  : BestEffort
     Negotiated QoS Direction : None
     Local QoS Status         : None
     Media Source IP Addr:Port: [8.3.3.5]:16384
     Media Dest IP Addr:Port  : [8.33.33.112]:30040
Options-Ping    ENABLED:NO    ACTIVE:NO
```

Related Commands

Command	Description
show voice register all	Displays all Cisco SIP SRST and Cisco CME configurations and register information.
show sip-ua calls	Displays active user agent client (UAC) and user agent server (UAS) information on SIP calls
show voice register pool	Displays all configuration information associated with a particular voice register pool.

show voice register pool registered

To display the details of phones that successfully register to Cisco Unified Communications Manager Express (Cisco Unified CME), use the **show voice register pool registered** command in privileged EXEC mode.

show voice register pool registered

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC (#)

Command History

Cisco IOS Release	Version	Modification
15.1(2)T	Cisco Unified CME 8.1 Cisco Unified SRST 8.1	This command was introduced.
15.2(4)M	Cisco Unified CME 9.1 Cisco Unified SIP SRST 9.1	This command was modified to display Key Expansion Module (KEM) details with the phone type information.

Usage Guidelines

Use the **show voice register pool registered** command to display the details of phones that are successfully registered to Cisco Unified CME and Cisco Unified Survivable Remote Site Telephony (Cisco Unified SRST).

Examples

Cisco Unified CME

The following is a sample output displaying information for a registered voice register pool in Cisco Unified CME:

```
Router# show voice register pool registered
Pool Tag 1
Config:
  Mac address is 001B.535C.D410
  Type is 7960
  Number list 1 : DN 1
  Number list 3 : DN 8
  Number list 4 : DN 7
  Proxy Ip address is 0.0.0.0
  Current Phone load version is Cisco-CP7960G/8.0
  DTMF Relay is disabled
  Call Waiting is enabled
  DnD is disabled
  Busy trigger per button value is 0
  call-forward phone all is 4566
  call-forward b2bua all 4555
  keep-conference is enabled
  Lpcor Type is none
  Transport type is udp
  service-control mechanism is supported
  registration Call ID is 001b535c-d410790d-17a6877e-5d04bbc5@8.3.3.111
  Privacy feature is not configured.
  Privacy button is disabled
```

```
  active primary line is: 45111
  contact IP address: 8.3.3.111 port 5060
Dialpeers created:
Dial-peers for Pool 1:
dial-peer voice 40001 voip
 destination-pattern 45111
 session target ipv4:8.3.3.111:5060
 session protocol sipv2
  call-fwd-all         4555
  after-hours-exempt   FALSE
Statistics:
  Active registrations  : 1
  Total SIP phones registered: 1
  Total Registration Statistics
    Registration requests  : 1
    Registration success   : 1
    Registration failed    : 0
    unRegister requests    : 0
    unRegister success     : 0
    unRegister failed      : 0
    Attempts to register
           after last unregister : 0
    Last register request time   : *11:40:32.263 UTC Wed Oct 14 2009
    Last unregister request time :
    Register success time        : *11:40:32.267 UTC Wed Oct 14 2009
    Unregister success time      :
```

The following is a sample output displaying information for a registered voice register pool with a Cisco Unified 9971 Session Initiation Protocol (SIP) IP phone attached to a Cisco SIP IP Phone CKEM 36-Button Line Expansion Module:

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

Cisco Unified SRST

```
The following is a sample output displaying information for a registered voice register
pool in Cisco Unified SRST:
Router# show voice register pool registered
 Pool Tag 1
Config:
  Ip address is 9.13.18.40, Mask is 255.255.0.0
  Number list 1 : DN 1
  Number list 2 : DN 2
```

```
  Number list 3 : DN 3
  Number list 4 : DN 4
  Number list 5 : DN 5
  Number list 6 : DN 6
  Number list 7 : DN 7
  Proxy Ip address is 0.0.0.0
  DTMF Relay is enabled, rtp-nte, sip-notify
  kpml signal is enabled
  Lpcor Type is none
Dialpeers created:
Dial-peers for Pool 1:
dial-peer voice 40004 voip
 destination-pattern 1000
 redirect ip2ip
 session target ipv4:9.13.18.40:19633
 session protocol sipv2
 dtmf-relay rtp-nte sip-notify
 digit collect kpml
 codec  g711ulaw bytes 160
  after-hours-exempt   FALSE
dial-peer voice 40001 voip
 destination-pattern 2000
 redirect ip2ip
 session target ipv4:9.13.18.40:19634
 session protocol sipv2
 dtmf-relay rtp-nte sip-notify
 digit collect kpml
 codec  g711ulaw bytes 160
  after-hours-exempt   FALSE
dial-peer voice 40002 voip
 destination-pattern 3000
 redirect ip2ip
 session target ipv4:9.13.18.40:19635
 session protocol sipv2
 dtmf-relay rtp-nte sip-notify
 digit collect kpml
 codec  g711ulaw bytes 160
  after-hours-exempt   FALSE
dial-peer voice 40003 voip
 destination-pattern 4000
 redirect ip2ip
 session target ipv4:9.13.18.40:19636
 session protocol sipv2
 dtmf-relay rtp-nte sip-notify
 digit collect kpml
 codec  g711ulaw bytes 160
  after-hours-exempt   FALSE
Statistics:
  Active registrations  : 4
  Total SIP phones registered: 1
  Total Registration Statistics
    Registration requests  : 4
    Registration success   : 4
    Registration failed    : 0
    unRegister requests    : 0
    unRegister success     : 0
    unRegister failed      : 0
    Attempts to register
           after last unregister : 0
    Last register request time   : .05:22:55.604 UTC Tue Oct 6 2009
    Last unregister request time :
    Register success time        : .05:22:55.604 UTC Tue Oct 6 2009
    Unregister success time      :
```

Table 23: show voice register pool registered Field Descriptions (Continued), on page 156 contains descriptions of significant fields shown in the **show voice register pool registered** command output, listed in alphabetical order.

Table 23: show voice register pool registered Field Descriptions (Continued)

Field	Description
Active registrations	Shows the current active registrations.
Application	Shows the **application** command configuration for this pool.
Call Waiting	Shows the setting of the **call-waiting** command.
Class of Restriction List Tag	Shows the COR tag.
Config	Shows the voice register pool.
Current phone-load	Shows the current version of the phone load.
Default preference	Shows the default preference value of this pool.
Dialpeers created	Results in a list of all dial peers created and their contents. Dial-peer contents differ for each application and are not described here.
DnD	Shows the setting of the **dnd-control** command.
DTMF Relay	Shows the setting of the **dtmf-relay** command.
Emergency response location	Shows the ephone's emergency response location to which an emergency response team is dispatched when an emergency call is made.
Incoming called number	Shows the **incoming called-number** command configuration.
Incoming corlist name	Shows the **cor** command configuration.
keep-conference	Shows the status of the **keep-conference** command.
Lpcor Incoming	Shows the setting of the **lpcor incoming** command.
Lpcor Outgoing	Shows the setting of the **lpcor outgoing** command.
Lpcor Type	Shows the setting of the **lpcor type** command.
Mac address	Shows the MAC address of this SIP phone as defined by the **id** command.
Network address and Mask	Shows network address and mask information when the **id** command is configured.
Number list, Pattern, and Preference	Shows the **number** command configuration.
Pool Tag	Shows the assigned tag number of the current pool.
Previous phone-load	Shows the version of the previous phone load.

Field	Description
Proxy IP address	Shows the **proxy** command configuration; that is, the IP address of the external SIP server.
Registration failed	Shows the failed registrations.
Registration requests	Shows the incoming registration requests.
Registration success	Shows the successful registrations.
Statistics	Shows the registration statistics for this pool.
statistics time-stamps	Shows the registration statistics for this pool with specific time stamps.
Template	Shows the template-tag number for the template applied to this SIP phone.
Total Registration Statistics	Shows the total registration statistics for this pool.
Translate outgoing called tag	Shows the **translate-outgoing** command configuration.
Type	Shows the phone type identified for this SIP phone using the **type** command.
unRegister failed	Reports the number of failed unregisters.
unRegister requests	Shows the incoming unregister/registration expiry requests.
unRegister success	Reports the number of successful unregisters.
Username Password	Shows the values within the authentication credential.

Related Commands

Command	Description
application (voice register pool)	Selects the session-level application for the dial peer associated with an individual Cisco Unified SIP IP phone in a Cisco Unified CME environment or for a group of phones in a Cisco Unified SIP SRST environment.
call-waiting (voice register pool)	Enables the call-waiting option on a SIP phone.
cor (voice register pool)	Configures a class of restriction on the VoIP dial peers associated with directory numbers.
dnd-control (voice register template)	Enables the Do-Not-Disturb (DND) soft key on SIP phones.
dtmf-relay (voice register pool)	Specifies the list of dual-tone multifrequency (DTMF) relay methods that can be used to relay DTMF audio tones between SIP endpoints.
id (voice register pool)	Explicitly identifies a locally available, individual Cisco Unified SIP IP phone or, when running Cisco Unified SIP SRST, a set of Cisco Unified SIP IP phones.

Command	Description
incoming called-number (dial peer)	Specifies a digit string that can be matched by an incoming call to associate the call with a dial peer.
keep-conference (voice register pool)	Allows IP phone conference initiators to exit from conference calls and keep the remaining parties connected.
lpcor incoming	Associates an incoming call with a logical partitioning class of restriction (LPCOR) resource-group policy.
lpcor outgoing	Associates an outgoing call with an LPCOR resource-group policy.
lpcor type	Specifies the LPCOR type for an IP phone.
number (voice register pool)	Indicates the E.164 phone numbers that the registrar permits to handle the Register message from a Cisco Unified SIP IP phone.
proxy (voice register pool)	Autogenerates additional VoIP dial peers to reach the main proxy whenever a Cisco Unified SIP IP phone registers with a Cisco Unified SIP SRST gateway.
show voice register all	Displays all Cisco Unified SIP SRST and Cisco Unified CME configurations and register information.
show voice register dial-peers	Displays details of all dynamically created VoIP dial peers associated with the Cisco Unified SIP SRST or Cisco Unified CME register event.
show voice register pool	Displays all configuration information associated with a particular voice register pool.
show voice register pool unregistered	Displays the details of voice register pools that do not have any phones registered.
translate-outgoing (voice register pool)	Allows an explicit setting of translation rules on the VoIP dial peer to modify a phone number dialed by any Cisco Unified IP phone user.
type (voice register pool)	Defines a phone type for a SIP phone.
voice register pool	Enters voice register pool configuration mode for SIP phones.

show voice register pool ringing

To display the details of phones that are currently in ringing state, use the **show voice register pool ringing** command in privileged EXEC mode.

show voice register pool ringing [**brief**]

Syntax Description

brief	(Optional) Displays brief details of SIP phones that are currently in ringing state.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Version	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 display the details of the phone that are currently in ringing state. When used with the brief keyword, the show voice register pool ringing brief command only displays information related to calls that are bound towards the SIP phones.

Examples

Cisco Unified CME and Cisco Unified SRST

The following is a sample output from this command displaying information for phones ringing in a voice register pool:

```
Router# show voice register pool ringing brief
Pool IP Address      Number               Remote Number
==== =============== ==================== ====================
2    8.33.33.112     45112                45111
```

Cisco Unified CME and Cisco Unified SRST

The following is a sample output from this command displaying information for phones ringing in a voice register pool:

```
Router# show voice register pool ringing
Pool tag: 2
==============
 MAC Address    : 0015.C68E.6D13
 Contact IP     : 8.33.33.112
 Phone Number   : 45112
 Remote Number  : 45111
Call 1
SIP Call ID                 : C0B5DA7-ADA311DE-8011803A-FF3E4CBC@8.3.3.5
   State of the call        : STATE_RECD_PROCEEDING (4)
   Substate of the call     : SUBSTATE_PROCEEDING_PROCEEDING (2)
   Calling Number           : 45111
   Called Number            : 45112
   Bit Flags                : 0xC00018 0x100 0x280
```

```
CC Call ID              : 5
Source IP Address (Sig ): 8.3.3.5
Destn SIP Req Addr:Port : [8.33.33.112]:5060
Destn SIP Resp Addr:Port: [8.33.33.112]:5060
Destination Name        : 8.33.33.112
Number of Media Streams : 1
Number of Active Streams: 1
RTP Fork Object         : 0x0
Media Mode              : flow-through
Media Stream 1
  State of the stream      : STREAM_ACTIVE
  Stream Call ID           : 5
  Stream Type              : voice+dtmf (1)
  Stream Media Addr Type   : 1
  Negotiated Codec         : No Codec    (0 bytes)
  Codec Payload Type       : 255 (None)
  Negotiated Dtmf-relay    : inband-voice
  Dtmf-relay Payload Type  : 0
  QoS ID                   : -1
  Local QoS Strength       : BestEffort
  Negotiated QoS Strength  : BestEffort
  Negotiated QoS Direction : None
  Local QoS Status         : None
  Media Source IP Addr:Port: [8.3.3.5]:16882
```

Related Commands

Command	Description
show sip-ua calls	Displays active user agent client (UAC) and user agent server (UAS) information on SIP calls
show voice register all	Displays all Cisco SIP SRST and Cisco CME configurations and register information.
show voice register pool	Displays all configuration information associated with a particular voice register pool.

show voice register pool telephone-number

To display the details of a phone line with a specific telephone-number, use the **show voice register pool telephone-number** command in privileged EXEC mode.

show voice register pool telephone-number number

Syntax Description

number	Number identifying a specific phone.

Command Modes

Privileged EXEC

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 display the details of the phone line with the specified telephone-number. If the line is registered, the contact ip address will be displayed. When the phone line is not registered and the pool ID type is network IP, the IP address is not displayed. When the phone line is not registered but some other line is registered for the same pool with MAC or IP address, then the IP address is displayed.

Cisco Unified CME

The following is a sample output from this command displaying all statistical information:

```
Router# show voice register pool telephone number 45112
Pool ID              IP Address      Ln DN  Number                State
==== =============== =============== == === ==================== ============
2    0015.C68E.6D13                  1  2   45112                UNREGISTERED
7    0018.BAC8.D2B1                  1  2   45112                UNREGISTERED
```

Cisco Unified SRST

```
Router# show voice register pool telephone-number 1000
Pool ID              IP Address      Ln DN  Number                State
==== =============== =============== == === ==================== ============
1    9.13.18.40      9.13.18.40      1  1   1000                 REGISTERED
```

The following table contains descriptions of significant fields shown in this output, listed in alphabetical order.

Table 24: show voice register pool telephone number field descriptions

Field	Description
DN	Directory number of the phone.
ID	Phone identification (ID) address.

Field	Description
IP Address	IP address and port number of the phones
LN	Line number of the phone.
Number	Number of the phones.
Pool	Shows the current pool.
State	Registration state.

Related Commands

Command	Description
show voice register all	Displays all Cisco Unified SIP SRST and Cisco Unified CME configurations and register information.
show voice register pool	Displays all configuration information associated with a particular voice register pool.
show voice register pool detail all	Displays the details of all the pools defined in the system.

show voice register pool unregistered

To display the details of the voice registration pools that do not have any phones registered, use the **show voice register pool unregistered** command in privileged EXEC mode.

show voice register pool unregistered

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Version	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 display the details of the pools that do not have any active registrations. In Cisco Unified SRST, if multiple phones are trying to register through the same pool and if one phone successfully registers and the others do not, the pool is not considered as an unregistered pool, as it does have an active registration of the registered phone.

Examples

Cisco Unified CME and Cisco Unified SRST

The following is a sample output from this command displaying information for pools with no active registeration:

```
Router# show voice register pool unregistered
Pool Tag: 2
 MAC Address                : 0015.C68E.6D13
 No. of attempts to register: 0
 Unregister time            :
 Last register request time :
 Reason for state unregister:
         No registration request since last reboot/unregister
 Pool Tag: 3
 MAC Address                : 0021.5553.8998
 No. of attempts to register: 0
 Unregister time            :
 Last register request time :
 Reason for state unregister:
         No registration request since last reboot/unregister
 Pool Tag: 4
 MAC Address                : 8989.9867.8769
 No. of attempts to register: 0
 Unregister time            :
 Last register request time :
 Reason for state unregister:
         No registration request since last reboot/unregister
```

Related Commands

Command	Description
show voice register all	Displays all Cisco SIP SRST and Cisco CME configurations and register information.
show voice register pool	Displays all configuration information associated with a particular voice register pool.
show voice register pool registered	Displays details of phones that sucessfully register to Cisco Unified CME or Cisco Unified SRST.
voice register pool	Enters voice register pool configuration mode for SIP phones.

show voip sip-oauth key-server status

To display key retrieval details for SIP OAuth from CUCM.

show voip sip-oauth key-server status

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC (#)

Command History

Release	Modification
Cisco IOS XE Cupertino 17.8.1a	This command was introduced.

```
CUBE1-1B#show voip sip-oauth key-server status
Key-server:                  10.1.10.50
Last Request Time:           11:40:58.389 UTC Fri Nov 12 2021
Last Success response Time: 11:40:58.456 UTC Fri Nov 12 2021
Current Status:              SUCCESS
Next Request Time:           11:40:58.389 UTC Sat Nov 13 2021
Total requests sent:         13
Total success responses:     3
Total failure responses:     10
```

```
CUBE1-1B#show voip sip-oauth key-server status
Key-server:                  10.10.10.40
Last Request Time:           11:29:26.985 UTC Fri Nov 12 2021
Last Success response Time:
Current Status:              FAILURE (Timeout Error)
Next Request Time:           11:29:26.985 UTC Sat Nov 13 2021
Total requests sent:         8
Total success responses:     0
Total failure responses:     8
```

show voice register statistics

To display statistics associated with the registration event, use the **show voice register statistics** command in privileged EXEC mode.

show voice register statistics [{**global** | **pool tag**}]

Syntax Description

global	(Optional) Displays aggregate statistics associated with the SIP phone registration event.
pool tag	(Optional) Displays registration pool statistics associated with a specific pool tag. The maximum number of pools is version and platform dependent. Type **?** to display a list of values.

Command Modes

Privileged EXEC

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 CME.
15.1(2)T	Cisco CME 8.1 Cisco SIP SRST 8.1	This command was modified. The global and pool keywords and tag argument were added. The output display was also modified to show more information about pools in unregistered state and time-stamps of registration event.

Usage Guidelines

When using the **show voice register statistics** command, you can verify that the number of Registration and unRegister successes for global statistics are the sum of the values in the individual pools. Because some Registrations fail even before matching a voice register pool, for Registration and unRegister failed statistics the value is not the sum of the values in the individual pools. Immediate failures are accounted in the global statistics.

In Cisco Unified CME 8.1 and Cisco Unified SIP SRST 8.1, the time-stamps for the events is displayed along with other registration related statistics. The command output also displays the reason for pools in unregistered state. Use the show voice register statistics command with pool tag keyword to display registration pool statistics associated with a specific pool.

When using the global keyword, the show voice register command output displays the aggregate statistics associated with SIP phone registration. The output of this command also displays the attempted-registrations table.

Examples

Cisco Unified CME and Cisco Unified SRST

The following is a sample output from this command displaying all statistical information:

```
Router# show voice register statistics
Sample Output:
Global statistics
  Active registrations  : 2
  Total SIP phones registered: 2
  Total Registration Statistics
    Registration requests  : 3
    Registration success   : 2
    Registration failed    : 1
    unRegister requests    : 0
    unRegister success     : 0
    unRegister failed      : 0
    Attempts to register
    after last unregister        : 1
    Last Register Request Time   : *11:42:31.783 UTC Wed Sep 16 2009
    Last Unregister Request Time :
    Register Success Time        : *11:11:56.707 UTC Wed Sep 16 2009
    Unregister Success Time      :
Register pool 1 statistics
  Active registrations  : 1
  Total SIP phones registered: 1
  Total Registration Statistics
    Registration requests  : 1
    Registration success   : 1
    Registration failed    : 0
    unRegister requests    : 0
    unRegister success     : 0
    unRegister failed      : 0
    Attempts to register
    after last unregister        : 0
    Last Register Request Time   : *11:11:54.615 UTC Wed Sep 16 2009
    Last Unregister Request Time :
    Register Success Time        : *11:11:54.623 UTC Wed Sep 16 2009
    Unregister Success Time      :
Register pool 2 statistics
  Active registrations  : 1
  Total SIP phones registered: 1
  Total Registration Statistics
    Registration requests  : 1
    Registration success   : 1
    Registration failed    : 0
    unRegister requests    : 0
    unRegister success     : 0
    unRegister failed      : 0
    Attempts to register
    after last unregister        : 0
    Last Register Request Time   : *11:11:56.707 UTC Wed Sep 16 2009
    Last Unregister Request Time :
    Register Success Time        : *11:11:56.707 UTC Wed Sep 16 2009
    Unregister Success Time      :
```

Cisco Unified CME and Cisco Unified SRST

The following is a sample output from this command displaying all statistical information:

```
Router# show voice register statistics global
Global Statistics:
  Active registrations  : 1
  Total SIP phones registered: 2
  Total Registration Statistics
```

```
   R egistration requests  : 97715
   Registration success   : 3
   Registration failed    : 97712
   unRegister requests    : 1
   unRegister success     : 1
   unRegister failed      : 0
   Attempts to register
          after last unregister : 97712
   Last register request time   : *06:45:11.127 UTC Wed Oct 14 2009
   Last unregister request time : *11:56:22.179 UTC Tue Oct 13 2009
   Register success time        : *12:10:37.263 UTC Tue Oct 13 2009
   Unregister success time      : *11:56:22.182 UTC Tue Oct 13 2009
Phones that have attempted registrations and have failed:
MAC address: 001b.535c.d410
IP address : 8.3.3.111
Attempts   : 97712
Time of first attempt : *12:20:32.775 UTC Tue Oct 13 2009
Time of latest attempt: *06:46:14.815 UTC Wed Oct 14 2009
Reason for failure    :
        Unauthorized registration request
```

Cisco Unified CME and Cisco Unified SRST

The following is a sample output from this command displaying all statistical information associated with pool 1:

```
Router# show voice register statistics pool 1
Pool 1 Statistics:
  Active registrations   : 0
  Total SIP phones registered: 1
  Total Registration Statistics
    Registration requests  : 2
    Registration success   : 2
    Registration failed    : 0
    unRegister requests    : 1
    unRegister success     : 1
    unRegister failed      : 0
    Attempts to register
           after last unregister : 0
    Last register request time   : *12:10:37.259 UTC Tue Oct 13 2009
    Last unregister request time : *11:56:22.179 UTC Tue Oct 13 2009
    Register success time        : *12:10:37.263 UTC Tue Oct 13 2009
    Unregister success time      : *11:56:22.182 UTC Tue Oct 13 2009
  Reason for unregistered state:
         No registration request since last reboot/unregister
```

Table 25: show voice register statistics Field Descriptions, on page 168 describes the significant fields shown in this output.

Table 25: show voice register statistics Field Descriptions

Field	Description
Statistics:	Used with the **all**, **pool**, and **statistics** keywords. Shows the registration statistics for this pool.
Active registrations	Used with the **all**, **pool**, and **statistics** keywords. Shows the current active registrations.

Field	Description
Last Register Request Time	Used with all, pool, and statistics keywords. Shows details such as day, date, and time when the phones requested to register the last time.
Last unRegister Request Time	Used with all, pool, and statistics keywords. Shows details such as day, date, and time when the phones requested to unregister the last time.
Total Registration Statistics	Used with the **all**, **pool**, and **statistics** keywords. Shows the total registration statistics for this pool.
Registration requests	Used with the **all**, **pool**, and **statistics** keywords. Shows the incoming registration requests.
Registration success	Used with the **all**, **pool**, and **statistics** keywords. Shows the successful registrations.
Registration failed	Used with the **all**, **pool**, and **statistics** keywords. Shows the failed registrations.
unRegister requests	Used with the **all**, **pool**, and **statistics** keywords. Shows the incoming unregister/registration expire requests.
unRegister success	Used with the **all**, **pool**, and **statistics** keywords. Reports the number of successful unregisters.
unRegister failed	Used with the **all**, **pool**, and **statistics** keywords. Reports the number of failed unregisters.
Global statistics	Used with the **statistics** keyword. Details all active registrations.
Register pool *number* statistics	Used with the **statistics** keyword. Details specific pool statistics.

Related Commands

Command	Description
show voice register all	Displays all Cisco Unified SIP SRST and Cisco Unified CME configurations and register information.
show voice register pool	Displays all configuration information associated with a particular voice register pool.
show voice register pool attempted-registrations	Displays the details of phones that attempt to register with Cisco Unified CME or Cisco Unified SRST and fail.

srtp-crypto

To assign a previously configured crypto-suite selection preference list globally or to a voice class tenant, use the **srtp-crypto** command. To remove the crypto-suite selection preference and return to default preference list, use the **no** or **default** form of this command.

srtp-crypto *crypto-tag*
no srtp-crypto
default srtp-crypto

Syntax Description

crypto-tag	Unique number that is assigned to the voice class. The range is 1–10000. This number maps to the tag created using the **voice class srtp-crypto** command available in global configuration mode.

Command Default

No crypto-suite preference assigned.

Command Modes

voice class tenant configuration (config-class)

voice service voice sip configuration (conf-serv-sip)

Voice register pool (config-register-pool).

Command History

Release	Modification
Cisco IOS XE Everest 16.5.1b	This command was introduced.
Cisco IOS XE Cupertino 17.8.1a	Implemented **voice class srtp-crypto** *tag* under voice register pool.

Usage Guidelines

From Cisco IOS XE Cupertino 17.8.1a, **voice class srtp-crypto** *tag* is implemented under voice register pool mode. This command associates SRTP crypto ciphers with the pool.

Note Ensure that SRTP voice-class is created using the **voice class srtp-crypto** *crypto-tag* command before executing the **srtp-crypto** *crypto tag* command to apply the crypto-tag under global or tenant configuration mode.

You can assign only one crypto-tag. If you assign another crypto-tag, the last crypto-tag that is assigned replaces the previous crypto-tag.

Example

Example for assigning a crypto-suite preference to a voice class tenant:

```
Device> enable
Device# configure terminal
Device(config)# voice class tenant 100
Device(config-class)# srtp-crypto 102
```

Example for assigning a crypto-suite preference globally:

```
Device> enable
Device# configure terminal
Device(config)# voice service voice
Device(conf-voi-serv)# sip
Device(conf-serv-sip)# srtp-crypto 102
```

The following is an example of **voice class srtp-crypto** *tag* under voice register pool mode.

```
Router(config)#voice class srtp-crypto 22
Router(config-class)#crypto ?


Router(config-class)#crypto ?
        <1-4>   Set the preference order for the cipher-suite (1 = Highest)

Router(config-class)#crypto 1 ?
 AEAD_AES_128_GCM   Allow secure calls with SRTP AEAD_AES_128_GCM
 AEAD_AES_256_GCM   Allow secure calls with SRTP AEAD_AES_256_GCM
 AES_CM_128_HMAC_SHA1_32  Allow secure calls with SRTP AES_CM_128_HMAC_SHA1_32
 AES_CM_128_HMAC_SHA1_80  Allow secure calls with SRTP AES_CM_128_HMAC_SHA1_80
Router(config-class)#crypto 1 AEAD_AES_256_GCM

Router(config-class)#do show run | sec srtp-cry
  voice class srtp-crypto 22
  crypto 1 AEAD_AES_256_GCM

Router(config)# voice register pool 17
Router(config-register-pool)# id network 10.1.10.217 mask 255.255.255.255
Router(config-register-pool)# dtmf-relay rtp-nte
Router(config-register-pool)# codec g711ulaw
```

When you configure **srtp-crypto 23**, which is not present:

```
Router(config-register-pool)#voice-class srtp-crypto 23
ERROR: There is no voice-class srtp-crypto 23
```

When you configure **srtp-crypto 22**, which is present:

```
Router(config-register-pool)#voice-class srtp-crypto 22
```

Show run output for pool:

```
Router#show running-config | sec voice register pool 17
voice register pool 17
id network 10.1.10.217 mask 255.255.255.255
dtmf-relay rtp-nte
voice-class srtp-crypto 22
codec g711ulaw
Router#
```

Note **voice class srtp-crypto** must be configured before adding to a voice register pool.

Related Commands

Command	Description
voice class sip srtp-crypto	Enters voice class configuration mode and assigns an identification tag for a srtp-crypto voice class.
crypto	Specifies the preference for the SRTP cipher-suite that will be offered by Cisco Unified Border Element (CUBE) in the SDP in offer and answer.

Command	Description
show sip-ua calls	Displays active user agent client (UAC) and user agent server (UAS) information on Session Initiation Protocol (SIP) calls.
show sip-ua srtp	Displays Session Initiation Protocol (SIP) user-agent (UA) Secure Real-time Transport Protocol (SRTP) information.

subnet

To define which IP phones are part of an emergency response location (ERL) for the enhanced 911 service, use the **subnet** command in voice emergency response location configuration mode. To remove the subnet definition, use the **no** form of this command.

subnet [{**1** | **2**}] *IPaddress* *mask*
no **subnet** [{**1** | **2**}]

Syntax Description

IPaddress	IP address that identifies which IP phones are part of the ERL.
mask	IP subnet mask for the network segment that is part of the ERL.

Command Default

No subnets are defined.

Command Modes

V
oice 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 to 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 the groups of IP phones that are part of an ERL. You can create up to 2 different subnets. To include all phones on a single ERL, you can set the subnet mask to 0.0.0.0 to indicate a “catch-all” subnet.

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
elin	Specifies a PSTN number that will replace the caller’s extension.

system message (call-manager-fallback)

To customize the system message text displayed on all Cisco IP phones units in fallback mode that are connected to a Cisco Unified Survivable Remote Site Telephony (SRST) router, use the **system message** command in call-manager-fallback configuration mode. To disable the customized message and return to the default system message, use the **no** form of this command.

system message {**primary** *primary-string* | **secondary** *secondary-string*}
no system message {**primary** *primary-string* | **secondary** *secondary-string*}

Syntax Description

primary	Sets the system message for Cisco IP phones that can support static text messages during fallback, such as the Cisco IP Phone 7940 and the Cisco IP Phone 7960.
primary-string	Text string of less than 32 characters.
secondary	Sets the system message for Cisco IP phones that do not support static text messages and that have a limited display space, such as the Cisco IP Phone 7910.
secondary-string	Text string of less than 20 characters.

Command Default

The default fallback display message for Cisco IP phones that support static text messages is “CM Fallback Service Operating.” For Cisco IP phones that do not support static text messages, the default message is “CM Fallback Service.”

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

Changes to the display message configuration occur only after a phone reset, at the end of each call, or on receipt of the next keepalive message from an idle phone.

The normal in-service static text message is controlled by Cisco Unified Communications Manager.

Secondary IP phones flash system messages during fallback.

Examples

The following example sets the system message to “Customized Message” for all Cisco IP Phone 7940 and Cisco IP Phone 7960 units connected to a Cisco Unified SRST router:

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# system
message primary Customized Message
```

Related Commands

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

system message (voice register global)

To define a message that displays on SIP phones in a Cisco Unified Survivable Remote Site Telephony (Cisco Unified SRST) system, use the **system message** command in voice register global configuration mode. To return to the default, use the **no** form of this command.

system message *string*
no system message

Syntax Description

string	Message that displays on SIP phones after the phones failover to Cisco Unified SRST. String can contain a maximum of 32 alphanumeric characters.

Command Default

"CM Fallback Service Operating" message from dictionary file displays.

Command Modes

Voice register global configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XJ	Cisco Unified SRST 4.1	This command was introduced.
12.4(15)T	Cisco Unified CME 4.1	This command was integrated into Cisco IOS Release 12.4(15)T.
Cisco IOS XE Amsterdam 17.2.1r	Cisco Unified SIP SRST 12.8	Introduced support for YANG models.

Usage Guidelines

The command allows you to customize the idle prompt message that displays on the status line of SIP phones after the phones lose connection with Cisco Unified Communications Manager and failover to Cisco Unified SRST. The default message that displays is from the dictionary file for the phone. The configured message displays until the phones fallback to Cisco Unified Communications Manager. For versions earlier than Cisco Unified SRST 4.1, the phones display the default message from the dictionary file.

This command is not supported on the Cisco Unified IP Phone 7905, 7912, 7940, or 7960.

Examples

The following example shows that SIP phones will display the message, "SRST service active" after the phones register to Cisco Unified SRST.

```
Router(config)# voice register global
Router(config-register-global)# system message SRST service active
```

Related Commands

Command	Description
show voice register global	Displays all global configuration parameters associated with SIP phones.

time-format (call-manager-fallback)

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

time-format {**12** | **24**}
no **time-format** {**12** | **24**}

Syntax Description

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

Command Default

12-hour format

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 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.

Examples

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

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

Related Commands

Command	Description
call-manager-fallback	Enables Cisco Unified Survivable Remote Site Telephony (SRST) support and enters call-manager-fallback configuration mode.

timeouts busy (call-manager-fallback)

To set the timeout value for call transfers to busy destinations, use the **timeouts busy** command in call-manager-fallback configuration mode. To return to the default value, use the **no** form of this command.

timeouts busy *seconds*
no timeouts busy

Syntax Description

seconds	Number of seconds after connection to a busy destination before a transferred call is disconnected. Range is from 0 to 30 seconds. Default is 10.

Command Default

10 seconds

Command Modes

Call-manager-fallback configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(8)T	Cisco SRST 2.0	This command was introduced.

Usage Guidelines

For calls that are transferred to busy destinations, this command sets the amount of time after connection to the busy destination before the call is disconnected.

Note that the timeout set by this command applies only to calls that are transferred to busy destinations and not to calls that directly dial busy destinations.

Examples

The following example sets the ringing timeout to 10 seconds:

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)#
timeouts busy 20
```

Related Commands

Command	Description
call-manager-fallback	Enables Cisco Unified Survivable Remote Site Telephony (SRST) and enters **call-manager-fallback** configuration mode.

timeouts interdigit (call-manager-fallback)

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

timeouts interdigit *seconds*
no timeouts interdigit

Syntax Description

seconds	Interdigit timeout duration, in seconds, for all Cisco IP phones. Valid entries are integers from 2 to 120.

Command Default

No default behavior or values.

Command Modes

Call-manager-fallback configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(2)XB	Cisco SRST 1.0	This command was introduced on the following platforms: Cisco 2600 series and Cisco 3600 series multiservice routers; Cisco IAD2420 series IADs; Cisco 7200 series routers.
12.2(2)XT	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 **timeouts interdigit** command specifies how long, in seconds, the system waits after a caller enters the initial digit or a subsequent digit of the dialed string. The interdigit timer is activated when the caller enters a digit and is restarted each time the caller enters subsequent digits until the destination address is identified. If the configured timeout value is exceeded before the destination address is identified, a tone sounds and the call is terminated.

Examples

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

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

In this example, the 5 seconds refers to the elapsed time after which an incompletely dialed number times out. For example, if you dial nine digits (408555010) instead of the required ten digits (4085550100), you hear a busy tone after the 5 “timeout” seconds have elapsed.

Related Commands

Command	Description
call-manager-fallback	Enables Cisco Unified Survivable Remote Site Telephony (SRST) support and enters call-manager-fallback configuration mode.
timeouts interdigit (voice port)	Configures the interdigit timeout value for a specified voice port.

timeouts ringing (call-manager-fallback)

To set the time before a disconnect code is returned on phones without a call-forward no-answer configuration, use the **timeouts ringing** command in call-manager-fallback configuration mode. To disable the time setting, use the **no** form of this command.

timeouts ringing *seconds*
no timeouts ringing

Syntax Description

seconds	The duration, in seconds, for which a voice port allows ringing to continue if a call is not answered. The range is from 5 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(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

This mechanism protects against hung inbound calls through interfaces that do not have forward disconnect supervision, such as Foreign Exchange Office (FXO).

Expiration of the timeout causes incoming calls to return a disconnect code to the caller.

Examples

The following example sets the ringing timeout to 10 seconds:

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# timeouts ringing 10
```

Related Commands

Command	Description
call-manager-fallback	Enables Cisco Unified Survivable Remote Site Telephony (SRST) and enters call-manager-fallback configuration mode.

transfer max-length (voice register pool)

To specify the maximum length of the transfer number, use the **transfer max-length** command in voice register pool or voice register template configuration mode. To disable the maximum length, use the **no** form of this command.

transfer max-length *max-length*
no transfer max-length *max-length*

Syntax Description

max-length	Maximum length of the transfer number. Range is 3 to 16.

Command Default

No maximum length is specified for the transfer number.

Command Modes

Voice register pool configuration (config-register-pool)

Voice register template configuration ((config-register-temp)

Command History

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

Usage Guidelines

The **transfer max-length** command is used to indicate the maximum length of the number being dialed for a call transfer. When only a specific number of digits are to be allowed during a call transfer, a value between 3 and 16 is configured.When the number dialed exceeds the maximum length configured, then the call transfer is blocked.

Examples

The following example shows how to configure the maximum length of the transfer number under voice register pool 1. Because the maximum length is configured as 5, only call transfers to Cisco Unified SIP IP phones with a five-digit directory number are allowed. All call transfers to directory numbers with more than five digits are blocked.

```
Router# configure terminal
Router(config)# voice register pool 1
Router(config-register-pool)# transfer max-length 5
```

The following example shows how to configure the maximum length of the transfer number for a set of phones under voice register template 2:

```
Router# configure terminal
Router(config)# voice register template 2
Router(config-register-temp)# transfer max-length 10
```

Command	Description
voice register pool	Enters voice register pool configuration mode and creates a pool configuration for a SIP IP phone in Cisco Unified CME or for a set of SIP phones in Cisco Unified SIP SRST

Command	Description
voice register template	Enters voice register template configuration mode and defines a template of common parameters for SIP phones.

transfer-digit-collect

To select the digit-collection method for consultative call-transfers, use the **transfer-digit-collect** command in telephony-service configuration mode for Cisco Unified CME or in call-manager-fallback configuration mode for Cisco Unified SRST. To reset to the default value, use the **no** form of this command.

transfer-digit-collect {**new-call** | **orig-call**}
no transfer-digit-collect

Syntax Description

new-call	Dialed digits are collected from new call leg. Default value.
orig-call	Dialed digits are collected from original call leg.

Command Default

Digits are collected from the new consultative call-leg (**new-call** keyword).

Command Modes

Telephony-service configuration (config-telephony)

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

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 whether the dialed digits of the target number are collected on the original call leg or on the new call leg that is created when a phone user initiates a consultative call-transfer.

For consultative transfers, a local number is matched on the **number** command in ephone-dn configuration mode; a PSTN number is matched on the **transfer-pattern** command in telephony service mode.

The **orig-call** keyword selects the method used in versions before Cisco Unified CME 4.3 and Cisco Unified SRST 4.3. After a phone user presses the Transfer soft key, the dialed digits of the target number are collected on the original call leg and buffered until either a local ephone-dn or transfer-pattern is matched. When the transfer-to number is matched, the original call is put on hold and an idle line or channel is seized to send the dialed digits from the buffer.

The **new-call** keyword selects the default method that is used in Cisco Unified CME 4.3 and later versions and Cisco Unified SRST 4.3 and later versions. The transfer-to digits are collected on a new consultative call-leg that is created when the user presses the Transfer soft key. The consultative call-leg is seized and the dialed digits are passed on without being buffered until the digits are matched and the consultative call-leg moves to the alerting state.

The **new-call** keyword is supported only if:

- The **transfer-system full-consult** command (default) is set in telephony-service configuration mode.
- The **transfer-mode consult** command (default) is set for transferor's directory number (ephone-dn).

- An idle line or channel is available for seizing, digit collection, and dialing.

A consultative transfer is one in which the transferring party either connects the caller to a ringing phone (ringback heard) or speaks with the third party before connecting the caller to the third party.

Examples

The following example shows the digit-collection set to the method used in versions before Cisco Unified CME 4.3 and Cisco Unified SRST 4.3:

```
Router(config)# telephony-service
Router(config-telephony)# transfer-digit-collect orig-call
```

Related Commands

Command	Description
transfer-mode	Specifies the type of call transfer for an individual directory number that uses the ITU-T H.450.2 standard.
transfer-pattern (telephony-service)	Allows the transfer of calls to phones outside the local Cisco Unified CME network.
transfer-system	Specifies the call transfer method for all IP phones on a Cisco Unified CME router using the ITU-T H.450.2 standard.

transfer-pattern

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

transfer-pattern *transfer-pattern*
no transfer-pattern

Syntax Description

transfer-pattern	String of digits for permitted call transfers. Wildcards are allowed.

Command Default

This feature is enabled.

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

When you define transfers to nonlocal numbers, it is important to note that transfer-pattern digit matching is performed before translation-rule operations. Therefore, you should specify in this command the digits that are actually entered by phone users before they are translated. For more information, see Enabling Digit Translation Rules section in the Cisco IOS Survivable Remote Site Telephony Version 3.3 System Administrator Guide.

Examples

The following example sets a transfer pattern:

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# transfer-pattern 55501..
```

A maximum of 32 transfer patterns can be entered. In this example, 55501.. (the two decimal points are used here as wildcards) permits transfers to any numbers in the range from 555-0100 to 555-0199.

Related Commands

Command	Description
call-manager-fallback	Enables Cisco Unified Survivable Remote Site Telephony (SRST) support and enters call-manager-fallback configuration mode.

transfer-pattern blocked (voice register pool)

To block all call transfers for a specific Cisco Unified SIP IP phone or a set of Cisco Unified SIP IP phone, use the **transfer-pattern blocked** command in voice register pool and voice register template configuration mode. To allow call transfers, use the **no** form of this command.

transfer-pattern blocked
no transfer-pattern blocked

Syntax Description

This command has no arguments or keywords.

Command Default

Call transfers for a specific Cisco Unified SIP IP phone or a set of Cisco Unified SIP IP phone are allowed.

Command Modes

Voice register pool configuration (config-register-pool)

Voice register template configuration ((config-register-temp)

Command History

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

Usage Guidelines

When the **transfer-pattern blocked** command is configured for a specific phone, no call transfers are allowed from that phone over the trunk.

This feature forces unconditional blocking of all call transfers from a specific phone to any other non-local numbers (external calls from one trunk to another trunk). No call transfers from this specific phone are possible even when a transfer pattern matches the dialed digits for transfer.

Examples

The following example shows how to block all call transfers for voice register pool 5:

```
Router(config)# voice register pool 5
Router(config-register-pool)# transfer-pattern ?
  blocked  global transfer pattern not allowed
Router(config-register-pool)# transfer-pattern blocked
```

The following example shows how to block all call transfers for a set of Cisco Unified SIP IP phones defined by voice register template 9:

```
Router(config)# voice register template 9
Router(config-register-temp)# transfer-pattern ?
  blocked  global transfer pattern not allowed
Router(config-register-temp)# transfer-pattern blocked
```

Related Commands

Command	Description
voice register pool	Enters voice register pool configuration mode and creates a pool configuration for a Cisco Unified SIP IP phone in Cisco Unified CME or for a set of Cisco Unified SIP IP phones in Cisco Unified SIP SRST.

Command	Description
voice register template	Enters voice register template configuration mode and defines a template of common parameters for Cisco Unified SIP IP phones.

transfer-system (call-manager-fallback)

To specify the call-transfer method for all IP phones on a Cisco Unified Survivable Remote Site Telephony (SRST) router using the ITU-T H.450.2 standard, use the **transfer-system** command in call-manager-fallback configuration mode. To disable the call-transfer method, use the **no** form of this command.

transfer-system {**blind** | **full-blind** | **full-consult** | **local-consult**}
no transfer-system

Syntax Description

blind	Transfers calls without consultation using a single phone line and the Cisco proprietary method. The keyword **blind** is not recommended. Use either the **full-blind** or **full-consult** keyword instead.
full-blind	Transfers calls without consultation using H.450.2 standard methods.
full-consult	Transfers calls using H.450.2 with consultation using the second phone line if available, or the calls fall back to full-blind if the second line is unavailable.
local-consult	Transfers calls with local consultation using the second phone line if available, or the calls fall back to blind for nonlocal consultation or transfer target. This method is intended for use primarily in VoFR networks because the Cisco VoFR call-transfer protocol does not support an end-to-end transfer with consultation mechanism.

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

Call transfers using the H.450.2 standard can be blind or consultative. A blind transfer is one in which the transferring phone connects the caller to a destination line before ringback begins. A consultative transfer is one in which the transferring party either connects the caller to a ringing phone (ringback heard) or speaks with the third party before connecting the caller to the third party. When H.450.2 call transfer is selected using the **full-blind** or **full-consult** keyword, the router must be configured with a Tool Command Language (Tcl) script that supports the H.450.3 protocol. The Tcl script is loaded on the Cisco Unified SRST router with the **call application voice** command.

Note Note: The keyword **blind** is not recommended. Use either the **full-blind** or **full-consult** keyword instead.

Examples

The following example sets full consultation as the call-transfer method for this Cisco Unified SRST phone network:

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# transfer-system full-consult
```

Related Commands

Command	Description
call application voice	Defines an application, indicates the location of the corresponding Tcl files that implement the application, and loads the selected Tcl script.
call-manager-fallback	Enables Cisco Unified SRST support and enters call-manager-fallback configuration mode.

translate (call-manager-fallback)

To apply a translation rule to modify the phone number dialed or received by any Cisco IP phone user during Cisco Unified Communications Manager fallback, use the **translate** command in call-manager-fallback configuration mode. To disable this feature, use the **no** form of this command.

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

Syntax Description

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

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, 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.

Usage Guidelines

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

Examples

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

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

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# translate called 20
```

Related Commands

Command	Description
call-manager-fallback	Enables Cisco Unified Survivable Remote Site Telephony (SRST) support and enters call-manager-fallback configuration mode.
translation-profile (call-manager-fallback)	Assigns a translation profile for incoming or outgoing call legs on a Cisco IP phone.
translation-rule	Creates a translation name and enters translation-rule configuration mode.

translate-outgoing (voice register pool)

To allow an explicit setting of translation rules on the VoIP dial peer in order to modify a phone number dialed by any Cisco IP phone user, use the **translate-outgoing** command in voice register pool configuration mode. To disable translation rules, use the **no** form of this command.

translate-outgoing {**called** | **calling**} *rule-tag*
no translate-outgoing {**called** | **calling**}

Syntax Description

called	Called party requires translation.
calling	Calling party requires translation.
rule-tag	The rule-tag is an arbitrarily chosen number by which the rule set is referenced. The range is from 1 to 2147483.

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 CME.

Usage Guidelines

Translation rules are a powerful general-purpose number-manipulation mechanism that perform operations such as automatically adding telephone area and prefix codes to dialed numbers. The translation rules are applied to VoIP dial peers created by Cisco Unified Session Initiation Protocol (SIP) Survivable Remote Site Telephony (SRST) or Cisco Unified Communications Manager Express (Cisco Unified CME)

.

During registration, a dial peer is created, and that dial peer includes a default translation rule. The **translate-outgoing** command allows you to change the translation rule, if desired. The **translate-outgoing** command allows you to select a preconfigured number translation rule to modify the number dialed by a specific extension.

Translation rules must be set by using the **translate-outgoing** command before the **alias** command is configured in Cisco Unified SIP SRST.

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

Examples

Cisco Unified CME

The following is partial sample output from the **show running-config** command showing that called-party 1 requires translation.

```
voice register pool 1
 id mac 0030.94C2.A22A
 preference 5
 cor incoming call91 1 91011
 translate-outgoing called 1
```

Cisco Unified SIP SRST

The following is partial sample output from the **show running-config** command showing that called-party 1 requires translation.

```
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
```

Related Commands

Command	Description
alias (voice register pool)	Allows Cisco SIP IP phones to handle inbound PSTN calls to telephone numbers that are unavailable when the main proxy is not available.
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.
translate-outgoing (dial-peer)	Applies a translation rule to manipulate dialed digits on an outbound POTS or VoIP call leg.
voice register pool	Enters voice register pool configuration mode for SIP phones.

translation-profile (call-manager-fallback)

To assign a translation profile for incoming or outgoing call legs on a Cisco IP phone, use the **translation-profile** command in call-manager-fallback configuration mode. To delete the translation profile from the voice port, use the **no** form of this command.

translation-profile {**incoming** | **outgoing**} *name*
no translation-profile {**incoming** | **outgoing**} *name*

Syntax Description

incoming	Specifies that this translation profile handles incoming calls.
outgoing	Specifies that this translation profile handles outgoing calls.
name	Name of the translation profile.

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 3.2	This command was introduced.

Usage Guidelines

Cisco Unified Survivable Remote Site Telephony (SRST) 3.2 and later versions support translation profiles. Translation profiles allow you to group translation rules together and to associate translation rules with the following:

- Called numbers
- Calling numbers
- Redirected called numbers

Use the **translation-profile** command to assign a global predefined translation profile to an incoming or outgoing call leg. For example, a company number can be assigned to overwrite an individual caller's phone number. That is, the **translation-profile** command modifies the phone number dialed or received by a Cisco IP phone user while in Communications Manager fallback mode.

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

Examples

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

```
voice translation-profile name1
 translation calling rule1
 translation called-direct rule2
call-manager-fallback
 translation-profile incoming name1
```

Related Commands

Command	Description
call-manager-fallback	Enables Cisco SRST support and enters call-manager-fallback configuration mode.
show voice translation-profile	Displays the configuration of a translation profile.
translate (call-manager- fallback)	Applies a translation rule to modify the phone number dialed or received by any Cisco IP phone user during Communications Manager fallback.
translation-rule	Creates a translation name and enters translation-rule configuration mode to apply rules to the translation name.
voice translation-profile	Defines a translation profile for voice calls.

translation-profile (voice register)

To apply a translation profile to incoming or outgoing call legs on a SIP phone in a Cisco Unified SRST system, use the **translation-profile** command in voice register dn or voice register pool configuration mode. To remove the translation profile, use the **no** form of this command.

translation-profile {**incoming** | **outgoing**} *name*
no translation-profile {**incoming** | **outgoing**}

Syntax Description

incoming	Specifies that this translation profile handles incoming calls.
outgoing	Specifies that this translation profile handles outgoing calls.
name	Name of the translation profile.

Command Default

Translation profile is not assigned to call legs on the phone.

Command Modes

Voice register dn configuration (config-register-dn)
Voice register pool configuration (config-register-pool)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(22)YB	Cisco Unified SIP SRST 7.1	This command was introduced.
12.4(24)T	Cisco Unified SIP SRST 7.1	This command was integrated into Cisco IOS Release 12.4(24)T.
Cisco IOS XE Dublin 17.10.1a	Unified SRST 14.3	Introduced support for YANG models.

Usage Guidelines

This command assigns a predefined translation profile to incoming or outgoing call legs to and from the Cisco Unified SRST router. Use this command to apply the translation profile to a specific directory number or to all directory numbers on a SIP phone. The translation profile that you assign is created by using the **voice translation-profile** command.

Examples

The following example assigns the translation profile named “profile1” to handle translation of outgoing calls from SIP phone 21:

```
Router(config)# voice register pool 21
Router(config-register-pool)# translation-profile outgoing profile1
```

The following example assigns the translation profile named “profile2” to handle translation of incoming calls to extension 1200:

```
Router(config)# voice register dn 12
Router(config-register-pool)# number 1200
Router(config-register-pool)# translation-profile incoming profile2
```

Related Commands

Command	Description
show voice translation-profile	Displays the configuration of a translation profile.
translate (translation profiles)	Assigns a translation rule to a translation profile.
voice translation-profile	Defines a translation profile for voice calls.

transport-tcp-tls (call-manager-fallback)

To configure a specific TLS version for Unified Secure SCCP SRST, use the **transport-tcp-tls** command in call-manager-fallback configuration mode. To enable the default command configuration, use the **no** form of this command.

To set default value which supports all TLS versions, use the **default** form of this command.

transport-tcp-tls { **v1.0** | **v1.1** | **v1.2** [**sha2**] | **v1.3** [**sha2**] }
no transport-tcp-tls { **v1.0** | **v1.1** | **v1.2** [**sha2**] | **v1.3** [**sha2**] }
default transport-tcp-tls

Syntax Description

v1.0	Enables TLS version 1.0.
v1.1	Enables TLS version 1.1.
v1.2	Enables TLS version 1.2.
v1.3	Enables TLS version 1.3.
sha2	Enables SHA2 ciphers with TLS version 1.2 or 1.3.

Command Default

In the default form, all the TLS versions **except TLS 1.0** are supported for this CLI command.

Command Modes

call-manager-fallback Configuration (config-cm-fallback)

Command History

Cisco IOS Release	Cisco Product	Modification
Cisco IOS XE Fuji 16.9.1	Unified SRST 12.3	This command was introduced.
Cisco IOS XE Cupertino 17.8.1a	Unified SRST 14.2	This command is enhanced to limit TLS1.2 to using SHA2 ciphers only.
Cisco IOS XE 17.14.1a	Unified SRST 14.4	This command is enhanced to support TLS version 1.3 ciphers. In addition, SHA2 cipher support for TLS version 1.3 is introduced. Introduced support for YANG model.

Usage Guidelines

Use the **transport-tcp-tls** command to define the version of transport layer security for the Secure SCCP Unified SRST.

Starting from Cisco IOS XE 17.14.1a, TLS version 1.3 is supported in addition to TLS versions 1.0, 1.1 and 1.2. It is recommended that TLS version 1.2 or 1.3 is used wherever possible to ensure security or compliance. To configure exclusivity, use the **transport-tcp-tls** *version* command.

From Unified SRST 12.3 and later releases, TLS versions 1.1 and 1.2 are supported for Analog Voice Gateways on Unified SRST. SCCP phones only support TLS version 1.0.

Starting from Unified SCCP SRST 14.4 release, TLS version 1.3 is supported for Cisco VG400, VG410, VG420, and VG450 Analog Voice Gateways.

When **transport-tcp-tls** is configured without specifying a version, the default behavior of the CLI command is enabled. In the default form, all the TLS versions (1.3, 1.2, and 1.1) are supported.

For Secure SIP and Secure SCCP endpoints that do not support TLS version 1.2, you need to configure TLS 1.0 for the endpoints to register to Unified Secure SRST 12.3 (Cisco IOS XE Fuji Release 16.9.1). This also means that endpoints which support 1.2 will also use the 1.0 suites.

For TLS 1.0 support on Cisco IOS XE Fuji Release 16.9.1 for SCCP endpoints, you need to specifically configure:

- transport-tcp-tls v1.0 under call-manager-fallback configuration mode

From Cisco IOS XE Cupertino 17.8.1a onwards, the **transport-tcp-tls v1.2** command is enhanced to allow only SHA2 ciphers by using the additional "sha2" keyword.

From Cisco IOS XE 17.14.1a, the **transport-tcp-tls v1.3** command supports SHA2 ciphers by using the additional "sha2" keyword. If SHA2 ciphers are configured, media packets are encrypted and sent using the AEAD_AES_256_GCM SRTP cipher suite.

Examples

The following example shows how to specify a TLS version for a secure SCCP phone using the **transport-tcp-tls** command:

```
Device(config)# call-manager-fallback
Device(config-cm-fallback)#transport-tcp-tls ?
  v1.0  Enable TLS Version 1.0
  v1.1  Enable TLS Version 1.1
  v1.2  Enable TLS Version 1.2
  v1.3  Enable TLS Version 1.3



Device(config-cm-fallback)#transport-tcp-tls v1.2 ?
  sha2  Allow SHA2 ciphers only
  <cr>  <cr>
Device(config-cm-fallback)#transport-tcp-tls v1.2 sha2 ?
  <cr>  <cr>



Device(config-cm-fallback)#transport-tcp-tls v1.3 ?
  sha2  Allow SHA2 ciphers only
  <cr>  <cr>
Device(config-cm-fallback)#transport-tcp-tls v1.3 sha2 ?
  <cr>  <cr>
```

Related Commands

Command	Description
transport (voice-register-pool)	Defines the default transport type supported by a new phone.

trustpoint (credentials)

To specify the name of the trustpoint to be associated with a Cisco Unified Communications Manager Express (Cisco Unified CME) CTL provider certificate or with the Cisco Unified Survivable Remote Site Telephony (SRST) router certificate, use the **trustpoint** command in credentials configuration mode. To change the specified trustpoint, use the **no** form of this command.

trustpoint *trustpoint-name*
no trustpoint

Syntax Description

trustpoint-name	Name of the trustpoint to be associated with the Cisco Unified CME CTL provider certificate or the Cisco Unified SRST device certificate.

Command Default

No default behavior or values.

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.
12.3(14)T	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 define the trustpoint for the CTL provider. This trustpoint will be used for TLS sessions with the CTL client.

Cisco Unified SRST

The name of the trustpoint must be consistent with the trustpoint name of the Cisco Unified SRST router.

Examples

Cisco Unified CME

The following example sets up a CTL provider on the Cisco Unified CME router with the IP address 172.19.245.1.

```
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, sets the IP source address and port, and specifies 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
credentials	Provides the Cisco Unified CME CTL provider certificate or the Cisco Unified SRST router certificate and enters credentials configuration mode.
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.
ip source-address (credentials)	Enables the router to receive messages through the specified IP address and port.
show credentials	Displays the credentials settings.

user-locale (call-manager-fallback)

To set the language by country for displays on the Cisco Unified IP Phone 7905G, Cisco Unified IP Phone 7912G, Cisco UnifiedIP Phone 7940G and Cisco UnifiedIP Phone 7960G, use the **user-locale** command in call-manager-fallback configuration mode. To disable the country selection and use the default country (United States), use the **no** form of this command.

user-locale *country-code*
no user-locale *country-code*

Syntax Description

country-code	The following ISO-3166 codes are available to Cisco Unified Survivable Remote Site Telephony (SRST) systems running under Cisco Unified Communications Manager V3.2 or later: • **DE**—German. • **DK**—Danish. • **ES**—Spanish. • **FR**—French. • **IT**—Italian. • **JP**—Japanese Katakana (available under Cisco Unified Communications Manager V4.0 or later). • **NL**—Dutch. • **NO**—Norwegian. • **PT**—Portuguese. • **RU**—Russian. • **SE**—Swedish. • **US**—United States English (default).

Command Default

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

Command Modes

Call-manager-fallback configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(11)YT	Cisco SRST 2.1	This command was introduced for use on the Cisco IP Phone 7940 and Cisco IP Phone 7960 for Cisco SRST systems running Cisco Communications Manager V3.2. Support includes country codes for France, Germany, Italy, Portugal, Spain, and the United States (default).
12.3(4)T	Cisco SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.3(7)XL	Cisco SRST 3.1.1	The **JP** keyword was added, providing support for Japanese Katakana.
12.3(11)T	Cisco SRST 3.2	This command was integrated into Cisco IOS Release 12.3(11)T and support was increased to include the Cisco Unified IP Phone 7905G and Cisco Unified IP Phone 7912G.

Usage Guidelines

Japanese Katakana is now supported with the **JP** keyword and is available to Cisco Unified SRST systems running under Cisco Unified Communications Manager V4.0. All other *country-code* options are available to Cisco Unified SRST systems running under Cisco Unified Communications Manager V3.2. Systems running Cisco Unified Communications Manager prior to V3.2 can use only the default country, United States (**US**).

Examples

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

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

Related Commands

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

url-button

To configure service url feature button on a line key, use the url-button command in ephone-template mode. To disable the service url feature button configuration on a line key, use the **no** form of this command.

{**url-button index type** | **url** [**name**]}
{**no url-button index type** | **url** [**name**]}

Syntax Description

index	Unique index number. Range: 1 to 8.
type	Type of service url button. Following types of url service buttons are available: • myphoneapp: My phone application configured under phone user interface. • em: Extension Mobility • snr: Single Number Reach • voicehuntgroups: Voice Hunt Groups • park-list: Displays list of parked calls
url name	Service url with maximum length of 31 characters.

Command Default

By default, URL-button configuration on a line key is disabled.

Command Modes

Ephone template configuration (config-ephone-template)

Command History

Cisco IOS Release	Cisco Product	Modification
15.1(3)T	Cisco Unified Enhanced SRST 8.5	This command was introduced.
15.4(3)M	Cisco Unified Enhanced SRST 10.5	This command was modified.

Usage Guidelines

Use this command to configure a url-button on a phone's line key.You can configure a line key to function as one of the following url-button types: extension mobility (EM), My Phone Apps, or single number reach (SNR). You can also configure a line button to function as a service url by configuring a url name of a maximum length of 31 characters.

Examples

The following examples shows three url buttons configured as line keys:

```
!
telephony-service
 max-ephones 25
 max-conferences 12 gain -6
 transfer-system full-consult
!
!
ephone-template  5
 url-button 1 em
 url-button 2 mphoneapp
 url-button 3 snr
!
ephone-template 6
conference drop-mode never
```

```
conference add-mode all
conference admin: No
max-calls-per-button 8
busy-trigger-per-button 0
privacy default
url-button 1 em
url-button 2 www.cisco.com www.cisco.com
url-button 3 snr
url-button 4 help help
url-button 7 myphoneapp
!
!
```

Related Commands

Command	Description
show telephony-service ephone-template	Displays the contents of all the ephone-templates defined.

utf8

To define Unicode UTF-8 support for a phone type, use the **utf8** command in ephone-type configuration mode. To reset to the default value, use the **no** form of this command.

utf8
no utf8

Syntax Description

This command has no arguments or keywords.

Command Default

Phone type supports Unicode UTF-8.

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 whether Unicode UTF-8 is supported by the type of phone that is being added with the phone-type template.

Examples

The following example shows that UTF-8 support is set to disabled for the Nokia E61 when creating the ephone-type template:

```
Router(config)# ephone-type E61
Router(config-ephone-type)# no utf8
```

Related Commands

Command	Description
device-id	Specifies the device ID for a phone type.
type	Assigns the phone type to an SCCP phone.

vad (voice register pool)

To enable voice activity detection (VAD) on a VoIP dial peer, use the **vad** command in voice register pool configuration mode. To disable VAD, use the **no** form of this command.

vad
no vad

Syntax Description

This command has no arguments or keywords.

Command Default

Enabled

Command Modes

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

VAD detects periods of silence in the voice signal and temporarily discontinues transmission of the signal during these periods to save bandwidth. Because VAD is enabled by default, there is no comfort noise during periods of silence. As a result, the call may seem to be disconnected and you may prefer to set **no vad** on the SIP phone pool.

Examples

The following example shows how to disable VAD for pool 1:

```
Router(config)# voice register pool 1
Router(config-register-pool)# no vad
```

Related Commands

Command	Description
voice register pool	Enters voice register pool configuration mode for SIP phones.

video (call-manager-fallback)

To enter video configuration mode for a Cisco Unified SRST router, use the **video** command in call-manager-fallback configuration mode. To reset global video parameters, use the **no** form of this command.

video
no video

Syntax Description

This command has no arguments or keywords.

Command Default

Global video parameters are configured.

Command Modes

Call-manager-fallback 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 the **video** command to enter video configuration mode and set video parameters for all applicable Cisco IP phones in a Cisco Unified SRST system.

Examples

The following example shows how to enter video configuration mode for a Cisco Unified SRST system. You must enter video configuration mode to set video parameters, such as maximum bit rate.

```
Router(config)#
call-manager-fallback
Router(config-call-manager-fallback)# video
Router(conf-cm-fallback-video)# maximum bit-rate 256
```

Related Commands

Command	Description
show call active video	Displays call information for SCCP video calls in progress.
show call history video	Displays call history information for SCCP video calls.
video (telephony-service)	Enters video configuration mode to set video parameters for all applicable Cisco IP phones associated with a Cisco Unified CME router.

vm-integration

To enter voice-mail integration configuration mode and enable voice-mail integration with dual tone multifrequency (DTMF) and analog voice-mail systems, use the **vm-integration** command in global configuration mode. To disable voice-mail integration, use the **no** form of this command.

vm-integration
no vm-integration

Syntax Description

This command has no arguments or keywords.

Command Default

No voice-mail integration is defined.

Command Modes

Global configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(11)YT	Cisco SRST 2.1	This command was introduced for Cisco Survivable Remote Site Telephony (SRST).
12.2(2)XT	Cisco CME 2.0	This command was introduced on the following platforms: Cisco 1750, Cisco 1751, Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(8)T	Cisco CME 2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	Cisco CME 2.0	This command was implemented on the Cisco 2600XM and Cisco 2691.
12.2(11)T	Cisco CME 2.01	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760.

Usage Guidelines

The **vm-integration** command is used to enter voice-mail integration configuration mode. Use voice-mail integration configuration mode to integrate a Cisco Unified Communications Manager Express (Cisco Unified CME) system with an analog voice-mail system.

Examples

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

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

Related Commands

Command	Description
pattern direct (vm-integration)	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when a user presses the Messages button on a phone.

Command	Description
pattern ext-to-ext busy (vm-integration)	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an internal extension reaches a busy extension and the call is forwarded to voice mail.
pattern ext-to-ext no-answer (vm-integration)	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an internal extension fails to connect to an extension and the call is forwarded to voice mail.
pattern trunk-to-ext busy (vm-integration)	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an external trunk call reaches a busy extension and the call is forwarded to voice mail.
pattern trunk-to-ext no-answer (vm-integration)	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when an external trunk call reaches an unanswered extension and the call is forwarded to voice mail.

voice class tls-cipher

To configure an ordered set of TLS cipher suites, use the **voice class tls-cipher** command. To disable this command or revert to default, use the **no** form of this command.

voice class tls-cipher *tag*
no voice class tls-cipher *tag*

tag	Voice class tls-cipher tag.

Command Default

No default behavior or values

Command Modes

Global configuration (config).

Release	Modification
Cisco IOS XE Cupertino 17.8.1a	Introduced **voice class tls-cipher** command.
Cisco IOS XE Dublin 17.10.1a	Introduced support for YANG models.

Usage Guidelines

The **voice class tls-cipher** command enables voice class configuration mode on the router, allowing you to configure an ordered list of TLS cipher suites.

```
Router(config)#voice class tls-cipher 123
Router(config-class)# cipher ?
    <1-10>  Set the preference order for the cipher-suite (1 = Highest)

Router(config-class)#cipher 1 ?
DHE_RSA_AES128_GCM_SHA256      supported in TLS 1.2 & above
DHE_RSA_AES256_GCM_SHA384      supported in TLS 1.2 & above
DHE_RSA_WITH_AES_128_CBC_SHA   supported in TLS 1.0 & above
DHE_RSA_WITH_AES_256_CBC_SHA   supported in TLS 1.0 & above
ECDHE_ECDSA_AES128_GCM_SHA256  supported in TLS 1.2 & above
ECDHE_ECDSA_AES256_GCM_SHA384  supported in TLS 1.2 & above
ECDHE_RSA_AES128_GCM_SHA256    supported in TLS 1.2 & above
ECDHE_RSA_AES256_GCM_SHA384    supported in TLS 1.2 & above
RSA_WITH_AES_128_CBC_SHA       supported in TLS 1.0 & above
RSA_WITH_AES_256_CBC_SHA       supported in TLS 1.0 & above


Router(config-class)# cipher 1 CIPHER_SUITE_ECDHE_RSA_AES256_GCM_SHA384 ?
<cr>   <cr>
Router(config-class)#
```

voice class tls-profile

To enable voice class configuration mode, and assign an identification tag for a TLS profile, use the command **voice class tls-profile** in global cofiguration mode. To remove a tls-profile, use the **no** form of this command.

voice class tls-profile *tag*

no voice class tls-profile *tag*

Syntax Description

tag	A number used to identify voice class tls profile. The range is 1-10000. There is no default value.

Command Default

No default behavior or values

Command Modes

Global configuration (config)

Command History

Release	Modification
Cisco IOS XE Cupertino 17.8.1a	**tls-profile** command is enhanced to include <tag>.
Cisco IOS XE Amsterdam 17.3.1a	This command was introduced.
Cisco IOS XE Dublin 17.10.1a	Introduced support for YANG models.

Usage Guidelines

The command **voice class tls-profile** enables voice class configuration mode on the router and provides you sub-options to configure commands required for a TLS session. This command allows you to configure under voice class, the options that can be configured at the global level via sip-ua.

The *tag* associates all the voice class configurations that are made through the command **voice class tls-profile** *tag* to the command **crypto signaling**. Following is the **crypto signaling** command with **tls-profile** *tag*:

crypto signaling {**remote-addr** *ip address subnet mask* | **default**} **tls-profile** *tag*

For more information on the updates to the command **crypto signaling**, see crypto signaling.

Examples

The following example configures the **voice class tls-profile** with *tag* '2' and enables voice class cofiguration mode:

```
Router(config)#voice class tls-profile 2
Router(config-class)#
```

The following section provides details of the sub-commands that can be configured under the command **voice class tls-profile** *tag*.

The following example configures CUBE to use the **trustpoint** *trustpoint-name* keyword and argument when it establishes or accepts the TLS connection with a remote device:

```
Router(config-class)#trustpoint CUBETP
```

The following example configures client verification trustpoint:

```
Router(config-class)#client-vtp TPname
```

The following example indicates the description for the TLS profile group:

```
Router(config-class)#description tlsgroupname
```

The following example configures the specific size of elliptic curves to be used for a TLS session:

```
Router(config-class)#cipher ecdsa-cipher curve-size 384
```

The following example configures CUBE to perform server identity validation through Common Name (CN) and Subject Alternate Name (SAN) fields in the server certificate:

```
Router(config-class)#cn-san-validate server
```

The following example enables Server Name Indication (SNI) required during the initial TLS handshake process:

```
Router(config-class)#sni send
```

The following example shows **cipher** command. The command associates a TLS cipher list with this profile.

Note **tls-cipher** should be created before adding to **tls-profile**.

To configure **tls-cipher**:

```
Router#configure terminal
Enter configuration commands, one per line.  End with CNTL/Z.
Router(config)#voice class tls-cipher 1122
Router(config-class)#?
VOICECLASS configuration commands:
  cipher  Configure a TLS cipher-suite
  exit    Exit from voice class configuration mode
  help    Description of the interactive help system
  no      Negate a command or set its defaults

Router(config-class)#cipher ?
  <1-10>  Set the preference order for the TLS cipher-suite (1 = Highest)

Router(config-class)#cipher 1 ?
  DHE_RSA_AES128_GCM_SHA256        supported in TLS 1.2 & above
  DHE_RSA_AES256_GCM_SHA384        supported in TLS 1.2 & above
  DHE_RSA_WITH_AES_128_CBC_SHA     supported in TLS 1.0 & above
  DHE_RSA_WITH_AES_256_CBC_SHA     supported in TLS 1.0 & above
  ECDHE_ECDSA_AES128_GCM_SHA256    supported in TLS 1.2 & above
  ECDHE_ECDSA_AES256_GCM_SHA384    supported in TLS 1.2 & above
  ECDHE_RSA_AES128_GCM_SHA256      supported in TLS 1.2 & above
  ECDHE_RSA_AES256_GCM_SHA384      supported in TLS 1.2 & above
  RSA_WITH_AES_128_CBC_SHA         supported in TLS 1.0 & above
  RSA_WITH_AES_256_CBC_SHA         supported in TLS 1.0 & above

Router(config-class)#cipher 1 ECDHE_RSA_AES256_GCM_SHA384 ?
  <cr>  <cr>

GW1-2A(config-class)#cipher 1 ECDHE_RSA_AES256_GCM_SHA384
GW1-2A(config-class)#
```

To configure **tls-profile**:

```
Router(config-class)#
Router(config-class)#voice class tls-profile 3344
Router(config-class)#?
VOICECLASS configuration commands:
  cipher       Configure a cipher-suite
```

```
  client-vtp   Assign a client verification trustpoint
  cn-san       Configure CN/SAN certificate options
  description  Description of the tls-profile group
  exit         Exit from voice class configuration mode
  help         Description of the interactive help system
  no           Negate a command or set its defaults
  sni          Enable TLS SNI (Server Name Indication) Extension
  trustpoint   Associate a trustpoint
Router(config-class)#ciph
Router(config-class)#cipher ?
  <1-10000>      Specify tls-cipher list tag
  ecdsa-cipher   Configure ECDSA ciphers
  strict-cipher  Configure ciphers mandated by SIP standards

Router(config-class)#cipher 1122 ?
  <cr>  <cr>

Router(config-class)#cipher 1122
```

To show **tls-profile**:

```
Router#show run | sec voice class tls-profile 3344
voice class tls-profile 3344
 cipher 1122
```

To show **tls-cipher**:

```
Router#show run | sec voice class tls-cipher 1122
voice class tls-cipher 1122
 cipher 1 ECDHE_RSA_AES256_GCM_SHA384
```

To show **tls-profile** under **sip-ua**:

```
Router# configure terminal
Router(config)# sip-ua
Router (config-sip-ua)#
 crypto signaling default tls-profile 3344
```

Related Commands

Command	Description
trustpoint	Creates a trustpoint to store the devices certificate that is generated as part of the enrollment process using Cisco IOS public-key infrastructure (PKI) commands.
description	Provides a description for the TLS profile group.
client-vtp	Assigns a client verification trustpoint.
cipher	Configures cipher setting.
cn-san	Enables server identity validation through Common Name (CN) and Subject Alternate Name (SAN) fields in the server certificate during client-side SIP /TLS connections
sni send	Enables TLS Server Name Indication (SNI) during the initial TLS handshake process.
crypto signaling	Identifies the trustpoint or the **tls-profile** *tag* that is used during the TLS handshake process.

voice emergency response location

To create a tag for identifying an emergency response location (ERL) for E911 services, use the **voice emergency response location** command in global configuration mode. To remove the ERL tag, use the **no** form of this command.

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

Syntax Description

tag	Unique number that identifies this ERL tag.

Command Default

No ERL tag is created.

Command Modes

Global configuration (config)

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)	**Address** and **name** commands introduced under **voice emergency response location** command. 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 creates an ERL that identifies an area where emergency teams can quickly locate a 911 caller. The ERL definition optionally includes which ELINs are associated with the ERL and which IP phones are located in the ERL. You can define two or fewer unique IP subnets and two or fewer ELINs. If you define one ELIN, this ELIN is always used for phones calling from this ERL. If you define two ELINs, the system alternates between using each ELIN. If you define zero ELINs and phones use this ERL, the outbound calls do not have their calling numbers translated. The PSAP sees the original calling numbers for these 911 calls. You can optionally add the civic address using the **address** command and an address description using the **name** command.

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. The civic address, 410 Main St, Tooly, CA, and a descriptive identifier, Bldg 3 are included.

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

```
address 1,408,5550100,410,Main St.,Tooly,CA
name Bldg 3
```

Related Commands

Command	Description
address	Specifies a comma separated text entry (up to 250 characters) of an ERL's civic address.
elin	Specifies a PSTN number that will replace the caller's extension.
name	Specifies a string (up to 32-characters) used internally to identify or describe the emergency response location.
subnet	Defines which IP phones are part of this ERL.

voice emergency response settings

To define 911 call behavior settings, use the **voice emergency response settings** command in global configuration mode. To remove the settings, use the **no** form of this command.

voice emergency response settings
no voice emergency response settings

Syntax Description

This command has no arguments or keywords.

Command Default

No default behavior or values

Command Modes

Global configuration (config)

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 enables definition of several 911 of the following call behavior settings:

- **elin**: Default ELIN to use if a 911 caller's IP phone's address does not match the subnet of any location in any zone.
- **expiry**: Number of minutes a 911 call is associated to an ELIN in case of a call back from the 911 operator.
- **callback**: Default number to contact if a 911 call back cannot find the last 911 caller.
- **logging**: Syslog informational message that is printed to the console each time an emergency call is made. This feature is enabled by default, however you can disable this feature by entering the **no** form of this command.

Examples

In the following example, if the 911 caller's IP phone address does not match any of the voice emergency response locations, the ELIN defined in the **voice emergency response settings** configuration (4085550101) is used. After the 911 call is placed to the PSAP, the PSAP has 120 minutes (2 hours) to call back 408 555-0101 to reach the 911 caller. If during a call back, the last caller's extension number cannot be found, the call is routed to extension 7500. The outbound 911 calls do not cause a syslog message to the logging facility (for example, to the 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.
logging	Syslog informational message printed to the console every time an emergency call is made.

voice emergency response zone

To create an emergency response zone, use the **voice emergency response zone** command in global configuration mode. To remove the defined voice emergency response zone, use the **no** form of this command.

voice emergency response zone *tag*
no voice emergency response zone *tag*

Syntax Description

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

Command Default

No default behavior or values

Command Modes

Global configuration (config)

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 voice emergency response zones that allow routing of 911 calls to different PSAPs.

Examples

The following example shows an assignment of ERLs to a voice emergency response zone. The calls have an ELIN from ERLs 8, 9, and 10. The locations for ERLs in zone 10 are searched in the order each CLI was entered for a phone address match because no priority order is assigned.

```
voice emergency response zone 10
location 8
location 9
location 10
```

Related Commands

Command	Description
location	Identifies locations within an emergency response zone and optionally assigns a priority order to the location.

voice hunt-group

To create a hunt group for phones in a Cisco Unified Communications Manager Express (Cisco Unified CME) or Cisco Unified Session Initiation Protocol (SIP) Survivable Remote Site Telephony (SRST) system, use the **voice hunt-group** command in global configuration mode. To delete a hunt group, use the **no** form of this command.

voice hunt-group *hunt-tag* {**longest-idle** | **parallel** | **peer** | **sequential**}
no voice hunt-group *hunt-tag*

Syntax Description

hunt-tag	Unique sequence number that identifies the hunt group. Range is 1 to 100.
longest-idle	Allows an incoming call to go first to the number that has been idle the longest for the number of hops specified when the hunt group was defined. The longest-idle time is determined from the last time that a phone registered, reregistered, or went on-hook.
parallel	Allows an incoming call to simultaneously ring all the numbers in the hunt group member list.
peer	Allows a round-robin selection of the first extension to ring. Ringing proceeds in a circular manner from left to right. The round-robin selection starts with the number left of the number that answered when the hunt-group was last called.
sequential	Allows an incoming call to ring all the numbers in the left-to-right order in which they were listed when the hunt group was defined.

Command Default

No voice hunt group is created.

Command Modes

Global configuration (config)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.
12.4(15)XZ	Cisco Unified CME 4.3	This command was modified to add support for Cisco Unified SCCP IP phones.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.
15.2(4)M	Cisco Unified SIP SRST 9.1	This command was introduced in Cisco Unified SIP SRST 9.1.

Usage Guidelines

The **voice hunt-group** command enters voice hunt-group configuration mode to define a hunt group. A hunt group is a list of phone numbers that take turns receiving incoming calls to a specific number (pilot number), which is defined with the **pilot** command. The specific extensions included in the hunt group and the order and maximum number of extensions allowed in the list are defined with the **list** command.

If a number in the list is busy or does not answer, the call is redirected to the next number in the list. The last number tried is the final number, which is defined with the **final** command. If the number of times that a call is redirected to a new number exceeds 5, you must use the **max-redirect** command to increase the allowable number of redirects in the Cisco Unified CME or Cisco Unified SIP SRST system.

To configure a new hunt group, you must specify the **longest-idle**, **parallel**, **peer**, or **sequential** keyword. To change an existing hunt group configuration, the keyword is not required. To change the type of hunt group, for instance from peer to sequential or sequential to peer, you must remove the existing hunt group first by using the **no** form of this command and then re-create it.

The **parallel** keyword creates a dial peer to allow an incoming call to ring multiple phones simultaneously. The use of parallel hunt groups is also referred to as application-level forking because it enables the forking of a call to multiple destinations. A pilot dial peer cannot be used as a voice hunt group and a hunt group at the same time.

While ephone hunt groups only support Cisco Unified SCCP IP phones, a voice hunt group supports either a Cisco Unified SCCP IP phone or a Cisco Unified SIP IP phone.

With the voice hunt group feature preconfigured in the Cisco Unified SIP SRST router, voice hunt groups continue to be supported after phones fallback from a Cisco Unified Communications Manager (Cisco Unified CM) to a Cisco Unified SIP SRST router.

Examples

The following example shows how to define longest-idle hunt group 1 with pilot number 7501, final number 8000, and nine numbers in the list. After a call is redirected six times (makes 6 hops), it is redirected to the final number 8000.

```
Router(config)# voice hunt-group 1 longest-idle
Router(config-voice-hunt-group)# pilot 7501
Router(config-voice-hunt-group)# list 7001, 7002, 7023, 7028, 7045, 7062, 7067, 7072, 7079
 Router(config-voice-hunt-group)# final 8000

Router(config-voice-hunt-group)# hops 6
Router(config-voice-hunt-group)# timeout 20

Router(config-voice-hunt-group)# exit
```

The following example shows how to define peer hunt group number 2. Callers dial the pilot number 5610 to reach the hunt group. The first extension to ring the first time that this hunt group is called is 5601. If 5601 does not answer, the hunt proceeds from left to right, beginning with the extension directly to the right. If none of those extensions answer, the call is forwarded to extension 6000, which is the number for the voice-mail service.

The second time someone calls the hunt group, the first extension to ring is 5602 if 5601 was answered during the previous call.

```
Router(config)# voice hunt-group 2 peer
Router(config-voice-hunt-group)# pilot 5610
Router(config-voice-hunt-group)# list 5601, 5602, 5617, 5633

Router(config-voice-hunt-group)# final 6000
Router(config-voice-hunt-group)# timeout 30

Router(config-voice-hunt-group)# exit
```

The following example shows how to define sequential hunt group number 3. When callers dial extension 5601, the first phone to ring is 5001, then 5002, 5017, and 5028. If none of those extensions answer, the call is forwarded to extension 6000, which is the number for the voice-mail service.

```
Router(config)# voice hunt-group 3 sequential
Router(config-voice-hunt-group)# pilot 5601
Router(config-voice-hunt-group)# list 5001, 5002, 5017, 5028

Router(config-voice-hunt-group)# final 6000
Router(config-voice-hunt-group)# timeout 30

Router(config-voice-hunt-group)# exit
```

The following example shows how to define a parallel hunt group. When callers dial extension 1000, extensions 1001, 1002, and so forth ring simultaneously. The first extension to answer is connected. All other call legs are disconnected. If none of the extensions answer, the call is forwarded to extension 2000, which is the number for the voice-mail service.

```
Router(config)# voice hunt-group 4 parallel
Router(config-voice-hunt-group)# pilot 1000
Router(config-voice-hunt-group)# list 1001, 1002, 1003, 1004
Router(config-voice-hunt-group)# final 2000

Router(config-voice-hunt-group)# timeout 20

Router(config-voice-hunt-group)# exit
```

Related Commands

Command	Description
final (voice hunt-group)	Defines the last extension in a voice hunt group.
hops (voice hunt-group)	Defines the number of times that a call is redirected to the next phone number in a peer voice hunt-group list before proceeding to the final phone number.
list (voice hunt-group)	Defines the phone numbers that participate in a voice hunt group.
max-redirect	Changes the number of times that a call can be redirected by call forwarding or transfer within a Cisco Unified CME system.
pilot (voice hunt-group)	Defines the phone number that callers dial to reach a voice hunt group.
timeout (voice hunt-group)	Sets the number of seconds after which a call that is not answered is redirected to the next number in the hunt-group list and defines the last phone number in the hunt group.

voice hunt-group login

To authorize an voice register dn or ephone-dn, use the **voice-hunt-groups login** command in voice register-dn configuration mode. To disable this capability, use the **no** form of this command.

voice hunt-groups login
no voice hunt-groups login

Syntax Description

This command has no arguments or keywords.

Command Default

A voice register-dn is not allowed to dynamically join and leave voice hunt groups.

Command Modes

voice register dn configuration (config-voice-register-dn)

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 **show voice hunt-groups** command to display current hunt group members, including those who joined the group dynamically.

Examples

The following example creates five voice-register-dns and a hunt group that includes the first two voice-register-dn and two wildcard slots. The last three voice-register-dns are enabled for group hunt dynamic membership. Each of them can join and leave the hunt group whenever one of the slots is available.

```
voice-register-dn 22
 number 4566
voice-register-dn 23
 number 4567
voice-register-dn 24
 number 4568
 voice-hunt login
voice-register-dn 25
 number 4569
 voice-hunt login
voice-register-dn 26
 number 4570
 voice-hunt-groups login
voice-hunt-groups 1 peer
 list 4566,4567,*,*
 final 7777
```

Related Commands

Command	Description
show voice hunt-groups	Displays voice-hunt group configuration, current status, and statistics.

voice moh-group

To enter voice-moh-group configuration mode and set up music on hold (MOH) group parameters, use the **voice moh-group** command in global configuration mode. To remove the music on hold (MOH) group parameters from the configuration for SCCP IP phones, use the **no** form of this command.

voice moh-group moh-group tag
no voice moh-group tag

Syntax Description

tag	Specifies a moh-group number tag (1-5) to be used for music on hold group parameters.

Command Default

No voice-moh-group is enabled.

Command Modes

Global configuration (config)

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.

Usage Guidelines

This command enters the voice-moh-group configuration mode for configuring music on hold (MOH) group parameters for SCCP IP phones in Cisco Unified CME or in Cisco Unified SRST.

Examples

The following example shows how to enter voice-moh-group configuration mode for configuring a moh group in Cisco Unified CME. This example also includes the command to configure a music on hold (MOH) flash file for this voice-moh- group.

```
Router(config)# voice-moh-group 1
Router(config-voice-moh-group)#moh minuet.wav
```

Related Commands

moh	Enables music on hold from a flash audio feed.
multicast moh	Enables multicast of the music-on-hold audio stream.
extension-range	Defines extension range for a clients calling a voice-moh-group.

voice register global

To enter voice register global configuration mode in order to set global parameters for all supported Cisco SIP IP phones in a Cisco Unified CME or Cisco Unified Session Initiation Protocol (SIP) Survivable Remote Site Telephony (SRST) environment, use the **voice register global** command in global configuration mode. To automatically remove the existing DNs, pools, and global dialplan patterns, use the **no** form of this command.

voice register global
no voice register global

Syntax Description

This command has no arguments or keywords.

Command Default

There are no system-level parameters that are configured for SIP IP phones.

Command Modes

Global configuration (config)

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.
15.0(1)XA	Cisco SIP SRST 8.0	This command was updated to display the signaling transport protocol.
15.1(2)T	Cisco Unified CME 8.1 Cisco Unified SRST 8.1	The no form of the command was modified.
Cisco IOS XE Cupertino 17.7.1a	Cisco Unified SIP SRST 12.8	Introduced support for YANG models.
Cisco IOS XE Cupertino 17.8.1a	Cisco Unified SIP SRST 12.8	Added the new CLI commands **key-server**, and **key-server source-interface <options>** under sip-oauth for **voice register global** mode.

Usage Guidelines

Cisco Unified CME

Use this command to set provisioning parameters for all supported SIP phones in a Cisco Unified CME system.

Cisco Unified SIP SRST

Use this command to set provisioning parameters for multiple pools; that is, all supported Cisco SIP IP phones in a SIP SRST environment.

Cisco Unified CME 8.1 enhances the no form of voice register global command. The no voice register global command clears global configuration along with pools and DN configuration and also removes the configurations for voice register template, voice register dialplan, and voice register session-server. A confirmation is sought before the cleanup is made.

In Cisco Unified SRST 8.1 and later versions, the no voice register global command removes pools and DNs along with the global configuration.

From Cisco IOS XE Cupertino 17.8.1a onwards, the **voice register global** command is enhanced to support the following:

- Key server IP and credentials on SRST to retrieve keys from CUCM and interface specification of source address for OAuth key server.

```
Router#show running-config | section voice register global
voice register global
 sip-oauth     SIP OAuth parameters for Unified SRST
  key-server source-interface GigabitEthernet 1
  key-server ipv4:2.2.2.2 username admin password 6 Ada_SWISQW^TYSN\ZREeFHJTZPCgMMAAB
```

Cisco Unified CME

The following is partial sample output from the **show voice register global** command. All of the parameters listed were set under voice register global configuration mode:

```
Router# show voice register global
CONFIG [Version=4.0(0)]
========================
Version 4.0(0)
Mode is cme
Max-pool is 48
Max-dn is 48
Source-address is 10.0.2.4 port 5060
Load 7960-40 is P0S3-07-4-07
Time-format is 12
Date-format is M/D/Y
Time-zone is 5
Hold-alert is disabled
Mwi stutter is disabled
Mwi registration for full E.164 is disabled
Dst auto adjust is enabled
 start at Apr week 1 day Sun time 02:00
 stop  at Oct week 8 day Sun time 02:00
```

Examples

Cisco Unified CME and Cisco Unified SRST

The following is a sample output from no voice register global command:

```
Router(config)# no voice register global
This will remove all the existing DNs, Pools, Templates,
Dialplan-Patterns, Dialplans and Feature Servers on the system.
Are you sure you want to proceed? Yes/No? [no]:
```

Related Commands

Command	Description
allow connections sip to sip	Allows connections between SIP endpoints in a Cisco multiservice IP-to-IP gateway.
application (voice register global)	Selects the session-level application for all dial peers associated with SIP phones.

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

voice register pool

To enter voice register pool configuration mode for SIP phones, use the **voice register pool** command in global configuration mode. To remove the pool configuration, use the **no** form of this command.

voice register pool *pool-tag*
no voice register pool *pool-tag*

Syntax Description

pool-tag	Unique number assigned to the pool. Range is 1 to 100. **Note** For Cisco Unified Communications Manager Express (Cisco Unified CME) systems, the upper limit for this argument is defined by the **max-pool** command.

Command Default

No default behavior or values

Command Modes

Global 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 CME.
Cisco IOS XE Amsterdam 17.2.1r	Cisco Unified SIP SRST 12.8	Introduced support for YANG models.
Cisco IOS XE Cupertino 17.8.1a	Cisco Unified SIP SRST 14.2	Added a new CLI command **sip-oauth** for **voice register pool** mode.

Usage Guidelines

Cisco Unified CME

Use this command to set phone-specific parameters for SIP phones in a Cisco Unified CME system. Before using this command, enable the **mode cme** command and set the maximum number of SIP phones supported in your system by using the **max-pool** command.

Cisco Unified SIP SRST

Use this command to enable user control on which registrations are to be accepted or rejected by a SIP SRST device. The voice register pool command mode can be used for specialized functions and to restrict registrations on the basis of MAC, IP subnet, and number range parameters.

From Cisco IOS XE Cupertino 17.8.1a onwards, you can enable or disable SIP OAuth based authentication under voice register pool.

For example:

```
Router# voice register pool  1
 sip-oauth
```

Cisco Unified CME

The following example shows how to enter voice register pool configuration mode and forward calls to extension 9999 when extension 2001 is busy:

```
Router(config)# voice register pool 10
Router(config-register-pool)# type 7960
Router(config-register-pool)# number 1 2001
Router(config-register-pool)# call-forward busy 9999 mailbox 1234
```

Cisco Unified SIP SRST

The following partial sample output from the **show running-config** command shows that several voice register pool commands are configured within voice register pool 3:

```
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
```

Enabling or Disabling SIP OAuth

The following is a partial sample output showing SIP OAuth enabled.

```
2055Router#show voice register pool all
 Pool Tag 20
Config:
  Device ID Name is test
  Proxy Ip address is 0.0.0.0
  DTMF Relay is disabled
  kpml signal is enabled
  Lpcor Type is none

  SIP-OAuth is enabled
  Reason for unregistered state:
       No registration request since last reboot/unregister

  paging-dn: config 0 [multicast]  effective 0 [multicast]

VRF:
  NA
```

Related Commands

Command	Description
max-pool (voice register global)	Sets the maximum number of SIP phones that are supported by a Cisco Unified CME system.
mode (voice register global)	Enables the mode for provisioning SIP phones in a Cisco Unified CME system.
number (voice register pool)	Configures a valid number for a SIP phone.

Command	Description
type (voice register pool)	Defines a Cisco IP phone type.
voice register dn	Enters voice register dn configuration mode to define an extension for a SIP phone line.
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 sip oauth get-keys

To retrieve OAuth keys from the CUCM on demand, use the **voice sip oauth get-keys** command.

voice sip oauth get-keys

Command Default Disabled by default.

Command Modes Global configuration mode.

Command History

Release	Modification
Cisco IOS XE Cupertino 17.8.1a	This command was introduced.

Usage Guidelines Use the **voice sip oauth get-keys** command on SRST to get keys from the call manager.

Note Key server should be configured under **voice register**.

Keys are fetched every 24 hours or within 24 hours of last command execution.

To get the keys using the **voice sip oauth get-keys** command, key server should be configured first.

```
Router#voice sip oauth get-keys
Successfully triggered http request to fetch sip-oauth keys
Router#
```

voice vrf

To configure a voice VRF, use the **voice vrf** command in global configuration mode. To remove the voice VRF configuration, use the **no** form of this command.

voice vrf *vrfname*
no voice vrf *vrfname*

Syntax Description

vrfname	A name assigned to the voice vrf.

Command Default

No voice VRF is configured.

Command Modes

Global configuration

Command History

Cisco IOS Release	Cisco Product	Modification
Cisco IOS XE Amsterdam 17.2.1r	Unified SRST 12.8	This command was introduced as part of VRF functionality support on Cisco 4000 Series Integrated Services Router for Unified SRST.

Usage Guidelines

You must create a VRF using the **vrf definition** *vrf-name* command before you can configure it as a voice VRF. This configuration is both for trunk side and Unified SRST line side.

Voice VRF configuration

The following is a sample configuration for **voice vrf** in Unified SRST line side:

```
vrf definition vrf1
 rd 100:101
 !
 address-family ipv4
 exit-address-family
voice vrf vrf1
interface GigabitEthernet0/0/0
 vrf forwarding vrf1
 ip address 8.44.22.77 255.255.0.0
ip route vrf vrf1 8.0.0.0 255.0.0.0 8.44.0.1
```

Related Commands

Command	Description
vrf definition *vrf-name*	Creates a VRF with the specified name.

voice-class codec (voice register pool)

To assign a previously configured codec selection preference list, use the **voice-class codec** command in voice register pool configuration mode. To remove the codec preference assignment from the voice register pool, use the no form of this command.

voice-class codec *tag*
no voice-class codec

Syntax Description

tag	Unique number assigned to the voice class. Range is from 1 to 10000. The tag number maps to the tag number created by using the **voice class codec** command in dial-peer configuration mode.

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 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 Communications Manager Express (Cisco Unified CME) registration, a dial peer is created, and that dial peer includes codec g729r8 by default. The **voice-class codec** command allows you to change the automatically selected default codec, if desired.

You can assign one voice class to each voice register pool. If you assign another voice class to a pool, the last voice class assigned replaces the previous voice class.

Note The **id** (voice register pool) command is required and must be configured before any other voice register pool commands. The **id** command identifies a locally available individual Cisco SIP IP phone or set of Cisco SIP IP phones.

Examples

The following partial sample output from the **show running-config** command shows that voice register pool 1 has been set up to use the previously configured codec voice class 1:

```
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
```

Related Commands

Command	Description
codec (voice register pool)	Specifies the codec supported by a single Cisco SIP phone or a VoIP dial peer in a Cisco Unified SIP SRST or a Cisco Unified CME 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.
voice register pool	Enters voice register pool configuration mode for SIP phones.
voice class codec (dial-peer)	Assigns a previously configured codec selection preference list (codec voice class) to a VoIP dial peer.

voicemail (call-manager-fallback)

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

voicemail *phone-number*
no voicemail

Syntax Description

phone-number	Phone number configured as a speed-dial number for retrieving messages.

Command Default

No phone number is configured, and the messages button is ineffective.

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 **voicemail** command configures the telephone number that is speed-dialed when the messages button on a Cisco IP phone is pressed. The same voice-mail telephone number is configured for all Cisco IP phones connected to the router.

Examples

The following example specifies 4085550100 as the speed-dial number that is dialed to retrieve messages when the messages button is pressed:

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# voicemail 914085550100
```

The number 914085550100 is called when the Cisco IP phone messages button is pressed to retrieve messages.

Related Commands

Command	Description
call-manager-fallback	Enables Cisco Unified Survivable Remote Site Telephony (SRST) support and enters call-manager-fallback configuration mode.

vrf definition

To configure a VRF with a specified name, use the **vrf definition** command in global configuration mode. To remove the VRF configuration, use the **no** form of this command.

vrf definition *vrf-name*
no vrf definition *vrf-name*

Syntax Description

vrf-name	A name assigned to the configured vrf.

Command Default

No VRF is configured.

Command Modes

Global configuration

Command History

Cisco IOS Release	Cisco Product	Modification
Cisco IOS XE Amsterdam 17.2.1r	Unified SRST 12.8	This command was introduced as part of VRF functionality support on Cisco 4000 Series Integrated Services Router for Unified SRST.

Usage Guidelines

You can create a VRF with a specified name using the **vrf definition** *vrf-name* command. Space is not allowed in the VRF name. For example, **vrf 1** is not a valid name for a VRF. However, **vrf1** as shown in the example below is a valid name.

VRF Definition configuration

The following is a sample configuration for **vrf definition** in Unified SRST line side:

```
vrf definition vrf1
 rd 100:101
 !
 address-family ipv4
 exit-address-family
```

Related Commands

Command	Description
voice vrf *vrfname*	Configures a Voice VRF in global configuration mode.

vrf forwarding

To associate a VRF instance with the tunnel, use the **vrf forwarding** command in interface configuration mode. To remove the association of VRF instance with the tunnel, use the **no** form of this command.

vrf forwarding *customer-vrf-name*
no vrf forwarding *customer-vrf-name*

Syntax Description

customer-vrf-name	A name assigned to the customer vrf.

Command Default

No VRF is forwarded.

Command Modes

Interface configuration

Command History

Cisco IOS Release	Cisco Product	Modification
Cisco IOS XE Amsterdam 17.2.1r	Unified SRST 12.8	This command was introduced as part of VRF functionality support on Cisco 4000 Series Integrated Services Router for Unified SRST.

Usage Guidelines

You can associate the customer VRF instance with the tunnel using the **vrf forwarding** *customer-vrf-name* command. Packets exiting the tunnel are forwarded to this VRF (inner IP packet routing).

VRF Forwarding configuration

The following is a sample configuration for **vrf forwarding** for Unified SRST:

```
voice vrf vrf1
interface GigabitEthernet0/0/0
 vrf forwarding vrf1
 ip address 8.44.22.77 255.255.0.0
ip route vrf vrf1 8.0.0.0 255.0.0.0 8.44.0.1
```

Related Commands

Command	Description
voice vrf *vrfname*	Configures a Voice VRF in global configuration mode.
vrf definition *vrf-name*	Creates a VRF with the specified name.

xmlschema (call-manager-fallback)

To specify the URL for a Cisco Unified Survivable Remote Site Telephony (SRST) eXtensible Markup Language (XML) application program interface (API) schema, use the **xmlschema** command in call-manager-fallback configuration mode. To set the URL for the XML API schema to the default, use the **no** form of this command.

xmlschema *schema-url*
no xmlschema

Syntax Description

schema-url	Local or remote URL as defined in RFC 2396.

Command Default

srst-its.xsd

Command Modes

Call-manager-fallback configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)T	Cisco SRST 3.2	This command was introduced.

Examples

The following example specifies a URL for an XML API schema:

```
Router(config)# call-manager-fallback
Router(config-cm-fallback)# xmlschema http://server2.example.com/schema/schema1.xsd
```

Related Commands

Command	Description
call-manager-fallback	Enable Cisco Unified SRST configuration mode.