



Cisco IOS SIP SRST Version 3.4 Command Reference

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Cisco IOS SIP SRST Version 3.4 Command Reference

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Command Reference: A through M

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

For detailed information on how to configure Cisco SIP SRST applications and features, see the *Cisco IOS SIP SRST Version 3.4 System Administrator Guide*.

- **after-hour exempt (voice register pool)**
- **alias (voice register pool)**
- **application (voice register global)**
- **application (voice register pool)**
- **b2bua**
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- **dialplan-pattern (voice register pool)**
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- **external-ring (voice register global)**
- **id (voice register pool)**
- **incoming called-number (voice register pool)**
- **max-pool (voice register global)**
- **max registrations (voice register pool)**

after-hour exempt (voice register pool)

To specify that for a particular voice register pool no outgoing calls are blocked even though global system call blocking is enabled, use the **after-hours exempt** command in voice register pool configuration mode. To return to the default, use the **no** form of this command.

after-hour exempt

no after-hour exempt

Syntax Description

This command has no arguments or keywords.

Defaults

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

Command Modes

Voice register pool configuration

Command History

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

Usage Guidelines

This command exempts individual Cisco SIP phones and phone extensions from call blocking.

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

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



Note

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

Examples

The following example exempts blocking of outgoing calls from SIP phone 23:

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

The following example specifies that outgoing calls from extension 5001 under voice register pool 2 are not blocked:

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

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

alias (voice register pool)

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

alias *tag pattern* **to** *target* [**preference** *value*]

no *alias tag*

Syntax Description

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

Defaults

None

Command Modes

Voice register pool configuration

Command History

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

Usage Guidelines

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

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

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

**Note**

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

Before the **alias** command is configured, translation rules must be set using the **translate-outgoing (voice register pool)** command. Translation rules are a general-purpose number-manipulation mechanism that perform operations such as automatically adding telephone area and prefix codes to dialed numbers.

Examples

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

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

Related Commands

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

application (voice register global)

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

application *application-name*

no application

Syntax Description

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

Defaults

Default application on router

Command Modes

Voice register global configuration

Command History

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

Usage Guidelines

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

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



Note

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

Examples

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

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

Related Commands

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

application (voice register pool)

To select the session-level application for the dial peer associated an individual SIP phone in a Cisco CME environment or for a group of phones in a Cisco SIP SRST environment, use the **application** command in voice register pool configuration mode. To disable use of the application, use the **no** form of this command.

application *application-name*

no application

Syntax Description

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

Defaults

Default application on router

Command Modes

Voice register pool configuration

Command History

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

Usage Guidelines

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

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



Note

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

Examples

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

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

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

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

Related Commands

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

b2bua

To configure a dial peer associated with an individual SIP phone in a Cisco CME environment or a group of phones in a Cisco SIP SRST environment to point to Cisco Unity Express, use the **b2bua** command in dial-peer configuration mode. To disable B2BUA call flow on the dial peer, use the **no** form of this command.

b2bua

no b2bua

Syntax Description This command has no arguments or keywords.

Defaults Disabled

Command Modes Dial-peer configuration

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

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



Note

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

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

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

Related Commands	Command	Description
	allow-connections	Enables calls between SIP endpoints in a VoIP network.
	dial-peer voice	Defines a dial peer and enters dial-peer configuration mode.
	mode (voice register global)	Enables the mode for provisioning SIP phones in a Cisco CallManager Express (Cisco CME) system.

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

call-forward b2bua all (voice register dn and voice register pool)

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

call-forward b2bua all *directory-number*

no call-forward b2bua all

Syntax Description

<i>directory-number</i>	Telephone number to which calls are forwarded. Represents a fully qualified E.164 number. Maximum length of the telephone number is 32.
-------------------------	---

Defaults

Disabled (no incoming call forwarding to another extension).

Command Modes

Voice register dn configuration
Voice register pool configuration

Command History

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

Usage Guidelines

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

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

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

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



Note

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

Examples

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

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


The following example forwards to extension 5005 all incoming calls to extension 5001 on pool number 4.

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

Related Commands

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

call-forward b2bua busy (voice register dn and voice register pool)

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

call-forward b2bua busy *directory-number*

no call-forward b2bua busy

Syntax Description	<i>directory-number</i>	Telephone number to which calls are forwarded. Represents a fully qualified E.164 number. Maximum length of the telephone number is 32.
---------------------------	-------------------------	---

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

Command Modes Voice register dn configuration
Voice register pool configuration

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

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

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

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

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



Note

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

Cisco CME

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

Examples

The following example forwards calls to extension 5005 when extension 5001 in pool number 4 is busy.

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

Related Commands

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

call-forward b2bua mailbox (voice register dn and voice register pool)

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

call-forward b2bua mailbox *directory-number*

no call-forward b2bua mailbox

Syntax Description

<i>directory-number</i>	Telephone number to which calls are forwarded when the forwarded destination is busy or does not answer. Represents a fully qualified E.164 number. Maximum length of the telephone number is 32.
-------------------------	---

Defaults

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

Command Modes

Voice register dn configuration
Voice register pool configuration

Command History

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

Usage Guidelines

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

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

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

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



Note

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

Examples

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

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

Related Commands

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

call-forward b2bua noan (voice register dn and voice register pool)

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

call-forward b2bua noan *directory-number* **timeout** *seconds*

no call-forward b2bua noan

Syntax Description		
	<i>directory-number</i>	Telephone number to which calls are forwarded. Represents a fully qualified E.164 number. Maximum length of the telephone number is 32.
	timeout <i>seconds</i>	Number of seconds that a call can ring with no answer before the call is forwarded to another extension. Range is 3 to 60000. Default is 20.

Defaults

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

Command Modes

Voice register dn configuration
Voice register pool configuration

Command History

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

Usage Guidelines

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

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

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

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



Note

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

Examples

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

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

Related Commands

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

codec (voice register pool)

To specify the codec supported by a single Cisco SIP phone or a VoIP dial peer in a Cisco SIP SRST or a Cisco CME environment, use the **codec** command in voice register pool configuration mode. To disable a specified codec, use the **no** form of this command.

codec *codec-type* [*bytes*]

no codec

Syntax Description

<i>codec-type</i>	Specifies the preferred codec: <ul style="list-style-type: none"> • g711alaw—G.711 a-law 64,000 bps • g711ulaw—G.711 mu-law 64,000 bps. • g729r8—G.729 8000 bps (this is the default).
<i>bytes</i>	(Optional) Specifies the number of bytes in the voice payload of each frame.

Defaults

g729r8

Command Modes

Voice register pool configuration

Command History

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

Usage Guidelines

This command sets the codec configuration for an individual phone and overrides any previously configured codec selection set with the **voice-class codec** command.



Note

Configure the **id** (voice register pool) command 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 example shows codec complexity set to g729r8 for a Cisco SIP IP phone in pool 1:

```
Router(config)# voice register pool 1
Router(config-register-pool)# codec g729r8
```

Related Commands

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

Command	Description
voice-class codec	Assigns a previously configured codec selection preference list.
voice register pool	Enters voice register pool configuration mode for SIP phones.

cor (voice register pool)

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

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

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

Syntax Description

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

Defaults

None

Command Modes

Voice register pool configuration

Command History

Cisco IOS Release	Version	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

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

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

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

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

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

**Note**

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

Examples

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

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

Related Commands

Command	Description
dial-peer cor custom	Specifies that named CORs apply to dial peers.
dial-peer cor list	Defines a COR list name.
id (voice register pool)	Explicitly identifies a locally available individual Cisco SIP IP phone, or when running Cisco SIP SRST, set of Cisco SIP IP phones.

Command	Description
member (dial-peer cor list)	Adds a member to a dial-peer COR list.
name (dial-peer custom cor)	Provides a name for a custom COR.
show dial-peer voice	Displays information for voice dial peers.
voice register pool	Enables Cisco SIP SRST voice register pool configuration commands.

debug voice register errors

To display debug information on voice register module errors during registration in a Cisco CME or Cisco SIP SRST environment, use the **debug voice register errors** command in privileged EXEC mode. To disable debugging, use the **no** form of the command.

debug voice register errors

no debug voice register errors

Syntax Description This command has no arguments or keywords

Defaults Disabled

Command Modes Privileged EXEC mode

Command History	Cisco IOS Release	Version	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 Registration errors include failure to match pools or any internal errors that happen during registration.

Examples

Cisco SIP SRST

The following is sample output from this command:

```
Router# debug voice register errors
```

```
*Apr 22 11:52:54.523 PDT: VOICE_REG_POOL: Contact doesn't match any pools
*Apr 22 11:52:54.539 PDT: VOICE_REG_POOL: Register request for (33015) from (10.2.152.39)
*Apr 22 11:52:54.539 PDT: VOICE_REG_POOL: Contact doesn't match any pools.
*Apr 22 11:52:54.559 PDT: VOICE_REG_POOL: Register request for (33017) from (10.2.152.39)
*Apr 22 11:53:04.559 PDT: VOICE_REG_POOL: Maximum registration threshold for pool(3) hit
```

If there are no voice register pools configured for a particular registration request, the message “Contact doesn’t match any pools” is displayed.

If the **max registrations** command is configured, when registration requests reach the maximum limit, the “Maximum registration threshold for pool(x) hit” message is displayed for the particular pool.

[Table 1](#) describes the significant fields shown in the display.

Table 1 *debug voice register errors Field Descriptions*

Field	Description
Contact (doesn't match any pools)	Contact refers to the location of the SIP devices and the IP address.
Register request for (<i>telephone number</i>) from (<i>IP address</i>).	The unique key for each registration is the telephone number.

Related Commands

Command	Description
debug voice register events	Displays debug information on voice register module events during SIP phone registrations in a Cisco CME or Cisco SIP SRST environment.

debug voice register events

To display debug information on voice register module events during SIP phone registrations in a Cisco CME or Cisco SIP SRST environment, use the **debug voice register events** command in privileged EXEC mode. To disable debugging, use the **no** form of this command.

debug voice register errors

no debug voice register errors

Syntax Description This command has no arguments or keywords

Defaults Disabled

Command Modes Privileged EXEC mode

Command History	Cisco IOS Release	Version	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 Using the **debug voice register events** command should suffice to view registration activity. Registration activity includes matching of pools, registration creation, and automatic creation of dial peers. For more details and error conditions, you can use the **debug voice register errors** command.

Cisco SIP SRST

The following is sample output from this command:

```
Router# debug voice register events

Apr 22 10:50:21.731 PDT: VOICE_REG_POOL: Contact matches pool 1
Apr 22 10:50:21.731 PDT: VOICE_REG_POOL: key(91011) contact(192.168.0.2) add to contact table
Apr 22 10:50:21.731 PDT: VOICE_REG_POOL: key(91011) exists in contact table
Apr 22 10:50:21.731 PDT: VOICE_REG_POOL: contact(192.168.0.2) exists in contact table, ref updated
Apr 22 10:50:21.731 PDT: VOICE_REG_POOL: Created dial-peer entry of type 1
Apr 22 10:50:21.731 PDT: VOICE_REG_POOL: Registration successful for 91011, registration id is 257
```

The phone number 91011 registered successfully, and *type 1* is reported in the debug, which means that there is a preexisting VoIP dial peer.

```
Apr 22 10:50:38.119 PDT: VOICE_REG_POOL: Register request for (91021) from (192.168.0.3)
Apr 22 10:50:38.119 PDT: VOICE_REG_POOL: Contact matches pool 2
```

```

Apr 22 10:50:38.123 PDT: VOICE_REG_POOL: key(91021) contact(192.168.0.3) add to contact
table
Apr 22 10:50:38.123 PDT: VOICE_REG_POOL: key(91021) exists in contact table
Apr 22 10:50:38.123 PDT: VOICE_REG_POOL: contact(192.168.0.3) exists in contact table, ref
updated
Apr 22 10:50:38.123 PDT: VOICE_REG_POOL: Created dial-peer entry of type 1
Apr 22 10:50:38.123 PDT: VOICE_REG_POOL: Registration successful for 91021, registration
id is 258

```

A dynamic VoIP dial peer has been created for entry 91021. The dial peer can be verified using the **show voice register dial-peers** and **show sip-ua status registrar** commands.

```

Apr 22 10:51:08.971 PDT: VOICE_REG_POOL: Register request for (95021) from (10.2.161.50)
Apr 22 10:51:08.971 PDT: VOICE_REG_POOL: Contact matches pool 3
Apr 22 10:51:08.971 PDT: VOICE_REG_POOL: key(95021) contact(10.2.161.50) add to contact
table
Apr 22 10:51:08.971 PDT: VOICE_REG_POOL: No entry for (95021) found in contact table
Apr 22 10:51:08.975 PDT: VOICE_REG_POOL: key(95021) contact(10.2.161.50) added to contact
table
Apr 22 10:51:08.979 PDT: VOICE_REG_POOL: Created dial-peer entry of type 0
Apr 22 10:51:08.979 PDT: VOICE_REG_POOL: Registration successful for 95021, registration
id is 259
Apr 22 10:51:09.019 PDT: VOICE_REG_POOL: Register request for (95012) from (10.2.161.50)
Apr 22 10:51:09.019 PDT: VOICE_REG_POOL: Contact matches pool 3
Apr 22 10:51:09.019 PDT: VOICE_REG_POOL: key(95012) contact(10.2.161.50) add to contact
table
Apr 22 10:51:09.019 PDT: VOICE_REG_POOL: No entry for (95012) found in contact table
Apr 22 10:51:09.023 PDT: VOICE_REG_POOL: key(95012) contact(10.2.161.50) added to contact
table
Apr 22 10:51:09.027 PDT: VOICE_REG_POOL: Created dial-peer entry of type 0
Apr 22 10:51:09.027 PDT: VOICE_REG_POOL: Registration successful for 95012, registration
id is 260
Apr 22 10:51:09.071 PDT: VOICE_REG_POOL: Register request for (95011) from (10.2.161.50)
Apr 22 10:51:09.071 PDT: VOICE_REG_POOL: Contact matches pool 3
Apr 22 10:51:09.071 PDT: VOICE_REG_POOL: key(95011) contact(10.2.161.50) add to contact
table
Apr 22 10:51:09.071 PDT: VOICE_REG_POOL: No entry for (95011) found in contact table
Apr 22 10:51:09.075 PDT: VOICE_REG_POOL: key(95011) contact(10.2.161.50) added to contact
table
Apr 22 10:51:09.079 PDT: VOICE_REG_POOL: Created dial-peer entry of type 0
Apr 22 10:51:09.079 PDT: VOICE_REG_POOL: Registration successful for 95011, registration
id is 261
Apr 22 10:51:09.123 PDT: VOICE_REG_POOL: Register request for (95500) from (10.2.161.50)
Apr 22 10:51:09.123 PDT: VOICE_REG_POOL: Contact matches pool 3
Apr 22 10:51:09.123 PDT: VOICE_REG_POOL: key(95500) contact(10.2.161.50) add to contact
table
Apr 22 10:51:09.123 PDT: VOICE_REG_POOL: No entry for (95500) found in contact table
Apr 22 10:51:09.127 PDT: VOICE_REG_POOL: key(95500) contact(10.2.161.50) added to contact
table
Apr 22 10:51:09.131 PDT: VOICE_REG_POOL: Created dial-peer entry of type 0
Apr 22 10:51:09.131 PDT: VOICE_REG_POOL: Registration successful for 95500, registration
id is 262
*Apr 22 11:52:54.523 PDT: VOICE_REG_POOL: Contact doesn't match any pools
*Apr 22 11:52:54.539 PDT: VOICE_REG_POOL: Register request for (33015) from (10.2.152.39)
*Apr 22 11:52:54.539 PDT: VOICE_REG_POOL: Contact doesn't match any pools
*Apr 22 11:52:54.559 PDT: VOICE_REG_POOL: Register request for (33017) from (10.2.152.39)

```

[Table 2](#) describes the significant fields shown in the display.

Table 2 *debug voice register events Field Descriptions*

Field	Description
Contact	Indicates the location of the SIP devices and may indicate the IP address.
contact table	The table that maintains the location of the SIP devices.
key	The phone number is used as the unique key to maintain registrations of SIP devices.
multiple contact	More than one registration matches the same phone number.
no entry	The incoming registration was not found.
type 0	Normal dial peer.
type 1	Existing normal dial peer.
type 2	Proxy dial peer.
type 3	Existing proxy dial peer.
type 4	Dial-plan dial peer.
type 5	Existing dial-plan dial peer.
type 6	Alias dial peer.
type 7	Existing alias dial peer.
un-registration successful	The incoming unregister was successful.
Register request/registration id <i>number</i>	The internal unique number for each registration; useful for debugging particular registrations.

Related Commands

Command	Description
debug voice register errors	Displays debug information on voice register module errors during registration in a Cisco CME or Cisco SIP SRST environment.
show sip-ua status registrar	Displays all the SIP endpoints that are currently registered with the contact address.
show voice register dial-peers	Displays details of SIP SRST configuration and of all dynamically created VoIP dial peers.

dialplan-pattern (voice register pool)



Note

Effective with Cisco IOS Release 12.4(4)T, the **dialplan-pattern** command is not visible in Cisco IOS software. For similar functionality, use the [translation-rule](#) command.

To create a global prefix that can be used to expand the abbreviated extension numbers (automatically obtained from the Cisco IP phone) into fully qualified E.164 telephone numbers, use the **dialplan-pattern** command in voice register pool configuration mode. To disable a global prefix, use the **no** form of this command.

```
dialplan-pattern tag pattern extension-length length [extension-pattern extension-pattern]
```

```
no dialplan-pattern tag
```

Syntax Description

<i>tag</i>	Dial-plan string tag used before a 10-digit telephone number. The range is from 1 to 5.
<i>pattern</i>	Dial-plan pattern, such as the area code, the prefix, and the first one or two digits of the extension number, plus dots (.) for the remainder of the extension number digits.
extension-length <i>length</i>	Sets the number of extension digits. The range is from 1 to 32.
extension-pattern <i>extension-pattern</i>	(Optional) Sets an extension number's leading digit pattern when it is different from the E.164 telephone number's leading digits defined in the <i>pattern</i> argument. The argument <i>extension-pattern</i> consists of one or more digits and wildcard markers or dots (.). For example, 5.. would include extensions 500 to 599, and 5... would include extensions 5000 to 5999.

Defaults

Global prefixes are disabled.

Command Modes

Voice register pool configuration

Command History

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

Usage Guidelines

The **dialplan-pattern** command creates a global prefix that can be used to expand the abbreviated extension numbers into fully qualified E.164 telephone numbers. The extension number should be greater than or equal to the extension length. Otherwise, the extension number cannot be converted into a qualified E.164 number.

The **extension-length** keyword enables the system to convert a full E.164 telephone number back into an extension number. For example, a company uses extension number range 0100 to 0199 across several sites, with only the extensions 0100 to 0199 present on the local router. An incoming call to 0199 arrives as 4085550199 in its full E.164 format. The **dialplan-pattern** command translates to the extension number and routes the call to Cisco SIP IP phone extension 0199. The **dialplan-pattern** command creates an additional VoIP dial peer with a destination pattern of 4085550199 when extension 0199 registers to SIP SRST. Then the IP address of extension 0199 is used as the session target. In order for full E.164 telephone calls to be accepted by Cisco SIP IP phone 0199, a translation rule must be applied in a voice register pool to convert the full E.164 telephone called number into the extension number.

The number of extension-pattern characters must match the extension length (for example, if the extension length is three, the extension pattern can be 8.., 1.., 5...0).

**Note**

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

Before configuring the **dialplan-pattern** command, translation rules must be set using the **translate-outgoing** command. Translation rules are a general-purpose number-manipulation mechanism that perform operations such as automatically adding telephone area and prefix codes to dialed numbers.

Examples

The following example shows how to create dial-plan pattern 1 for extension numbers 0100 to 0199 with the telephone prefix starting with 408555. If the following commands are configured, the routers recognize that the number 4085550100 matches dial-plan pattern 1 and use the **extension-length** keyword to extract the last four digits of the number 4085550100 and present this as the caller ID for the incoming call.

```
Router(config)# voice register pool
Router(config-register-pool)# dialplan-pattern 1 40855501.. extension-length 4
```

Related Commands

Command	Description
id (voice register pool)	Explicitly identifies a locally available individual Cisco SIP IP phone or set of Cisco SIP IP phones.
translate-outgoing	Applies a translation rule to manipulate dialed digits on an outbound POTS or VoIP call leg.
translation-rule	Creates a translation name and enters translation-rule configuration mode to apply rules to the translation name.
voice register pool	Enables SIP SRST voice register pool configuration commands.

dtmf-relay (voice register pool)

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

dtmf-relay [cisco-rtp] [rtp-nte] [sip-notify]

no dtmf-relay

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

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

Command Modes Voice register pool configuration

Command History	Cisco IOS Release	Version	Modification
	12.3(4)T	Cisco SIP SRST 3.0	This command was introduced.
	12.4(4)T	Cisco CME 3.4 and Cisco SIP SRST 3.4	This command was added to Cisco CME.

Usage Guidelines During Cisco SIP SRST or Cisco CME registration, a dial peer is created and that dial peer has a default DTMF relay of in-band. The **dtmf-relay** command allows you to change the default to a desired value. DTMF audio tones are generated when you press a button on a touchtone phone. The tones are compressed at one end of the call and when the digits are decompressed at the other end, there is a risk that they can become distorted. DTMF relay reliably transports the DTMF audio tones generated after call establishment out-of-band.

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

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

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

**Note**

- The **cisco-rtp** keyword is a proprietary Cisco implementation. If the proprietary Cisco implementation is not supported, the DTMF relay feature does not function, and the gateway sends DTMF tones in-band.
- The **sip-notify** keyword is available only if the VoIP dial peer is configured for SIP.

Examples**Cisco CME**

The following example enables the RTP-NTE and SIP-NOTIFY mechanisms for DTMF relay for SIP phone 4:

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

Cisco SIP SRST

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

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

Related Commands

Command	Description
dtmf-relay (voice over IP)	Specifies how an H.323 or SIP gateway relays DTMF tones between telephony interfaces and an IP network.
voice register pool	Enters voice register pool configuration mode for SIP phones.

external-ring (voice register global)

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

```
external-ring {bellcore-dr1 | bellcore-dr2 | bellcore-dr3 | bellcore-dr4 | bellcore-dr5}
```

```
no external-ring
```

Syntax Description

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

Defaults

The default ring sound is an internal ring pattern.

Command Modes

Voice register global configuration

Command History

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

Usage Guidelines

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

Examples

The following example specifies that Bellcore DR1 be used for external ringing on Cisco SIP IP phones:

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

Related Commands

Command	Description
voice register global	Enters voice register global configuration mode in order to set global parameters for all supported Cisco SIP phones in a Cisco CME or Cisco SIP SRST environment.

id (voice register pool)

To explicitly identify a locally available individual Cisco SIP IP phone, or when running Cisco SIP SRST, set of Cisco SIP IP phones, use the **id** command in voice register pool configuration mode. To remove local identification, use the **no** form of this command.

```
id {network address mask mask | ip address mask mask | mac address}
```

```
no id {network address mask mask | ip address mask mask | mac address}
```

Syntax Description

network <i>address mask mask</i>	The network <i>address mask mask</i> keyword/argument combination is used to accept SIP Register messages for the indicated phone numbers from any IP phone within the specified IP subnet.
ip <i>address mask mask</i>	The ip <i>address mask mask</i> keyword/argument combination is used to identify an individual phone.
mac <i>address</i>	The mac <i>address</i> keyword/argument combination is used to identify the MAC address of a particular Cisco IP phone.

Command Default

None

Command Modes

Voice register pool configuration

Command History

Release	Version	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

Configure the **id** (voice register pool) command before any other voice register pool commands.

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

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

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

**Note**

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

Examples

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

```
voice register pool 1
  id mac 0030.94C2.A22A
  preference 5
  cor incoming call91 1 91011
  translate-outgoing called 1
  proxy 10.2.161.187 preference 1 monitor probe icmp-ping
  alias 1 94... to 91011 preference 8
  voice-class codec 1
```

Related Commands

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

incoming called-number (voice register pool)

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

incoming called-number [*number*]

no incoming called-number

Syntax Description	<i>number</i> (Optional) Sequence of digits that represent a phone number prefix.
---------------------------	---

Defaults	None
-----------------	------

Command Modes	Voice register pool configuration
----------------------	-----------------------------------

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

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

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

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

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

max-pool (voice register global)

To set the maximum number of SIP voice register pools that are supported in a Cisco SIP SRST or Cisco CME environment, use the **max-pool** command in voice register global configuration mode. To reset the maximum number to the default, use the **no** form of this command.

max-pool *max-voice-register-pools*

no max-pool

Syntax Description

<i>max-voice-register-pools</i>	Maximum number of SIP voice register pools supported by the Cisco router. The upper limit of voice register pools is version- and platform-dependent; see Cisco IOS command-line interface (CLI) help. Default is 0.
---------------------------------	--

Defaults

Default is 0 pools.

Command Modes

Voice register global configuration

Command History

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

Usage Guidelines

This command limits the number of SIP phones supported by a Cisco CME or Cisco SIP SRST environment. The **max-pool** command is platform specific and defines the limit for the **voice register pool** command. The **max-dn** command similarly limits the number of directory numbers (extensions) in a Cisco CME system.



Note

You can increase the number of phones; but after the maximum allowable number is configured, you cannot reduce the limit of the SIP phones without rebooting the router.

Examples

The following example sets the maximum number of Cisco SIP IP phones in a Cisco SIP SRST environment to 24:

```
Router(config)# voice register global
Router(config-register-global)# max-pool 24
```

Related Commands

Command	Description
max-dn (voice register global)	Set the maximum number of SIP phone directory numbers (extensions) that are supported by a Cisco CallManager Express (Cisco CME) router.
voice register global	Enters voice register global configuration mode in order to set global parameters for all supported Cisco SIP phones in a Cisco CME or Cisco SIP SRST environment.
voice register pool	Enters voice register pool configuration mode for Cisco SIP IP phones.

max registrations (voice register pool)



Note

Effective with Cisco IOS Release 12.4(4)T, the **max registrations** command is not visible in Cisco IOS software. For similar functionality, use the **max-pool (voice register global)** command.

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

max registrations *value*

no max registrations

Syntax Description

<i>value</i>	Digit, beginning with 0, that represents the maximum number of registrations. The maximum registration value is platform dependent.
--------------	---

Defaults

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

Command Modes

Voice register pool configuration

Command History

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

Usage Guidelines

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

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

Examples

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

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

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

■ max registrations (voice register pool)



Command Reference: N through Z

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

For detailed information on how to configure Cisco SIP SRST applications and features, see the [Cisco IOS SIP SRST Version 3.4 System Administrator Guide](#).

- **notify redirect (dial peer)**
- **notify redirect (voice service)**
- **number (voice register pool)**
- **preference (voice register pool)**
- **proxy (voice register pool)**
- **registrar server (SIP)**
- **show sip-ua status registrar**
- **show voice register all**
- **show voice register dial-peers**
- **show voice register global**
- **show voice register pool**
- **show voice register statistics**
- **translate-outgoing (voice register pool)**
- **vad (voice register pool)**
- **voice-class codec (voice register pool)**
- **voice register global**
- **voice register pool**

notify redirect (dial peer)

To send a redirect facility to the application handling the redirect request on a specific VoIP dial peer using the Cisco IOS voice gateway, use the **notify redirect** command in the dial-peer configuration mode. To return to the default, use the **no** form of this command.

notify redirect {ip2ip | ip2pots}

no notify redirect

Syntax Description

ip2ip	Sends redirect facility to the application handling redirect requests for IP-to-IP calls.
ip2pots	Sends redirect facility to the application handling redirect requests for IP-to-POTS calls.

Defaults

Disabled

Command Modes

Dial peer configuration

Command History

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

Usage Guidelines

The **notify redirect (dial peer)** command must be configured on the inbound dial peer of the gateway. This command enables, on a per dial peer basis, IP-to-IP or IP-to-POTS notify redirection for the gateway.

When notify redirect is configured in dial-peer configuration mode, the configuration for the specific dial peer is activated only if the dial peer is an inbound dial peer. To enable notify redirect globally, use the **notify redirect (voice service)** command.



Note

Use the **notify redirect (dial peer)** command to configure Cisco SIP SRST 3.4 only after using the **allow-connections** command to enable B2BUA call flow on the SRST gateway.

Examples

The following is partial sample output from the **show running-config** command showing that notify redirect has been set up for IP-to-POTS calls on VoIP dial peer 8000:

```
dial-peer voice 8000 voip
destination-pattern 80..
notify redirect ip2pots
session protocol sipv2
session target ipv4:1.5.33.200
```



```
dtmf-relay rtp-nte
codec g711ulaw
!
```

Related Commands

Command	Description
dial peer	Enters dial peer configuration mode for defining a particular dial peer and specifying the method of voice encapsulation.
notify redirect (voice service)	Enables global IP-to-IP or IP-to-POTS notify redirection for all VoIP dial peers.

notify redirect (voice service)

To send a redirect facility to the application handling redirect requests for all VoIP dial peers on the Cisco IOS voice gateway, use the **notify redirect** command in the voice-service configuration mode. To return to the default, use the **no** form of this command.

notify redirect {ip2ip | ip2pots}

no notify redirect

Syntax Description

ip2ip	Sends redirect facility to the application handling redirect requests for IP-to-IP calls.
ip2pots	Sends redirect facility to the application handling redirect requests for IP-to-POTS calls.

Defaults

Disabled

Command Modes

Voice-service configuration

Command History

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

Usage Guidelines

Use this command to enable notify redirection globally on a gateway. Use the **notify redirect (dial-peer)** command to configure IP-to-IP or IP-to-POTS notify redirection on a specific inbound dial peer.



Note

Use the **notify redirect (voice service)** command to configure Cisco SIP SRST 3.4 only after using the **allow-connections** command to enable B2BUA call flow on the SRST gateway.

Examples

The following is partial sample output from the **show running-config** command showing that notify redirect has been set up globally for IP-to-POTS calling:

```
voice service voip
  notify redirect ip2pots
  allow-connections h323 to h323
  allow-connections h323 to sip
  allow-connections sip to sip
  no supplementary-service h450.2
  no supplementary-service h450.3
  sip
  registrar server expires max 600 min 60
!
```

Related Commands

Command	Description
voice service	Enters voice-service configuration mode.
notify redirect (dial peer)	Enables, on a per dial peer basis, IP-to-IP or IP-to-POTS notify redirection on the Cisco IOS voice gateway.

number (voice register pool)

To indicate the E.164 phone numbers that the registrar permits to handle the Register message from the Cisco 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**]

no number *tag* *number-pattern*

Syntax Description

<i>tag</i>	Number from 1 to 10 and the distinguishing factor when there are multiple number commands.
<i>number-pattern</i>	Phone numbers (including wildcards and patterns) that are permitted by the registrar to handle the Register message from the Cisco SIP IP phone.
preference value	(Optional) Defines the number list preference order. Range is from 0 to 10. The highest preference is 0. There is no default.
huntstop	(Optional) Stops hunting if the dial peer is busy.

Defaults

None (see the syntax description for syntax-level defaults)

Command Modes

Voice register pool configuration

Command History

Cisco IOS Release	Version	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

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

Examples

The following is partial sample output from the **show running-config** command showing the **number** command set to the telephone number pattern 95. Thus, all 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 SIP IP phone, or when running Cisco SIP SRST, set of Cisco SIP IP phones.
voice register dn	Enters voice register dn configuration mode to define an extension for a SIP phone line.
voice register pool	Enters voice register pool configuration mode for SIP phones.

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 directory number. Range is 0 to 10. Default is 0, which is the highest preference.

Defaults

0 (highest preference order)

Command Modes

Voice register pool configuration

Command History

Cisco IOS Release	Version	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 you create a voice register pool for a SIP phone or a group of SIP phones in a Cisco CallManager Express (Cisco CME) or Cisco SRST environment, you automatically create a virtual voice port and one to four virtual dial peers to be used by the number associated with that pool. This command sets a preference value for the number that you are creating. 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 telephone 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 sets 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 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 SIP IP phone registers with a SIP 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

<i>ip-address</i>	IP address of the SIP proxy.
preference <i>value</i>	(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. <ul style="list-style-type: none"> icmp-ping—Enables monitoring of proxy dial peers using ICMP ping. rtr—Enables monitoring of proxy dial peers using RTR probes. <i>alternate-ip-address</i>—(Optional) Enables monitoring of alternate IP addresses other than the proxy address. For example, to monitor a gateway front end to a SIP proxy.

Defaults

Proxy dial peer is disabled.

Command Modes

Voice register pool configuration

Command History

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

Usage Guidelines

The **proxy** command autogenerates additional VoIP dial peers to reach the main proxy whenever a Cisco SIP IP phone registers with a 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 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.

Defaults

Disabled

Command Modes

SIP configuration

Command History

Cisco IOS Release	Version	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 SIP SRST or Cisco 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 CME or Cisco 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 CME or Cisco SIP SRST environment.
	voice register pool	Enters voice register pool configuration mode for SIP phones.

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

Examples The following is sample output from this command:

```
Router# show sip-ua status registrar

Line          destination      expires(sec)    contact
              call-id
              peer
=====
2800          10.0.2.50       29              10.0.2.50
              00127f3b-90250002-23b0fa96-5900e864@10.0.2.50
              40003
1000          10.0.2.52       372             10.0.2.52
              00127f57-60aa0002-23a74903-48d80f79@10.0.2.52
              40001
2801          10.0.2.51       31              10.0.2.51
              00127f57-628f0002-23b08297-5f020ea1@10.0.2.51
              40002
```

Table 3 describes significant fields shown in this output.

Table 3 *show sip-ua status registrar* Field Descriptions

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

Related Commands

Command	Description
registrar server	Enables SIP registrar functionality.

show voice register all

To display all Cisco SIP SRST and Cisco 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	Version	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.

Examples

Cisco SIP SRST

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

```
Router# show voice register all

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 : 47
Registration success : 47
Registration failed : 0
unRegister requests : 45
unRegister success : 45
unRegister failed : 0
```

Cisco CME

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

```
Router# show voice register all
```

```
VOICE REGISTER GLOBAL
=====
CONFIG [Version=4.0(0)]
=====
Version 4.0(0)
Mode is cme
Max-pool is 24
Max-dn is 72
Source-address is 172.18.202.243 port 5060
Load ata ATA030200SIP041111A.zup
Load 7960-40 is POS3-07-4-00
Time-format is 12
Date-format is YY-M-D
Time-zone is 5
Hold-alert is enabled
Mwi stutter is enabled
Mwi registration for full E.164 is enabled
Forwarding local is enabled
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
Voicemail number is 7788
Max redirect number is 20
Telnet Level: 2
Tftp path is system:/cme/sipphone
Generate text file is enabled
Tftp files are created, current syncinfo 0002917733516824
OS79XX.TXT is not created

VOICE REGISTER DN
=====
Dn Tag 1
Config:
Number is 7001
Preference is 0
Huntstop is disabled
Name christoper robert
Auto answer is disabled
Label is jennifer nicole
Dn Tag 2
Config:
Number is 7002
Preference is 0
Huntstop is disabled
Name Jenny
Auto answer is disabled
Dn Tag 3
Config:
Number is 7003
```

■ show voice register all

```

Preference is 0
Huntstop is disabled
Name nino
Auto answer is disabled
Dn Tag 4
Config:
Number is 7004
Preference is 0
Huntstop is disabled
Auto answer is disabled
Dn Tag 5
Config:
Number is 7005
Preference is 0
Huntstop is disabled
Name ABBY
Auto answer is disabled
Dn Tag 6
Config:
Number is 7006
Preference is 0
Huntstop is disabled
Name jayce
Auto answer is disabled
MWI registration is enabled.
Dn Tag 7
Config:
Number is 7007
Preference is 0
Huntstop is disabled
Name bugs
Auto answer is enabled
Label is daffy
Dn Tag 8
Config:
Number is 7008
Preference is 0
Huntstop is disabled
Name Bob
Auto answer is disabled

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
Temp Tag 2
Config:
Attended Transfer is enabled
Blind Transfer is enabled
Semi-attended Transfer is enabled
Conference is disabled
Caller-ID block is disabled
DnD control is enabled
Anonymous call block is disabled
Voicemail is 7788, timeout 5
Temp Tag 3
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
Temp Tag 5
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
```

```
VOICE REGISTER POOL
=====
Pool Tag 1
Config:
Mac address is 000D.ED22.EDFE
Type is 7960
Number list 1 : DN 1
Proxy Ip address is 0.0.0.0
Default preference is 1
DTMF Relay is disabled
Call Waiting is disabled
DnD is disabled
keep-conference is enabled
template is 1
```

Dialpeers created:

```
Statistics:
Active registrations : 0
```

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

```
Pool Tag 2
Config:
Mac address is 000D.ED23.CBA0
Type is 7960
Number list 1 : DN 2
Number list 2 : DN 2
Proxy Ip address is 0.0.0.0
Default preference is 1
DTMF Relay is enabled, rtp-nte
Call Waiting is enabled
DnD is disabled
speed-dial 3 7001
speed-dial 4 7701
keep-conference is enabled
template is 1
```

Dialpeers created:

■ show voice register all

```
dial-peer voice 40003 voip
<-----
-----
destination-pattern 7002
redirect ip2ip
session target ipv4:172.18.202.251:5060
session protocol sipv2
dtmf-relay rtp-nte
after-hours-exempt FALSE
```

```
Statistics:
Active registrations : 2
```

```
Total Registration Statistics
Registration requests : 2
Registration success : 2
Registration failed : 0
unRegister requests : 0
unRegister success : 0
unRegister failed : 0
```

```
Pool Tag 3
Config:
Mac address is 0030.94C3.035E
Type is 7960
Number list 1 : DN 3
Number list 3 : DN 3
Proxy Ip address is 0.0.0.0
DTMF Relay is disabled
Call Waiting is enabled
DnD is disabled
keep-conference is enabled
template is 2
```

```
Dialpeers created:
```

```
Statistics:
Active registrations : 0
```

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

```
Pool Tag 5
Config:
Mac address is 0012.019B.3FD8
Type is ATA
Number list 1 : DN 5
Proxy Ip address is 0.0.0.0
Default preference is 1
DTMF Relay is disabled
Call Waiting is enabled
DnD is disabled
keep-conference is enabled
```

```
Dialpeers created:
```

```
Statistics:
```

```
Active registrations : 0
```

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

```
Pool Tag 6
Config:
Mac address is 0012.019B.3E88
Type is ATA
Number list 1 : DN 6
Number list 2 : DN 7
Proxy Ip address is 0.0.0.0
Default preference is 1
DTMF Relay is enabled, rtp-nte
Call Waiting is enabled
DnD is disabled
call-forward b2bua all 7788
keep-conference is enabled
template is 2
```

```
Dialpeers created:
```

```
dial-peer voice 40001 voip
<-----
-----
destination-pattern 7006
redirect ip2ip
session target ipv4:172.18.202.32:5060
session protocol sipv2
dtmf-relay rtp-nte
call-fwd-all 7788
after-hours-exempt FALSE

dial-peer voice 40002 voip
destination-pattern 7007
redirect ip2ip
session target ipv4:172.18.202.32:5060
session protocol sipv2
dtmf-relay rtp-nte
call-fwd-all 7788
after-hours-exempt FALSE
```

```
Statistics:
Active registrations : 2
```

```
Total Registration Statistics
Registration requests : 2
Registration success : 2
Registration failed : 0
unRegister requests : 0
unRegister success : 0
unRegister failed : 0
```

```
Nothing configured yet
Pool Tag 8
Config:
Mac address is 0006.D737.CC42
```

■ show voice register all

```
Type is 7940
Number list 1 : DN 8
Proxy Ip address is 0.0.0.0
Default preference is 1
DTMF Relay is disabled
Call Waiting is enabled
DnD is disabled
keep-conference is enabled
template is 5

Dialpeers created:

Statistics:
Active registrations : 0

Total Registration Statistics
Registration requests : 0
Registration success : 0
Registration failed : 0
unRegister requests : 0
unRegister success : 0
unRegister failed : 0

Pool Tag 9
Config:
Mac address is 0030.94C3.0831
Proxy Ip address is 0.0.0.0
DTMF Relay is disabled
Call Waiting is enabled
DnD is disabled
keep-conference is enabled

Dialpeers created:

Statistics:
Active registrations : 0

Total Registration Statistics
Registration requests : 0
Registration success : 0
Registration failed : 0
unRegister requests : 0
unRegister success : 0
unRegister failed : 0

Pool Tag 10
Config:
Mac address is 000D.ED22.EDFE
Proxy Ip address is 0.0.0.0
DTMF Relay is disabled
Call Waiting is disabled
DnD is disabled
call-forward b2bua all 1234
keep-conference is enabled

Dialpeers created:

Statistics:
Active registrations : 0

Total Registration Statistics
Registration requests : 0
```

```

Registration success : 0
Registration failed : 0
unRegister requests : 0
unRegister success : 0
unRegister failed : 0

```

Nothing configured yet

Table 4 describes significant fields shown in this output.

Table 4 *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.
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.

Table 4 *show voice register all Field Descriptions (continued)*

Field	Description
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 all	Displays all Cisco SIP SRST and Cisco CME configurations and register information.
show voice register dial-peer	Displays details of all dynamically created VoIP dial peers associated with the Cisco SIP SRST or Cisco 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 SIP SRST or Cisco CME register event, use the **show voice register dial-peers** command in privileged EXEC mode.

show voice register dial-peers

Syntax Description This command has no arguments or keywords.

Command Modes Privileged EXEC

Command History	Cisco IOS Release	Version	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.

Examples

Cisco SIP SRST

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

```
Router# show voice register dial-peers
```

```
dial-peer voice 40024 voip
corlist incoming call191
preference 5
destination-pattern 91011
redirect ip2ip
session target ipv4:192.168.0.2
session protocol sipv2
translate-outgoing called 1
voice-class codec 1
```

```
dial-peer voice 40025 voip
destination-pattern 40891011
redirect ip2ip
session target ipv4:192.168.0.2
session protocol sipv2
translate-outgoing called 1
voice-class codec 1
```

```
dial-peer voice 40026 voip
preference 8
destination-pattern 94...
redirect ip2ip
session target ipv4:192.168.0.2
session protocol sipv2
translate-outgoing called 1
voice-class codec 1
```

```
dial-peer voice 40027 voip
```

show voice register dial-peers

```

preference 1
destination-pattern 91011
redirect ip2ip
session target ipv4:10.2.161.187
session protocol sipv2
voice-class codec 1
monitor probe icmp-ping 10.2.161.187

```

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

Examples

Cisco CME

The following is sample output from this command:

```
Router# show voice register global
CONFIG [Version=3.4(0)]
=====
Version 3.4(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 POS3-07-4-07
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
Forwarding local is enabled
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
Telnet Level: 2
Tftp path is system:/cme/sipphone
Generate text file is disabled
Tftp files are created, current syncinfo 0002830590524159
OS79XX.TXT is not created
Router#
```

Cisco SIP SRST

```
Router# show voice register global

CONFIG [Version=3.4(0)]
=====
```

show voice register global

```
Version 3.4(0)
Mode is SIP SRST
Max-pool is 10
Max-dn is 10
```

Table 5 contains descriptions of significant fields shown in this output, listed in alphabetical order.

Table 5 *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 SIP CME or Cisco 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 SIP SRST or Cisco 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 CME or Cisco SIP SRST.
MWI registration	Setting of mwj command.
MWI stutter	Setting of mwj 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 SIP SRST or Cisco CME version number.

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 SIP SRST and Cisco CME configurations and register information.
show voice register dial-peer	Displays details of all dynamically created VoIP dial peers associated with the Cisco SIP SRST or Cisco CME register event.
voice register global	Enters voice register global configuration mode in order to set global parameters for all supported Cisco SIP phones in a Cisco CME or Cisco SIP SRST environment.

show voice register pool

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

show voice register pool *pool-tag*

Syntax Description	<i>pool-tag</i>	Tag number of the voice register pool for which to display information. The maximum number of pools is version and platform dependent; refer to Cisco IOS command-line interface (CLI) help.
---------------------------	-----------------	--

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

Command History	Cisco IOS Release	Version	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	This command was introduced.

Examples

Cisco CME

The following is sample output from this command displaying information for voice register pool 33:

```
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
  Number list 4 : DN 4
  Number list 5 : DN 5
  Number list 6 : DN 6
  Proxy Ip address is 0.0.0.0
  DTMF Relay is disabled
  Call Waiting is enabled
  keep-conference is enabled
  template is 1
```

Cisco SIP SRST

The following is sample output from this command displaying all information for voice register pool 1:

```
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
```

■ **show voice register pool**

```
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
```

Table 6 contains descriptions of significant fields shown in the Cisco SIP SRST and Cisco CME output, listed in alphabetical order.

Table 6 *show voice register pool Field Descriptions*

Field	Description
Active registrations	Used with the all , pool , and statistics keywords. Shows the current active registrations.
Application	Used with the all and pool keywords. Shows the application command configuration for this pool.
Call Waiting	Setting of call-waiting command.
Config:	Used with the all and pool keywords. Shows the voice register pool.
Class of Restriction List Tag	Used with the all and pool keywords. Shows the COR tag.
Default preference	Used with the all and pool keywords. Shows the default preference value of 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.
DnD	Setting of dnd-control command.
DTMF Relay	Setting of dtmf-relay command.
Incoming called number	Used with the all and pool keywords. Shows the incoming called-number command configuration.

Table 6 *show voice register pool Field Descriptions (continued)*

Field	Description
Incoming corlist name	Used with the all and pool keywords. Shows the cor command configuration.
keep-conference	Status of keep-conference command.
Mac address	MAC address of this SIP phone as defined by using the id command.
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 (voice register pool) command configuration.
Pool Tag	Used with the all and pool keywords. Shows the assigned tag number of the current pool.
Proxy Ip address	Used with the all and pool keywords. Shows the proxy command configuration; that is, the IP address of external SIP server.
Registration failed	Used with the all , pool , and statistics keywords. Shows the failed registrations.
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.
Statistics:	Used with the all , pool , and statistics keywords. Shows the registration statistics for this pool.
Template	Template-tag number for template that is applied to this SIP phone.
Translate outgoing called tag	Used with the all and pool keywords. Shows the translate-outgoing command configuration.
Total Registration Statistics	Used with the all , pool , and statistics keywords. Shows the total registration statistics for this pool.
Type	Phone type identified for this SIP phone using the type command.
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.
Username Password	Values within authentication credential.

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 SIP SRST and Cisco CME configurations and register information.

Command	Description
show voice register dial-peer	Displays details of all dynamically created VoIP dial peers associated with the Cisco SIP SRST or Cisco CME register event.
voice register pool	Enters voice register pool configuration mode for SIP phones.

show voice register statistics

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

show voice register statistics

Syntax Description This command has no arguments or keywords.

Command Modes Privileged EXEC

Command History	Cisco IOS Release	Version	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 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.

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

```
Router# show voice register statistics

Global statistics
Active registrations : 3
Total Registration Statistics
Registration requests : 7
Registration success : 4
Registration failed : 3
unRegister requests : 1
unRegister success : 1
unRegister failed : 0
Register pool 1 statistics
Active registrations : 1
Total Registration Statistics
Registration requests : 3
Registration success : 2
Registration failed : 1
unRegister requests : 1
unRegister success : 1
unRegister failed : 0
Register pool 2 statistics
Active registrations : 2
```

show voice register statistics

```
Total Registration Statistics
Registration requests : 2
Registration success : 2
Registration failed : 0
unRegister requests : 0
unRegister success : 0
unRegister failed : 0
```

Table 7 describes significant fields shown in this output.

Table 7 *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.
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 <i>number</i> statistics	Used with the statistics keyword. Details specific pool statistics.

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 SIP SRST and Cisco CME configurations and register information.
show voice register dial-peers	Displays details of all dynamically created VoIP dial peers associated with the Cisco SIP SRST or Cisco CME register event.
show voice register pool	Displays all configuration information associated with a particular voice register pool.

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.
<i>rule-tag</i>	The <i>rule-tag</i> is an arbitrarily chosen number by which the rule set is referenced. The range is from 1 to 2147483.

Defaults

None

Command Modes

Voice register pool configuration

Command History

Cisco IOS Release	Version	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

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 SIP SRST or Cisco 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.



Note

Translation rules must be set by using the **translate-outgoing** command before the **alias** command is configured in Cisco 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 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 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 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.

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.

Defaults Enabled

Command Modes Voice register pool configuration

Command History	Cisco IOS Release	Version	Modification
	12.4(4)T	Cisco CME 3.4 and 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 disables 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.

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	<i>tag</i>	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 (dial peer) command.
---------------------------	------------	---

Command Default	None
------------------------	------

Command Modes	Voice register pool configuration
----------------------	-----------------------------------

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

Usage Guidelines During Cisco SIP SRST or Cisco 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 SIP SRST or a Cisco CME environment.
	id (voice register pool)	Explicitly identifies a locally available individual Cisco SIP IP phone, or when running Cisco 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.

voice register global

To enter voice register global configuration mode in order to set global parameters for all supported Cisco SIP phones in a Cisco CME or Cisco SIP SRST environment, use the **voice register global** command in global configuration mode. To remove the configuration, use the **no** form of this command.

voice register global

no voice register global

Syntax Description This command has no arguments or keywords.

Defaults No default behavior or values

Command Modes Global configuration

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

Usage Guidelines

Cisco CME
Use this command to set provisioning parameters for all supported SIP phones in a Cisco CME system.

Cisco 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.

Examples

Cisco 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 POS3-07-4-07
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
```

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.
mode (voice register global)	Enables the mode for provisioning SIP phones in a Cisco CallManager Express (Cisco CME) system.
voice register dn	Enters voice register dn configuration mode to define an extension for a SIP phone line.
voice register pool	Enters voice register pool configuration mode for Cisco SIP IP phones.
voice register template	Enters voice register template configuration mode and defines a template of common parameters for SIP phones.

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	<i>pool-tag</i>	Unique number assigned to the pool. Range is 1 to 100.
	Note	For Cisco CME systems, the upper limit for this argument is defined by the max-pool command.

Defaults No default behavior or values

Command Modes Global configuration

Command History	Cisco IOS Release	Version	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

Cisco CME

Use this command to set phone-specific parameters for SIP phones in a Cisco CallManager Express (Cisco 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 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.

Examples

Cisco CME

The following example enters voice register pool configuration mode and forwards 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 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
```

Related Commands

Command	Description
max-pool (voice register global)	Sets the maximum number of SIP phones that are supported by a Cisco CallManager Express (Cisco CME) system.
mode (voice register global)	Enables the mode for provisioning SIP phones in a Cisco CME system.
number (voice register pool)	Configures a valid number for a SIP phone.
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 CME or Cisco SIP SRST environment.



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