

Cisco Unified Survivable Remote Site Telephony 8.1 New Features

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This document describes the following new and enhanced features in Cisco Unified Survivable Remote Site Telephony 8.1 (Cisco Unified SRST):

- Toll Fraud Prevention Enhancement, page 2
- Enhancements to SIP Phone Configuration, page 4

Finding Feature Information

Your software release may not support all the features documented in this module. For the latest feature information and caveats, see the release notes for your platform and software release. To find information about the features documented in this module, and to see a list of the releases in which each feature is supported, see the "Feature Information for Cisco Unified SRST 8.1" section on page 116.

Use Cisco Feature Navigator to find information about platform support and Cisco IOS, Catalyst OS, and Cisco IOS XE software image support. To access Cisco Feature Navigator, go to http://www.cisco.com/go/cfn. An account on Cisco.com is not required.

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Prerequisites for Cisco Unified SRST 8.1

- Cisco Unified SRST 8.1
- Cisco IOS Release 15.1(2)T

Information About Cisco Unified SRST 8.1

To configure Cisco Unified SRST features, you should understand the following concepts:

- Toll Fraud Prevention Enhancement, page 2
- Enhancements to SIP Phone Configuration, page 4

Toll Fraud Prevention Enhancement

Cisco Unified SRST8.1 enhances the Toll Fraud Prevention feature to secure the Cisco Unified SRST system against potential toll fraud exploitation by unauthorized users. The following are the enhancements to Toll Fraud Prevention in Cisco Unified SRST:

- IP Address Trusted Authentication
- Direct Inward Dial for Incoming ISDN Calls
- Disconnecting ISDN Calls With no Matching Dial-peer
- Blocking Two-stage Dialing Service on Analog and Digital FXO Ports

IP Address Trusted Authentication

IP address trusted authentication process blocks unauthorized calls and helps secure the Cisco Unified SRST system against potential toll fraud exploitation by unauthorized users. In Cisco Unified SRST, **IP address trusted authentication** is enabled by default. When IP address trusted authenticate is enabled, Cisco Unified CME accepts incoming VoIP (SIP/H.323) calls only if the remote IP address of an incoming VoIP call is successfully validated from the system **IP address trusted list**. If the IP address trusted authentication fails, an incoming VoIP call is then disconnected by the application with a user- defined cause code and a new application internal error code 31 message (TOLL_FRAUD_CALL_BLOCK) is logged. For more information, see the, "Configuring IP Address Trusted Authentication for Incoming VoIP Calls" section on page 7.

Cisco Unified SRST maintains an **IP address trusted list** to validate the remote IP addresses of incoming VOIP calls. Cisco Unified SRST saves an IPv4 session target of VoIP dial-peer to add the trusted IP addresses to **IP address trusted list** automatically. The IPv4 session target is identified as a trusted IP address only if the status of VoIP dial-peer in operation is "UP". Up to 10050 IPv4 addresses can be defined in the trusted IP address list. No duplicate IP addresses are allowed in the trusted IP

address list. You can manually add up to 100 trusted IP addresses for incoming VOIP calls. For more information on manually adding trusted IP addresses, see the, "Adding Valid IP Addresses For Incoming VoIP Calls" section on page 10.

A call detail record (CDR) history record is generated when the call is blocked as a result of IP address trusted authentication failure. A new voice Internal Error Code (IEC) is saved to the CDR history record. The voice IEC error messages are logged to syslog if "voice iec syslog" option is enabled. The following is an IEC toll fraud call rejected syslog display:

*Aug 14 19:54:32.507: %VOICE_IEC-3-GW: Application Framework Core: Internal Error (Toll fraud call rejected): IEC=1.1.228.3.31.0 on callID 3 GUID=AE5066C5883E11DE8026A96657501A09

The **IP** address trusted list authentication must be suspended when Cisco Unified SRST is defined with "gateway" and a VoIP dial-peer with "session-target ras" is in operational UP status. The incoming VOIP call routing is then controlled by the gatekeeper. Table 1 shows administration state and operational state in different trigger conditions.

Table 1 Administration and Operation States of IP Address Trusted Authentication

Trigger Condition	Administration State	Operation State
When ip address trusted authenticate is enabled.	Down	Down
When "gateway" is defined and a VoIP dial-peer with "ras" as a session target is in "UP" operational state	Up	Down
When ip address trusted authenticate is enabled and either "gateway" is not defined or no voip dial-peer with "ras" as session target is in "UP" operational state	Up	Up



We recommend enabling SIP authentication before enabling Out-of-dialog REFER (OOD-R) to avoid any potential toll fraud threats.

Direct Inward Dial for Incoming ISDN Calls

In Cisco Unified SRST 8.1 and later versions the **direct-inward-dial isdn** feature in enabled to prevent the toll fraud for incoming ISDN calls. The called number of an incoming ISDN enbloc dialing call is used to match the outbound dial-peers even if the **direct-inward-dial** option is disabled from a selected inbound plain old telephone service (POTS) dial-peer. If no outbound dial-peer is selected for the outgoing call set up, the incoming ISDN call is disconnected with cause-code "unassigned-number (1)". For more information on direct-inward dial for incoming ISDN calls, see the, "Configuring Direct Inward Dial for Incoming ISDN Calls" section on page 12.

Disconnecting ISDN Calls With no Matching Dial-peer

Cisco Unified SRST 8.1 and later versions disconnect unauthorized ISDN calls when no matching inbound voice dial-peer is selected. Cisco Unified SRST and voice gateways use the **dial-peer no-match disconnect-cause** command to disconnect an incoming ISDN call when no inbound dial-peer is selected to avoid default POTS dial-peer behavior including two-stage dialing service to handle the incoming ISDN call.

Blocking Two-stage Dialing Service on Analog and Digital FXO Ports

Cisco Unified SRST 8.1 and later versions block the two-stage dialing service which is initiated when an Analog or Digital FXO port goes offhook and the private line automatic ringdown (PLAR) connection is not setup from the voice-port. As a result, no outbound dial-peer is selected for an incoming analog or digital FXO call and no dialed digits are collected from an FXO call. Cisco Unified SRST and voice gateways disconnect the FXO call with cause-code "unassigned-number (1)". Cisco Unified SRST uses the **no secondary dialtone** command by default from FXO voice-port to block the two-stage dialing service on Analog or digital FXO ports. For more information on blocking two-stage dialing service on Analog and Digital FXO port, see Blocking Secondary Dialtone on Analog and Digital FXO Ports, page 14.

Enhancements to SIP Phone Configuration

Cisco Unified SRST 8.1 and later versions, allows you to verify SIP phone registration process, remove global registration parameters, and display details on phones that attempted to register with Cisco Unified SRST and fail. Following are the enhancements to SIP Phone configuration:

- Removing Global Registration Parameters for SIP Phones, page 4
- Verifying SIP Phone Registration, page 5

Removing Global Registration Parameters for SIP Phones

In Cisco Unified SRST 8.1 and later versions, you can use the **no voice register global** command to automatically remove existing DNs and pools. The **no voice register global** command also removes the voice register dialplan, template, and session-server configurations when the system is in Cisco Unified CME mode. For more information, see the "Command Reference" section on page 19

Verifying SIP Phone Registration

In Cisco Unified CME 8.1 and later versions, you can verify SIP phone registration details and other information - related to SIP phones. Table 2 shows a list of new show commands and the information that these commands provide. For detailed information on these show commands, see the "Command Reference" section on page 19.

Table 2 New commands and display information

Commands	Display Information
attempted-registrations size	Allows you to set the size of the table that stores information related to SIP phones that attempted to register and failed.
clear voice register attempted-registrations	Allows you to clear all the entries in attempted-registration table.
show voice register dn all	Displays the details of all the DNs defined in the system.
show voice register dial-peers pool	Displays the dialpeers a pool with a specific tag.
show voice register dialplan all	Displays the details of all the Dialplans that are defined in the system.
show voice register pool after-hour-exempt	Displays the details of the phones with "after-hour exempt" set.
show voice register pool attempted-registrations	Displays the details of the phones that have attempted to register and failed.
show voice register pool connected	Displays detailed information of phones that are in connected state (in conversation).
show voice register pool connected brief	Displays a brief information of phones that are in connected state (in conversation).
show voice register pool ip	Displays the details of a phone with a specific IPv4 address.
show voice register pool mac	Displays the details of a phone with a specific mac address.
show voice register pool network	Displays the information of the pools with a specific network address configured.
show voice register pool on-hold	Displays detailed information of phones that are currently on-hold.
show voice register pool on-hold brief	Displays brief information of phones that are currently on-hold.
show voice register pool phone-load	Displays the phone-load details of registered phones.
show voice register pool registered	Displays the details of the phones that are successfully registered.
show voice register pool remote	Displays remote phones (phones with no Address Resolution Protocol (ARP) entry.
show voice register pool ringing	Displays the detailed information of the phones that are currently ringing.
show voice register pool ringing brief	Displays brief information of phones that are currently ringing.

Table 2 New commands and display information

Commands	Display Information
show voice register pool telephone-number	Displays the details of the phone with a specified telephone-number.
show voice register pool unregistered	Displays the details of the pools that do not have any registered phones.
show voice register statistics global	Displays the global statistics of registered SIP phones.
show voice register statistics pool	Displays the pool statistics associated with a specific pool.

How to Configure Cisco Unified SRST 8.1 New Features

This section contains the following tasks.

- Configuring IP Address Trusted Authentication for Incoming VoIP Calls, page 7
- Adding Valid IP Addresses For Incoming VoIP Calls, page 10
- Configuring Direct Inward Dial for Incoming ISDN Calls, page 12
- Blocking Secondary Dialtone on Analog and Digital FXO Ports, page 14
- Troubleshooting Tips for Toll Fraud Prevention, page 15

Configuring IP Address Trusted Authentication for Incoming VolP Calls

Prerequisites

Cisco Unified SRST 8.1 or a later version.

Restrictions

- IP address trusted authentication is skipped if an incoming SIP call is originated from a SIP phone.
- IP address trusted authentication is skipped if an incoming call is an IPv6 call.
- For an incoming VoIP call, IP trusted authentication must be invoked when the IP address trusted authentication is in "UP" operational state.

SUMMARY STEPS

- 1. enable
- 2. configure terminal
- 3. voice service voip
- 4. ip address trusted authenticate
- 5. ip-address trusted call-block cause < code>
- 6. end
- 7. show ip address trusted list

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable	Enables privileged EXEC mode.
	<pre>Example: Router> enable</pre>	Enter your password if prompted.
Step 2	configure terminal	Enters global configuration mode.
	Example: Router# configure terminal	
Step 3	voice service voip	Enters voice service voip configuration mode.
	<pre>Example: Router(config) # voice service voip</pre>	
Step 4	ip address trusted authenticate	Enables IP address authentication on incoming H.323 or SIP trunk calls for toll fraud prevention support.
	<pre>Example: Router(conf-voi-serv)# ip address trusted authenticate</pre>	IP address trusted list authenticate is enabled by default. Use the "no ip address trusted list authenticate" command to disable the IP address trusted list authentication.
Step 5	ip-address trusted call-block cause code	Issues a cause-code when the incoming call is rejected to the IP address trusted authentication.
	<pre>Example: Router(conf-voi-serv)#ip address trusted call-block cause call-reject</pre>	Note If the IP address trusted authentication fails, a call-reject (21) cause-code is issued to disconnect the incoming VoIP call.
Step 6	end	Returns to privileged EXEC mode.
	<pre>Example: Router()# end</pre>	
Step 7	show ip address trusted list	Verifies a list of valid IP addresses for incoming H.323 or SIP trunk calls, Call Block cause for rejected incoming
	Example:	calls.
	Router# #show ip address trusted list IP Address Trusted Authentication Administration State: UP Operation State: UP	
	<pre>IP Address Trusted Call Block Cause: call-reject (21)</pre>	

Examples

Router #show ip address trusted list

```
IP Address Trusted Authentication
Administration State: UP
Operation State:
IP Address Trusted Call Block Cause: call-reject (21)
VoIP Dial-peer IPv4 Session Targets:
Peer Tag
         Oper State Session Target
_____
              _____
                             -----
11
              DOWN
                             ipv4:1.3.45.1
             UP
                             ipv4:1.3.45.1
IP Address Trusted List:
ipv4 172.19.245.1
ipv4 172.19.247.1
ipv4 172.19.243.1
ipv4 171.19.245.1
ipv4 172.19.245.0 255.255.255.0''
```

Adding Valid IP Addresses For Incoming VolP Calls

Prerequisites

• Cisco Unified SRST 8.1 or a later version.

SUMMARY STEPS

- 1. enable
- 2. configure terminal
- 3. voice service voip
- 4. ip address trusted list
- 5. ipv4 ipv4 address network mask
- 6. end
- 7. show ip address trusted list

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable	Enables privileged EXEC mode.
		Enter your password if prompted.
	Example:	
	Router> enable	
Step 2	configure terminal	Enters global configuration mode.
	Example: Router# configure terminal	
Step 3	voice service voip	Enters voice service voip configuration mode.
	<pre>Example: Router(config)# voice service voip</pre>	
Step 4	ip address trusted list	Enters ip address trusted list mode and allows to manually add additional valid IP addresses.
	<pre>Example: Router(conf-voi-serv)# ip address trusted list Router(cfg-iptrust-list)#</pre>	
Step 5	<pre>ipv4 {<ipv4 address=""> [<network mask="">]}</network></ipv4></pre>	Allows you to add up to 100 IPv4 addresses in ip address trusted list . Duplicate IP addresses are not allowed in the
	Example:	ip address trusted list.
	Router(config) #voice service voip Router(conf-voi-serv) #ip taddress trusted list Router(cfg-iptrust-list) #ipv4 172.19.245.1 Router(cfg-iptrust-list) #ipv4 172.19.243.1	• (Optional) <i>network mask</i> — allows to define a subnet IP address.

	Command or Action	Purpose
Step 6	end	Returns to privileged EXEC mode.
	<pre>Example: Router(config-register-pool)# end</pre>	
Step 7	show ip address trusted list	Displays a list of valid IP addresses for incoming H.323 or SIP trunk calls.
	Example: Router# show shared-line	

Examples

The following example shows 4 IP addresses configured as trusted IP addresses:

```
Router#show ip address trusted list
IP Address Trusted Authentication
Administration State: UP
Operation State: UP
```

IP Address Trusted Call Block Cause: call-reject (21)

VoIP Dial-peer IPv4 Session Targets:

Peer Tag	Oper State	Session Target
11	DOWN	ipv4:1.3.45.1
1	UP	ipv4:1.3.45.1

```
IP Address Trusted List:

ipv4 172.19.245.1

ipv4 172.19.247.1

ipv4 172.19.243.1

ipv4 171.19.245.1

ipv4 171.19.10.1
```

Configuring Direct Inward Dial for Incoming ISDN Calls

To configure Direct Inward Dial for incoming ISDN calls, perform the following steps:

Restrictions

• Direct-inward-dial isdn is not supported for incoming ISDN overlap dialing call.

SUMMARY STEPS

- 1. enable
- 2. configure terminal
- 3. voice service pots
- 4. direct-inward-dial isdn
- 5. end

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable	Enables privileged EXEC mode.
		• Enter your password if prompted.
	Example:	
	Router> enable	
Step 2	configure terminal	Enters global configuration mode.
	Example: Router# configure terminal	
Step 3	voice service pots	Enters voice service configuration mode with voice telephone-service encapsulation type (pots).
	<pre>Example: Router(config) # voice service pots Router(conf-voi-serv) #</pre>	
Step 4	<pre>direct-inward-dial isdn Example: Router(conf-voi-serv)#direct-inward-dial isdn</pre>	Enables direct-inward-dial (DID) for incoming ISDN number. The incoming ISDN (enbloc dialing) call is treated as if the digits were received from the DID trunk. The called number is used to select the outgoing dial peer. No dial tone is presented to the caller.
Step 5	exit	Exits voice service pots configuration mode.
	Example: Router(conf-voi-serv)# exit	

Examples

```
voice service voip
ip address trusted list
ipv4 172.19.245.1
ipv4 172.19.247.1
 ipv4 172.19.243.1
 ipv4 171.19.245.1
ipv4 171.19.10.1
 allow-connections h323 to h323
allow-connections h323 to sip
allow-connections sip to h323
 allow-connections sip to sip
 supplementary-service media-renegotiate
sip
registrar server expires max 120 min 120
dial-peer voice 1 voip
destination-pattern 5511...
session protocol sipv2
session target ipv4:1.3.45.1
incoming called-number 5522...
direct-inward-dial
dtmf-relay sip-notify
codec g711ulaw
dial-peer voice 100 pots
destination-pattern 91...
incoming called-number 2...
 forward-digits 4
```

Blocking Secondary Dialtone on Analog and Digital FXO Ports

To block secondary dialtone on Analog and Digital FXO port, perform the following steps:

SUMMARY STEPS

- 1. enable
- 2. configure terminal
- 3. voice-port
- 4. no secondary dialtone
- 5. exit

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable	Enables privileged EXEC mode.
		• Enter your password if prompted.
	Example:	
	Router> enable	
Step 2	configure terminal	Enters global configuration mode.
	Example: Router# configure terminal	
Step 3	voice-port	Enters voice-port configuration mode.
		Type your Analog or Digital FXO port number.
	<pre>Example: Router(config) #voice-p 2/0/0</pre>	
Step 4	no secondary dialtone	Blocks the secondary dialtone on Analong and Digital FXO port.
	Example:	
	Router((config-voiceport)# no secondary	
	dialtone	
Step 5	end	Returns to privileged EXEC mode.
	Example:	
	Router(conf-voiceport)# exit	
Step 6	show run	Verifies that the secondary dialtone is disabled on the specific voice-port.
	Example: Router# show run sec voice-port 2/0/0	

Examples

```
Router# conf t
Router(config) #voice-p 2/0/0
Router(config-voiceport) # no secondary dialtone
!
end

Router# show run | sec voice-port 2/0/0
Foreign Exchange Office 2/0/0 Slot is 2, Sub-unit is 0, Port is 0
Type of VoicePort is FXO
Operation State is DORMANT
Administrative State is UP
...
Secondary dialtone is disabled
```

Troubleshooting Tips for Toll Fraud Prevention

When incoming VOIP call is rejected by IP address trusted authentication, a specific internal error code (IEC) **1.1.228.3.31.0** is saved to the call history record. You can monitor the failed or rejected calls using the IEC support. Follow these steps to monitor any rejected calls:

Step 1 Use the **show voice iec description** command to find the text description of an IEC code.

```
Router# show voice iec description 1.1.228.3.31.0

IEC Version: 1

Entity: 1 (Gateway)

Category: 228 (User is denied access to this service)

Subsystem: 3 (Application Framework Core)

Error: 31 (Toll fraud call rejected)

Diagnostic Code: 0
```

Step 2 View the IEC statistics information using the **Enable iec statistics** command. The example below shows that 2 calls were rejected due to toll fraud call reject error code.

Example:

```
Router# Enable iec statistics
Router(config) #voice statistics type iec
Router#show voice statistics iec since-reboot
Internal Error Code counters
------
Counters since reboot:
SUBSYSTEM Application Framework Core [subsystem code 3]
[errcode 31] Toll fraud call rejected 2
```

Step 3 Use the **enable IEC syslog** command to verify the syslog message logged when a call with IEC error is released.

Example:

```
Router# Enable iec syslog
Router (config) #voice iec syslog

Feb 11 01:42:57.371: %VOICE_IEC-3-GW: Application Framework Core:
Internal Error (Toll fraud call rejected): IEC=1.1.228.3.31.0 on
callID 288 GUID=DB3F10AC619711DCA7618593A790099E
```

Step 4 Verify the source address of an incoming VOIP call using the show call history voice last command.

Example:

```
Router# show call history voice last 1

GENERIC:
SetupTime=3306550 ms
Index=6
...
InternalErrorCode=1.1.228.3.31.0
...
RemoteMediaIPAddress=1.5.14.13
...
```

Step 5 IEC is saved to VSA of Radius Accounting Stop records. Monitor the rejected calls using the external RADIUS server.

Example:

```
Feb 11 01:44:06.527: RADIUS: Cisco AVpair [1] 36 "internal-error-code=1.1.228.3.31.0"
```

Step 6 Retrieve the IEC details from cCallHistoryIec MIB object. More information on IEC is available at: ttp://www.cisco.com/en/US/docs/ios/voice/monitor/configuration/guide/vt_voip_err_cds_ps6350_TSD _Products_Configuration_Guide_Chapter.html

Example:

```
getmany 1.5.14.10 cCallHistoryIec
cCallHistoryIec.6.1 = 1.1.228.3.31.0
>getmany 172.19.156.132 cCallHistory
cCallHistorySetupTime.6 = 815385
cCallHistoryPeerAddress.6 = 1300
cCallHistoryPeerSubAddress.6 =
cCallHistoryPeerId.6 = 8000
cCallHistoryPeerIfIndex.6 = 76
cCallHistoryLogicalIfIndex.6 = 0
cCallHistoryDisconnectCause.6 = 15
cCallHistoryDisconnectText.6 = call rejected (21)
cCallHistoryConnectTime.6 = 0
cCallHistoryDisconnectTime.6 = 815387
cCallHistoryCallOrigin.6 = answer(2)
cCallHistoryChargedUnits.6 = 0
cCallHistoryInfoType.6 = speech(2)
cCallHistoryTransmitPackets.6 = 0
cCallHistoryTransmitBytes.6 = 0
cCallHistoryReceivePackets.6 = 0
cCallHistoryReceiveBytes.6 = 0
cCallHistoryReleaseSrc.6 = internalCallControlApp(7)
cCallHistoryIec.6.1 = 1.1.228.3.31.0
>getone 172.19.156.132 cvVoIPCallHistoryRemMediaIPAddr.6
cvVoIPCallHistoryRemMediaIPAddr.6 = 1.5.14.13
```

Additional References

The following sections provide references related to Cisco Unified SRST.

Related Documents

Related Topic	Document Title
Cisco Unified CME configuration	Cisco Unified Communications Manager Express System Administrator Guide
	• Cisco Unified Communications Manager Express Command Reference
Cisco Unified CME network design	Cisco Unified CallManager Express Solution Reference Network Design Guide
Cisco IOS voice configuration	Cisco IOS Voice Configuration Library
	Cisco IOS Voice Command Reference
Phone documentation for Cisco Unified CME	User Documentation for Cisco Unified IP Phones

Standards

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

MIBs

MIB	MIBs Link
No new or modified MIBs are supported by this feature, and support for existing MIBs has not been modified by this feature.	To locate and download MIBs for selected platforms, Cisco IOS releases, and feature sets, use Cisco MIB Locator found at the following URL:
	http://www.cisco.com/go/mibs

RFCs

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

Technical Assistance

Description	Link
The Cisco Support website provides extensive online resources, including documentation and tools for troubleshooting and resolving technical issues with Cisco products and technologies.	http://www.cisco.com/techsupport
To receive security and technical information about your products, you can subscribe to various services, such as the Product Alert Tool (accessed from Field Notices), the Cisco Technical Services Newsletter, and Really Simple Syndication (RSS) Feeds.	
Access to most tools on the Cisco Support website requires a Cisco.com user ID and password.	

Command Reference

The following commands are introduced or modified in the features documented in this module.

New and Modified Commands

- attempted-registrations size, page 20
- clear voice register attempted-registrations, page 23
- dial-peer no-match isdn disconnect-cause, page 26
- direct-inward-dial isdn, page 28
- ip address trusted call-block cause, page 27
- ip address trusted authenticate, page 33
- ip address trusted list, page 36
- secondary dialtone (voice port), page 39
- show voice register all, page 41
- show voice register dial-peers, page 51
- show voice register dn, page 55
- show voice register global, page 59
- show voice register pool, page 63
- show voice register pool mac, page 74
- show voice register pool ip, page 76
- show voice register pool network, page 78
- show voice register pool telephone-number, page 79
- show voice register pool after-hour-exempt, page 81
- show voice register pool attempted-registrations, page 84
- show voice register pool connected, page 87
- show voice register pool on-hold, page 91
- show voice register pool phone-load, page 95
- show voice register pool registered, page 97
- show voice register pool remote, page 100
- show voice register pool ringing, page 102
- show voice register pool unregistered, page 105
- show voice register statistics, page 108
- voice register global, page 113

attempted-registrations size

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

attempted-registrations size size

no attempted-registrations size size

ntax		

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

Command Default

The default size for attempted registration table is 10.

Command Modes

voice register global

Command History

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

Usage Guidelines

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

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

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

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

Examples

The following example shows attempted-registrations size:

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

Related Commands

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

clear voice register attempted-registrations

To clear the attempted-registrations, use the **clear voice register attempted-registrations** command in voice register global mode.

clear voice register attempted registrations [ip ip-address | mac H.H.H.]

Syntax Description

ip ip-address	(Optional) IP address of the SIP phone attempting to register.
mac H.H.H	(Optional) MAC address of the SIP phone attempting to register.

Command Default

The attempted-registration entries are not cleared.

Command Modes

Privileged EXEC.

Command History

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

Usage Guidelines

Use this command to delete the entries in the attempted-registration table. The **clear voice register attempted-registrations** command does not alter the table size, but clears the existing entries. A user confirmation is sought before the cleanup is done.

The primary key to recognize the SIP phones that fail to register is through their MAC address (hardware address) and the secondary key is the IP address. You can clear the attempted registration entry for a specific phone that failed to register by providing its IP address or MAC address and create more space for new attempted registration entries in the attempted-registrations table. When no options (IP or MAC) are selected, all the entries are removed. A user confirmation is sought in such a case, before clearing the attempted-registrations table.

The **ip** keyword allows you to delete entries corresponding to a specific IP address. Similarly, the **mac** keyword allows you to clear the entries related to a specific MAC address. User confirmation is not sought if **ip** or **mac** option is used.

Examples

Router #clear voice regis attempted-registrations
This will clear all the entries. Proceed? Yes/No? [no]: Yes

Router#clear voice register attempted-registrations ?

ip IP Address of the phone
mac MAC Address of the phone

Related Commands

Command	Description
attempted-registrations size	Allows to set the size of the attempted-registrations table.
show voice register attempted-registration	Displays details of phones that attempted to register and failed.

dial-peer no-match isdn disconnect-cause

To disconnect the incoming ISDN call when no inbound voice dial peer is matched, use the dial-peer no-match disconnect-cause command in global configuration mode. To restore the default incoming call handling behavior, use the **no** form of this command.

dial-peer no-match isdn disconnect-cause cause-code

no dial-peer no-match isdn disconnect-cause cause-code

ntax		

cause-code An ISDN cause code number. Range is from 1 to 188.

Command Default

Dial-peer no-match isdn disconnect-cause command is disabled. Incoming ISDN calls are not forced to disconnect if no inbound dial-peer is matched

Command Modes

Global configuration

Command History

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

Usage Guidelines

Use this command to disconnect unathorized ISDN calls when no inbound voice or modem dial peer is matched.

Refer to the ISDN Cause Values table in the *Cisco IOS Debug Command Reference*, for a list of ISDN cause codes.

Examples

The following example shows that ISDN cause code 28 has been specified to match inbound voice or modem dial peers:

Router# dial-peer no-match disconnect-cause 28

Related Commands

Command	Description
show dial-peer voice	Displays configuration information for dial peers.

direct-inward-dial isdn

To enable incoming ISDN enbloc dialing calls, use the **direct-inward-dial isdn** command in voice service voip mode. To disable incoming ISDN enbloc dialing calls use the **no** form of the command.

direct-inward-dial isdn

no direct-inward-dial isdn

Syntax Description

This command has no arguments or keywords.

Command Default

The direct inward dial isdn is command is enabled.

Command Modes

voice service pots

Command History

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

Usage Guidelines

Use the direct-inward-dial-isdn command to enable the direct-inward-dial (DID) call treatment for an incoming ISDN call. When this feature is enabled, the incoming ISDN call is treated as if the digits were received from the DID trunk. The called number is used to select the outgoing dial peer. No dial tone is presented to the caller to collect dialed digits even if "no direct-inward-dial" of the selected inbound dial-peer is defined for an incoming ISDN call.

Use the no form of this command to turn off the global direct-inward-dial setting for incoming ISDN calls. When this command line is disabled, the "direct-inward-dial" setting of a selected inbound dial-peer is used to handle the incoming ISDN calls.'

Examples

The following is sample output from this command displaying DID enabled for ISDN:

```
voice service voip
 ip address trusted list
 ipv4 172.19.245.1
  ipv4 172.19.247.1
  ipv4 172.19.243.1
 ipv4 171.19.245.1
  ipv4 171.19.10.1
 allow-connections h323 to h323
 allow-connections h323 to sip
 allow-connections sip to h323
 allow-connections sip to sip
 supplementary-service media-renegotiate
  registrar server expires max 120 min 120
dial-peer voice 1 voip
destination-pattern 5511...
 session protocol sipv2
 session target ipv4:1.3.45.1
 incoming called-number 5522...
direct-inward-dial
!
```

Related Commands

Command	Description Enters voice service configuration mode.	
voice service		

ip address trusted call-block cause

To issues a cause-code when the incoming call is rejected by the IP address trusted authentication, use the **ip address trusted call-block cause** command in voice service voip mode. To stop the IP address trusted authentication process from sending a call-block cause, use the **no** form of this command.

ip-address trusted call-block cause code-id

no ip-address trusted call-block cause code-id

SyntaDescription

code-id

Q.850 call-disconnect cause code. Range is from 1 to 127.

Command Default

A call-reject (21) cause-code is issued to disconnect the incoming VoIP calls.

Command Modes

Voice Service voip

Command History

Cisco IOS Release	Cisco Product	Modification
15.(1)T	Cisco Unified CME 8.1	This command was introduced.

Usage Guidelines

Use this command to issue a cause-code when the incoming call is rejected by the IP address trusted authentication. You can issue a specific call-block cause code using any one of the Q.850 call reject cause codes.

Examples

The following is sample output from this command displaying the default call block cause code:

Router #show ip address trusted list

IP Address Trusted Authentication Administration State: UP Operation State: UP

IP Address Trusted Call Block Cause: call-reject (21)

VoIP Dial-peer IPv4 Session Targets:

Related Commands

Command	Description
ip address trusted list	Allows to manually add additional valid IP addresses.
ip address trusted authenticate	Enables IP address trusted authentication for incoming VoIP calls.

ip address trusted authenticate

To enable ip address trusted authentication for incoming VoIP (H.323/SIP) calls, use the **ip address trusted authenticate** command in voice service voip mode. To disable ip address trusted authentication, use the **no** form of this command.

ip address trusted authenticate

no ip address trusted authenticate

Syntax Description

This command has no arguments or keywords.

Command Default

IP address trusted list authenticate is enabled.

Command Modes

Voice Service Voip

ipv4 171.19.10.1

Command History

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

Usage Guidelines

Use this command to enable the ip address trusted authentication for incoming H.323 or SIP trunk calls for toll fraud prevention on Cisco Unified CME.

Examples

The following is sample output from this command displaying IP address trusted authentication enabled for incoming calls:

```
IP Address Trusted Authentication
Administration State: UP
 Operation State:
IP Address Trusted Call Block Cause: call-reject (21)
VoIP Dial-peer IPv4 Session Targets:
Peer Tag
               Oper State
                            Session Target
11
                               ipv4:1.3.45.1
               DOWN
                               ipv4:1.3.45.1
IP Address Trusted List:
 ipv4 172.19.245.1
 ipv4 172.19.247.1
 ipv4 172.19.243.1
 ipv4 171.19.245.1
```

Related Commands	Command Description	
	ip address trusted list	Allows to manually add additional valid IP addresses.
	ip address trusted call- block cause	Allows to issues a cause-code when the incoming call is rejected by the IP address trusted authentication.

ip address trusted list

To manually add multiple IP addresses for incoming VoIP (H.323/SIP) calls, use the **ip address trusted list** command in voice service voip mode. To turn off the list, use the **no** form of this command.

ip address trusted list {ipv4 <ipv4 address> < network mask>}

no ip address trusted list {ipv4 <*ipv4 address*> < *network mask*>}

Synta Description

ipv4-address	IPv4 address of the incoming H.323 or SIP calls.
network mask	Subnet IP address.

Command Default

IP address trusted list is disabled.

Command Modes

Voice Service Voip

Command History

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

Usage Guidelines

Use this command to manually add unique and multiple IP addresses to a list of trusted IP addresses. You can add up to 100 IPv4 addresses in the **ip address trusted list**. No duplicate IP addresses are allowed.

Examples

The following is sample output from this command displaying a list of trusted IP addresses:

```
Router #show ip address trusted list
IP Address Trusted Authentication
Administration State: UP
Operation State:
IP Address Trusted Call Block Cause: call-reject (21)
VoIP Dial-peer IPv4 Session Targets:
Peer Tag
                Oper State
                                Session Target
11
                DOWN
                                ipv4:1.3.45.1
                                ipv4:1.3.45.1
1
                IJΡ
IP Address Trusted List:
 ipv4 172.19.245.1
 ipv4 172.19.247.1
 ipv4 172.19.243.1
 ipv4 171.19.245.1
 ipv4 171.19.10.1
```

Related Commands

Command	Description
IP address trusted authenticate	Enables IP address trusted authentication for incoming VoIP calls.
IP address trusted code-block cause	Allows to issues a cause-code when the incoming call is rejected by the IP address trusted authentication.

secondary dialtone (voice port)

To allow dialed digits to be collected from the remote switch when "connection plar" is not defined from the analog FXO voice-port, use the **secondary dialtone** command in global configuration mode. To disable the secondary dialtone, use the **no** form of the command.

secondary dialtone

no secondary dialtone

Syntax Description

This command has no arguments or keywords.

Command Default

The secondary dialtone command is disabled.

Command Modes

Global configuration

Command History

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

Usage Guidelines

Use the secondary dialtone command to allow dialed digits to be collected from the remote switch when "connection plar" is not defined from the analog FXO voice-port.

The following is sample output from this command:

Router(config)# voice-port 2/0/0
Router (config-voiceport)#no secondary ?
dialtone Secondary dialtone option for FXO port
Router (config-voiceport)#no secondary dialtone
"secondary dialtone" is used to enable 2-stage dialing for an incoming call

Related Commands

I

Command	Description
voice service	Enters voice service configuration mode.

show voice register all

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

show voice register all

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco SIP SRST 3.0	This command was introduced.
12.3(4)T	Cisco SIP SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.4(4)T	Cisco CME 3.4 and Cisco SIP SRST 3.4	This command was added to Cisco CME.
15.1(2)T	Cisco Unified CME 8.1 Cisco Unified SIP SRST 8.1	This command was modified. The output display was modified.

Examples

Cisco Unified SIP SRST

The following is 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 Unified 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 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 Anoymous 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

```
Anoymous 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
Anoymous 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
Anoymous 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
```

keep-conference is enabled

```
template is 1
Dialpeers created:
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
```

```
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
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 7 describes significant fields shown in this output.

Table 3 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.

Table 3 show voice register all Field Descriptions (continued)

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

Related Commands

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

show voice register dial-peers

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

show voice register dial-peers [pool tag]

Descripti	

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

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco SIP SRST 3.0	This command was introduced.
12.3(4)T	Cisco SIP SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.4(4)T	Cisco CME 3.4 Cisco SIP SRST 3.4	This command was added to Cisco CME.
15.1(2)T	Cisco CME 8.1 Cisco SIP SRST 8.1	This command was modified. Pool <i>tag</i> keyword and argument was added. Command output display was also modified to display dial-peers specific to a pool.

Usage Guidelines

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

Examples

Cisco Unified CME adn Cisco Unified SIP SRST

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

```
Router#show voice register dial-peers
Dial-peers for Pool 1
dial-peer voice 40001 voip
destination-pattern 45111
session target ipv4:8.3.3.111:5060
session protocol sipv2
 call-fwd-all
after-hours-exempt FALSE
dial-peer voice 40002 voip
  destination-pattern 45113
  session target ipv4:8.33.33.111:5060
  session protocol sipv2
 after-hours-exempt FALSE
Dial-peers for Pool 2
 dial-peer voice 40003 voip
destination-pattern 45112
session target ipv4:8.33.33.112:5060
session protocol sipv2
call-fwd-noan-timeou 8
 call-fwd-noan
after-hours-exempt TRUE
```

Cisco Unified CME and Cisco Unified SRST

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

```
Router# show voice register dial-peers pool 1
```

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

Related Commands

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

show voice register dn

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

show voice register dn tag [all]

Syntax Description

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

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

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

Examples

Cisco Unified SIP CME

The following is a sample output from this command:

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

Cisco Unified SIP SRST

The following is a sample output from this command:

```
Router# show voice register dn 2
Dn Tag 1
Config:
   Number is 11
   Preference is 10
   Huntstop is enabled
   Translation-profile incoming saaa
   Allow watch is enabled
   Pool 1 has this DN configured for line 1
```

Cisco Unified SIP SRST

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

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

Cisco Unified SIP CME

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

Router# show voice register dn all

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

```
Preference is 0
  Huntstop is disabled
  Auto answer is disabled
  call-forward b2bua all 87687
          has this DN configured for line 1
  Pool 3
         has this DN configured for line 1, 2
Dn Tag 4
Config:
 Auto answer is disabled
Dn Tag 7
Config:
 Number is 451110
 Preference is 0
 Huntstop is disabled
 Auto answer is disabled
 after-hour exempt
          has this DN configured for line 4
 Pool 1
Dn Tag 8
Config:
 Auto answer is disabled
 call-forward b2bua all 678
 after-hour exempt
  Pool 1 has this DN configured for line 3
```

Table 7 contains descriptions of significant fields shown in this output, listed in alphabetical order.

Table 4 show voice register dn Field Descriptions

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

Related Commands

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

show voice register global

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

show voice register global

Syntax Description

This command has no arguments or keywords.

Command Default

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification	
12.4(4)T	Cisco CME 3.4 Cisco SIP SRST 3.4	This command was introduced.	
15.1(2)T	Cisco Unified CME 8.1 Cisco Unified SIP SRST 8.1	This command was modified. The output display now includes global statistics.	

Examples

Cisco Unified 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 ${\tt Max}$ 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 Unified SIP SRST

Router# show voice register global

CONFIG [Version=3.4(0)]

-----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 Unified SIP CME or Cisco Unified SIP SRST router as configured with the max-dn command. The maximum possible number is platform-dependent.			
Max-pool	Reports the maximum number of SIP voice register pools supported by the Cisco Unified SIP SRST or Cisco Unified CME router as configured with the max-pool command. The maximum possible number is platform-dependent.			
Max redirect number	Maximum number of redirects set with the max-redirect command.			
Mode	Reports the mode as configured with the mode command. Value can be either Cisco Unified CME or Cisco Unified SIP SRST.			
MWI registration	Setting of mwi command.			
MWI stutter	Setting of mwi stutter command.			
Time-format Value of time-format command.				
Time-zone	Number of the timezone selected with the timezone command.			
TFTP path	Directory location of provisioning files for SIP phones that is specified with the tftp-path command.			
Version	Reports the Cisco Unified SIP SRST or Cisco Unified CME version number.			

Related Commands

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

Command	Description		
show voice register dial-peer	Displays details of all dynamically created VoIP dial peers associated with the Cisco Unified SIP SRST or Cisco Unified 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 Unified CME or Cisco Unified SIP SRST environment.		

show voice register pool

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

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

Syntax Description

brief	(Optional) Displays brief information for a specific voice register pool.
all	Displays information for all pools configured in a system.
tag	Tag number of the voice register pool for which to display information. The maximum number of pools is version and platform dependent. Type ? to display a list of values.

Command Modes

Privileged EXEC (#)

Command History

Cisco IOS Release	Cisco Product	Modification		
12.2(15)ZJ	Cisco SIP SRST	This command was introduced.		
12.3(4)T	Cisco SIP SRST	This command was integrated into Cisco IOS Release 12.3(4)T.		
12.4(4)T	Cisco CME 3.4 Cisco SIP SRST 3.4	This command was added to Cisco Unified CME.		
12.4(15)XY	Cisco Unified CME 4.2(1) Cisco Unified SIP SRST 4.2(1)	Emergency response location (ERL) information displays in the output.		
12.4(20)T	Cisco Unified CME 7.0 Cisco Unified SIP SRST 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.		
15.0(1)XA	Cisco Unified CME 8.0	This command was modified. Logical partitioning class of restriction (LPCOR) information was added to the output.		
15.1(2)T	Cisco Unified CME 8.1	This command was modified. The all and brief keywords were added. Voice-class stun-usage information is displayed in the output.		

Examples

I

Cisco Unified CME Example 1:

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

Router# show voice register pool 1

Pool Tag 1 Config: Mac address is 001B.535C.D410 Type is 7960 Number list 1 : DN 1 Number list 3 : DN 8 Number list 4 : DN 7

```
Proxy Ip address is 0.0.0.0
  Current Phone load version is Cisco-CP7960G/8.0
  DTMF Relay is disabled
  Call Waiting is enabled
  DnD is disabled
  Busy trigger per button value is 0
  call-forward phone all is 4566
  call-forward b2bua all 4555
  keep-conference is enabled
  Lpcor Type is none
  Transport type is udp
  service-control mechanism is supported
  registration Call ID is 001b535c-d410790d-17a6877e-5d04bbc5@8.3.3.111
  Privacy feature is not configured.
  Privacy button is disabled
  active primary line is: 45111
  contact IP address: 8.3.3.111 port 5060
Dialpeers created:
Dial-peers for Pool 1:
dial-peer voice 40001 voip
destination-pattern 45111
 session target ipv4:8.3.3.111:5060
 session protocol sipv2
  call-fwd-all 4555
  after-hours-exempt FALSE
Statistics:
  Active registrations : 1
  Total SIP phones registered: 1
  Total Registration Statistics
   Registration requests : 1
   Registration success : 1
   Registration failed
   unRegister requests
                          : 0
   unRegister success
                         : 0
   unRegister failed
                          : 0
   Attempts to register
          after last unregister: 0
   Last register request time : *11:40:32.263 UTC Wed Oct 14 2009
   Last unregister request time :
   Register success time : *11:40:32.267 UTC Wed Oct 14 2009
   Unregister success time
```

Cisco Unified CME Example 2:

The following is sample output from this command displaying information for all the voice register pools:

```
Router# show voice register pool all
Pool Tag 1
Config:
   Mac address is 001B.535C.D410
   Type is 7960
   Number list 1 : DN 3
   Number list 3 : DN 8
   Number list 4 : DN 7
```

```
Proxy Ip address is 0.0.0.0
  Current Phone load version is Cisco-CP7960G/8.0
  DTMF Relay is disabled
  Call Waiting is enabled
  DnD is disabled
  Busy trigger per button value is 0
  call-forward phone all is 4566
  call-forward b2bua all 4555
  keep-conference is enabled
  template is 3
  Lpcor Type is none
  Transport type is udp
  service-control mechanism is supported
  registration Call ID is 001b535c-d4100002-6d241f50-0fac9bc9@8.3.3.111
  Privacy feature is not configured.
  Privacy button is disabled
  active primary line is: 45111
  contact IP address: 8.3.3.111 port 5060
  Reason for unregistered state:
        No registration request since last reboot/unregister
Dialpeers created:
Dial-peers for Pool 1:
Statistics:
  Active registrations : 0
  Total SIP phones registered: 1
  Total Registration Statistics
    Registration requests : 2
    Registration success : 2
   Registration failed : 0
   unRegister requests : 1
    unRegister success
    unRegister failed
    Attempts to register
          after last unregister : 0
   Last register request time : *12:10:37.259 UTC Tue Oct 13 2009
   Last unregister request time : *11:56:22.179 UTC Tue Oct 13 2009
    Register success time
                               : *12:10:37.263 UTC Tue Oct 13 2009
   Unregister success time
                               : *11:56:22.182 UTC Tue Oct 13 2009
Pool Tag 2
Config:
  Mac address is 0015.C68E.6D13
  Type is 7960
  Number list 1 : DN 2
  Proxy Ip address is 0.0.0.0
  Current Phone load version is Cisco-CP7960G/8.0
  DTMF Relay is disabled
  Call Waiting is enabled
  DnD is disabled
  Busy trigger per button value is 0
  call-forward phone noan is 9886, timeout 98
  keep-conference is enabled
  username pool2 password lab
  Lpcor Type is none
  Transport type is udp
  service-control mechanism is supported
  registration Call ID is 0015c68e-6d130002-24aebd5d-43d8d548@8.33.33.112
```

```
Privacy feature is not configured.
 Privacy button is disabled
  active primary line is: 45112
 contact IP address: 8.33.33.112 port 5060
Dialpeers created:
Dial-peers for Pool 2:
dial-peer voice 40002 voip
destination-pattern 45112
session target ipv4:8.33.33.112:5060
session protocol sipv2
 call-fwd-noan-timeou 8
 call-fwd-noan 999
 after-hours-exempt TRUE
Statistics:
 Active registrations : 1
 Total SIP phones registered: 1
 Total Registration Statistics
   Registration requests : 1
   Registration success : 1
   Registration failed : 0
   unRegister requests : 0
   unRegister success : 0
   unRegister failed
   Attempts to register
          after last unregister: 0
   Last register request time : *11:56:21.407 UTC Tue Oct 13 2009
   Last unregister request time :
   Register success time : *11:56:21.411 UTC Tue Oct 13 2009
   Unregister success time
```

Cisco Unified CME Example 3:

The following is sample output from this command displaying brief information for the voice register pools:

Cisco Unified CME Example 4:

The following is sample output from this command displaying brief information for all the voice register pools:

Router# show voice register pool all brief

Pool	ID	IP Address	Ln	DN	Number	State
====	==========	==========	==	===	============	=========
1	001B.535C.D410	8.3.3.111	1	1	45111	REGISTERED
			4	7	451110	UNREGISTERED
2	0015.C68E.6D13		1	2	45112	UNREGISTERED
3	0021.5553.8998		1	3	45113	UNREGISTERED

			2	3	45113	UNREGISTERED
4	8989.9867.8769					UNREGISTERED
7	0018.BAC8.D2B1		1	2	45112	UNREGISTERED
10	9.13.18.40	9.13.18.40	1	1	1000	REGISTERED

Cisco Unified SIP SRST Example 1:

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:
 Mac address is 0019.06FC.A377
  Number list 1 : DN 1
 Number list 2 : DN 2
  Proxy Ip address is 0.0.0.0
  DTMF Relay is disabled
  kpml signal is enabled
  Lpcor Type is none
  Reason for unregistered state:
        No registration request since last reboot/unregister
Dialpeers created:
Statistics:
 Active registrations : 0
  Total SIP phones registered: 0
  Total Registration Statistics
   Registration requests : 0
   Registration success : 0
   Registration failed : 0
   unRegister requests : 0
   unRegister success : 0
                          : 0
   unRegister failed
    Attempts to register
          after last unregister : 0
   Last register request time
   Last unregister request time :
   Register success time
   Unregister success time
```

Cisco Unified SIP SRST Example 2:

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

```
Router# show voice register pool all
Pool Tag 1
Config:
  Ip address is 9.13.18.40, Mask is 255.255.0.0
  Number list 1 : DN 1
 Number list 2 : DN 2
  Number list 3 : DN 3
  Number list 4 : DN 4
  Number list 5 : DN 5
  Number list 6 : DN 6
  Number list 7 : DN 7
  Proxy Ip address is 0.0.0.0
  DTMF Relay is enabled, rtp-nte, sip-notify
  kpml signal is enabled
  Lpcor Type is none
Dialpeers created:
```

```
Dial-peers for Pool 1:
dial-peer voice 40004 voip
destination-pattern 1000
redirect ip2ip
session target ipv4:9.13.18.40:19633
 session protocol sipv2
dtmf-relay rtp-nte sip-notify
 digit collect kpml
 codec g711ulaw bytes 160
 after-hours-exempt FALSE
dial-peer voice 40001 voip
destination-pattern 2000
redirect ip2ip
session target ipv4:9.13.18.40:19634
 session protocol sipv2
 dtmf-relay rtp-nte sip-notify
digit collect kpml
 codec g711ulaw bytes 160
 after-hours-exempt FALSE
Statistics:
  Active registrations : 4
  Total SIP phones registered: 1
  Total Registration Statistics
    Registration requests : 4
   Registration success : 4
   Registration failed
                          : 0
   unRegister requests : 0
   unRegister success
   unRegister failed
                          : 0
   Attempts to register
          after last unregister : 0
   Last register request time : .05:22:55.604 UTC Tue Oct 6 2009
   Last unregister request time :
   Register success time : .05:22:55.604 UTC Tue Oct 6 2009
   Unregister success time
Pool Tag 2
Config:
 Mac address is 0019.06FC.A377
 Number list 1 : DN 1
 Number list 2 : DN 2
  Proxy Ip address is 0.0.0.0
  DTMF Relay is disabled
  kpml signal is enabled
 Lpcor Type is none
  Reason for unregistered state:
        No registration request since last reboot/unregister
Dialpeers created:
Statistics:
  Active registrations : 0
  Total SIP phones registered: 0
  Total Registration Statistics
   Registration requests : 0
    Registration success : 0
   Registration failed : 0
                          : 0
    unRegister requests
```

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

Cisco Unified SIP SRST Example 3:

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

Router# show voice register pool all brief Pool ID IP Address Ln DN Number 001B.535C.D410 8.3.3.111 1 1 45111 REGISTERED 3 8 UNREGISTERED 4 7 451110 UNREGISTERED 8.33.33.112 8.3.3.112 1 2 45112 8.3.0.0 8.3.3.116 1 3 45113 2 REGISTERED 3 REGISTERED

Cisco Unified SIP SRST Example 4:

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

Router# show voice register pool 2 brief

Pool	ID	IP Address	Ln	DN	Number	State
====	==========	==========	==	===	=======================================	=========
2	9.13.18.40	9.13.18.40	1	1	1000	REGISTERED
			2	2	2000	REGISTERED
			3	3	3000	REGISTERED

Voice class stun usage Example 5:

The following is sample output from this command displaying voice-class stun-usage information for voice register pool 51:

Router# show voice register pool 51

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

```
Transport type is udp
  service-control mechanism is not supported
  registration Call ID is 2BA38EE3-17D311DB-800BCD81-A9AD11F0
  Privacy feature is not configured.
  Privacy button is disabled
  active primary line is: 16263646
  contact IP address: 192.168.0.87 port 5060
  Reason for unregistered state:
        No registration request since last reboot/unregister
  voice-class stun-usage is enabled. tag is 1
Dialpeers created:
Dial-peers for Pool 51:
Statistics:
  Active registrations : 0
 Total SIP phones registered: 0
  Total Registration Statistics
   Registration requests : 2
   Registration success : 2
   Registration failed : 0
   unRegister requests : 2
   unRegister success : 2
   unRegister failed
   Attempts to register
          after last unregister: 0
   Last register request time : 13:43:27.839 IST Tue Apr 20 2010
```

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

Table 6 show voice register pool Field Descriptions

Field	Description
Active registrations	Shows the current active registrations. Used with the all , pool , and statistics keywords.
Application	Shows the application command configuration for this pool. Used with the all and pool keywords.
Current phone-load	Shows the current version of the phone load.
Call Waiting	Setting of call-waiting command.
Config:	Shows the voice register pool. Used with the all and pool keywords.
Class of Restriction List Tag	Shows the COR tag. Used with the all and pool keywords.
Default preference	Shows the default preference value of this pool. Used with the all and pool keywords.
Dialpeers created:	Results in a list of all dial peers created and their contents. Dial-peer contents differ per application and are not described here. Used with the all and pool keywords.
DnD	Setting of dnd-control command.
DTMF Relay	Setting of dtmf-relay command.

Table 6 show voice register pool Field Descriptions (continued)

Field	Description
Emergency response location	The ephone's emergency response location to which an emergency response team is dispatched when an emergency call is made.
Incoming called number	Shows the incoming called-number command configuration. Used with the all and pool keywords.
Incoming corlist name	Shows the cor command configuration. Used with the all and pool keywords.
keep-conference	Status of keep-conference command.
Lpcor Incoming	Setting of the lpcor incoming command.
Lpcor Outgoing	Setting of the lpcor outgoing command.
Lpcor Type	Setting of the lpcor type command.
Mac address	MAC address of this SIP phone as defined by using the id command.
Network address and Mask	Shows network address and mask information if the id command is configured. Used with the all and pool keywords.
Number list, Pattern, and Preference	Shows the number (voice register pool) command configuration. Used with the all and pool keywords.
Pool Tag	Shows the assigned tag number of the current pool. Used with the all and pool keywords.
Previous phone-load	Shows the version of the previous phone-load.
Proxy IP address	Shows the proxy command configuration; that is, the IP address of the external SIP server. Used with the all and pool keywords.
Registration failed	Shows the failed registrations. Used with the all , pool , and statistics keywords.
Registration requests	Shows the incoming registration requests. Used with the all , pool , and statistics keywords.
Registration success	Shows the successful registrations. Used with the all , pool , and statistics keywords.
Statistics:	Shows the registration statistics for this pool. Used with the all , pool , and statistics keywords.
statistics time-stamps	Shows the registration statistics for this pool with specific time-stamps.
Template	Template-tag number for template that is applied to this SIP phone.
Translate outgoing called tag	Shows the translate-outgoing command configuration. Used with the all and pool keywords.
Total Registration Statistics	Shows the total registration statistics for this pool. Used with the all , pool , and statistics keywords.
Туре	Phone type identified for this SIP phone using the type command.
unRegister requests	Shows the incoming unregister/registration expiry requests. Used with the all , pool , and statistics keywords.
unRegister success	Reports the number of successful unregisters. Used with the all , pool , and statistics keywords.

Table 6 show voice register pool Field Descriptions (continued)

Field	Description
unRegister failed	Reports the number of failed unregisters. Used with the all , pool , and statistics keywords.
Username Password	Values within the authentication credential.
Voice class stun usage	Displays the configured voice-class stun-usage information.

Related Commands

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

show voice register pool mac

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

show voice register pool mac H.H.H

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H.H.H	MAC address of	of the SIP	phone attem	pting to	register.

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

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

Examples

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

Router# show voice register pool mac

Pool	ID	IP Address	Ln	DN	Number	State
====	==========	==========	==	===	=======================================	=========
1	001B.535C.D410	8.3.3.111	1	1	45111	REGISTERED
			4	7	451110	UNREGISTERED

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

show voice register pool ip

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

show voice register pool ip ip-address

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->V/11	IAKJESO		116311
O 9 11		<i>-</i> 1110	uvii

ip-address	IPv4 address of the incoming H.323 or SIP calls.
.,	

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

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

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

Examples

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

Router#	show	voice	register	pool	ip

Pool	ID	IP Address	Ln	DN	Number	State
====	==========	==========	==	===	=======================================	=========
1	001B.535C.D410	8.3.3.111	1	1	45111	REGISTERED
			4	7	451110	UNREGISTERED

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

show voice register pool network

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

show voice register pool network

0 1	\mathbf{r}	. , .
Vinta	VIACO	rintion
DVIII	かしろし	ription

	ID 4 11 C.1 : II 222 CID 11
ip-address	IPv4 address of the incoming H.323 or SIP calls.

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

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

Examples

I

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

Pool	ID	IP Address	Ln	DN	Number	State
====	===========	============	==	===	=======================================	=========
7	78.89.0.0		1	1	6576	UNREGISTERED

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

show voice register pool telephone-number

To display the details of voice register pool of a phone that has DND enabled, use the **show voice** register pool telephone-number command in privileged EXEC mode.

show voice register pool telephone-number number

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Synt	aDesc	crip	tıon

number	Number identifying a	specific phone.

Command Modes

Privileged EXECv

Command History

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

Usage Guidelines

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

Examples

I

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

Router#	show	voice	register	pool	telephone	number	980965	
---------	------	-------	----------	------	-----------	--------	--------	--

Pool	ID	IP Address	Ln	DN	Number	State
====	==========	==========	==	===	=======================================	=========
2	0015.C68E.6D13		1	2	45112	UNREGISTERED
7	0018.BAC8.D2B1		1	2	45112	UNREGISTERED

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

show voice register pool after-hour-exempt

To display the details of a phone that has after-hour-exempt enabled on it, use the **show voice register after-hour-exempt** command in privileged EXEC mode.

show voice register after-hour-exempt

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

Use this command to display the details of a phone that has "after-hour-exempt" enabled. Individual phones can be exempted from call blocking using the after-hour exempt.

Examples

The following is a sample output from this command displaying information for two phones with after after-hour-exempt:

Router#	show	voice	register	pool	after-hour-exempt
---------	------	-------	----------	------	-------------------

Pool	ID	IP Address	Ln	DN	Number	State
====	==========	==========	==	===	=======================================	========
1	001B.535C.D410	8.3.3.111	3	8		UNREGISTERED
			4	7	451110	UNREGISTERED
2	0015.C68E.6D13		1	2	45112	UNREGISTERED
3	0021.5553.8998		1	3	45113	UNREGISTERED
			2	3	45113	UNREGISTERED
7	0018.BAC8.D2B1		1	2	45112	UNREGISTERED

Table 7 contains descriptions of significant fields shown in this output, listed in alphabetical order.

Table 7 show voice register pool after-hour exempt field descriptions

Field	Description
IP Address/port	IP add and port number of the phones
MAC	MAC address of the phone.
Number	Number of the phones that have after-hour exempt enabled.
Pool	Shows the current pool.
State	Registration state.

Command	Description
after hour exempt	Specifies that an IP phone does not have any of its outgoing calls blocked although call blocking is defined.
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.
voice register pool	Enters voice register pool configuration mode for SIP phones.

show voice register pool attempted-registrations

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

show voice register pool attempted-registration

Synta Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

Use this command to display the details of the phones that attempt to register with Cisco Unified CME or Cisco Unified SRST and fail. The details of registration requests from the phones which cannot be mapped to the existing pools on the system are used to display information about the phones that have attempted to register with to the Cisco Unified CME or Cisco Unified SRST. If the phone registers successfully after some time, the attempted registration entry will still show up in the attempted-registration table. Use the **clear voice register attempted-registrations** command to remove the entry from the attempted registration table.

Examples

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

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

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

show voice register pool connected

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

show voice register pool connected [brief]

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

	4
Command Modes	Privileged EXEC

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

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

Cisco Unified CME and Cisco Unified SRST

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

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

Stream Media Addr Type : 1

Negotiated Codec : g729r8 (20 bytes)

Codec Payload Type : 18
    Negotiated Dtmf-relay : inband-voice
    Dtmf-relay Payload Type : 0
    OoS ID
                         : -1
    Local QoS Strength
                             : BestEffort
    Negotiated QoS Strength : BestEffort
    Negotiated QoS Direction: None
    Local QoS Status : None
    Media Source IP Addr:Port: [8.3.3.5]:17580
    Media Dest IP Addr:Port : [8.3.3.111]:26298
Options-Ping
             ENABLED:NO ACTIVE:NO
Inbound calls to SIP line phones:
Pool tag: 2
==========
MAC Address : 0015.C68E.6D13
Contact IP : 8.33.33.112
Phone Number : 45112
Remote Number : 45111
  Call ID : 4DA52F97-ADA311DE-8019803A-FF3E4CBC08.3.3.5
State of the call : STATE_ACTIVE (7)
Call 3
SIP Call ID
  Substate of the call : SUBSTATE_NONE (0)
                         : 45111
  Calling Number
  Called Number
                         : 45112
```

: 0xC04018 0x100 0x80 Bit Flags CC Call ID : 8 Source IP Address (Sig): 8.3.3.5 Destn SIP Req Addr:Port : [8.33.33.112]:5060 Destn SIP Resp Addr:Port: [8.33.33.112]:5060 Destination Name : 8.33.33.112 Number of Media Streams : 1Number of Active Streams: 1 RTP Fork Object : 0x0Media Mode : flow-through Media Stream 1 State of the stream : STREAM_ACTIVE
Stream Call ID : 8 State or the 1 Stream Call ID : 8 : voice-only (0) Stream Media Addr Type : 1 Negotiated Codec : g729r8 (20 bytes) Dtmf-relay Payload Type : 0 : -1 QoS ID Local QoS Strength : BestEffort Negotiated QoS Strength : BestEffort Negotiated QoS Direction : None Local QoS Status : None Media Source IP Addr:Port: [8.3.3.5]:16384 Media Dest IP Addr:Port : [8.33.33.112]:30040

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

Router# show voice register pool connected brief

Pool	IP Address	Number	Remote Number
====	==========	=======================================	=======================================
1	8.3.3.111	45111	45112
Inbound calls to SIP line phones:			
Pool	IP Address	Number	Remote Number
====	===========	=======================================	=======================================
2	8.33.33.112	45112	45111

Related Commands

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

show voice register pool on-hold

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

show voice register pool on-hold [brief]

Syntax	Dac	orir	tion	
Svillax	Desi	CTIL	111011	

brief	(Optional) Displays brief detail	ils of SIP phones that ar	e currently on-hold.
-------	----------------------------------	---------------------------	----------------------

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

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

Examples

Cisco Unified CME and Cisco Unified SRST

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

Router# show voice register pool on-hold brief

Outbound calls from SIP line phones:

Pool	IP Address	Number	Remote Number	Hold Origin
====	==========	=======================================	=======================================	==========
1	8.3.3.111	45111	45112	Remote & Local

Inbound calls to SIP line phones:

Pool	IP Address	Number	Remote Number	Hold Origin
====	==========	=======================================	=======================================	==========
2	8.33.33.112	45112	45111	Remote & Local

Cisco Unified CME and Cisco Unified SRST

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

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

: 0xC04018 0x4100 0x80 Bit Flags CC Call ID : 8 Source IP Address (Sig): 8.3.3.5 Destn SIP Req Addr:Port : [8.33.33.112]:5060 Destn SIP Resp Addr:Port: [8.33.33.112]:5060 Destination Name : 8.33.33.112 Number of Media Streams : 1Number of Active Streams: 1 RTP Fork Object : 0x0 Media Mode : flow-through Media Stream 1 State of the stream : STREAM_ACTIVE
Stream Call ID : 8 Stream Call ID : 8 : voice-only (0) Stream Media Addr Type : 1 Negotiated Codec : g729r8 (20 bytes) codec Payload Type : 18
Negotiated Dtmf-relay : inband-voice
Dtmf-relay Paylor 1 Dtmf-relay Payload Type : 0 : -1 QoS ID Local QoS Strength : BestEffort Negotiated QoS Strength : BestEffort Negotiated QoS Direction : None Local QoS Status : None Media Source IP Addr:Port: [8.3.3.5]:16384 Media Dest IP Addr:Port : [8.33.33.112]:30040 Options-Ping ENABLED:NO ACTIVE:NO

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

show voice register pool phone-load

To display the details of phone-loads associated with phones that are registered to Cisco Unified CME, use the **show voice register pool phone-load** command in privileged EXEC mode.

show voice register pool phone-load

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

Use this command to display the details of the phone-loads associated with phones that are registered with Cisco Unified CME. The phone-load information is extracted from the REGISTER message sent by the phone and is stored as a part of the pool details.

Examples

The following is sample output from this command displaying information for voice register pool phone-load:

Router# show voice register pool phone-load

Phone-load:

Pool	Type	Current	Previous
1	7960	8.5.2	8.5.1
2	7965	8.4.0	_

Command	Description	
show voice register all Displays all Cisco SIP SRST and Cisco CME configurations 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.	
voice register pool	Enters voice register pool configuration mode for SIP phones.	

show voice register pool registered

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

show voice register pool registered

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

Use this command to display the details of phones that are successfully registered to Cisco Unified CME and Cisco Unified SRST.

Examples

The following is sample output from this command displaying information for voice register dn 148:

Router# show voice register pool registered

Pool Tag 3 Config:

> Mac address is 0018.BAC8.D2B1 Type is 7960

Number list 1 : DN 2 Proxy Ip address is 0.0.0.0

DTMF Relay is disabled

Call Waiting is enabled

DnD is disabled

keep-conference is enabled

service-control mechanism is supported

registration Call ID is 0018bac8-d2b10002-51402c47-52001f25@8.3.3.113

active primary line is: 0394

contact IP address: 8.3.3.113 port 5060

Command	Description
show voice register all	Displays all Cisco SIP SRST and Cisco CME configurations and register information.
show voice register pool	Displays all configuration information associated with a particular voice register pool.
show voice register pool unregistered	Displays the details of voice register pools that do not have any phones registered.

Command	Description	
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.	

show voice register pool remote

To display the details of phones are at a remote location, use the **show voice register pool remote** command in privileged EXEC mode.

show voice register pool remote

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

Use this command to display the details of the phones that are at remote location and do not have an ARP entry.

Examples

The following is sample output from this command displaying information for remote phones:

Router# show voice register pool remote

Remote	Phones:				
Pool	MAC	IP Address/port	Contact IP	State	Phone numbers
1	0015.C66B.ED65	73.3.114/ 5060	73.4.4.1	Registered	79879 78798
2	0018.BAC8.D2B	173.3.3.113/5060	73.4.4.4	Unregistered	76569

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

show voice register pool ringing

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

show voice register pool ringing [brief]

	Desc	

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

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

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

Examples

Cisco Unified CME and Cisco Unified SRST

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

Router# show voice register pool ringing brief

Pool	IP Address	Number	Remote Number
====	==========	=======================================	=======================================
2	8 33 33 112	15112	15111

Cisco Unified CME and Cisco Unified SRST

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

```
Router# show voice register pool ringing
Pool tag: 2
=========
MAC Address
                : 0015.C68E.6D13
                : 8.33.33.112
Contact IP
Phone Number : 45112
Remote Number : 45111
Call 1
  P Call ID : C0B5DA7-ADA311DE-8011803A-FF3E4CBC@8.3.3.5
State of the call : STATE_RECD_PROCEEDING (4)
Substate of the call : SUBSTATE_PROCEEDING_PROCEEDING (2)
SIP Call ID
   Calling Number
                           : 45111
   Called Number
                           : 45112
   Bit Flags
                           : 0xC00018 0x100 0x280
   CC Call ID
                           . 5
   Source IP Address (Sig ): 8.3.3.5
   Destn SIP Req Addr:Port : [8.33.33.112]:5060
   Destn SIP Resp Addr:Port: [8.33.33.112]:5060
   Destination Name : 8.33.33.112
   Number of Media Streams: 1
   Number of Active Streams: 1
   RTP Fork Object : 0x0
   Media Mode
                           : flow-through
   Media Stream 1
     State of the stream
                             : STREAM_ACTIVE
     Stream Call ID
                             : 5
     Stream Type
                              : voice+dtmf (1)
     Stream Media Addr Type : 1
     Negotiated Codec : No Codec Codec Payload Type : 255 (None
                                             (0 bytes)
     Codec Payload Type : 255 (None)
Negotiated Dtmf-relay : inband-voice
     : -1
     QoS ID
                           : BestEffort
     Local QoS Strength
     Negotiated QoS Strength : BestEffort
     Negotiated QoS Direction : None
     Local QoS Status : None
     Media Source IP Addr:Port: [8.3.3.5]:16882
```

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

show voice register pool unregistered

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

show voice register pool unregistered

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

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

Examples

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

```
Router# show voice register pool unregistered Pool Tag: 2
```

```
MAC Address
                          : 0015.C68E.6D13
No. of attempts to register: 0
Unregister time
Last register request time :
Reason for state unregister:
        No registration request since last reboot/unregister
Pool Tag: 3
MAC Address
                          : 0021.5553.8998
No. of attempts to register: \mathbf{0}
Unregister time :
Last register request time :
Reason for state unregister:
       No registration request since last reboot/unregister
Pool Tag: 4
                          : 8989.9867.8769
MAC Address
No. of attempts to register: 0
Unregister time
Last register request time :
Reason for state unregister:
        No registration request since last reboot/unregister
```

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

show voice register statistics

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

show voice register statistics [global | pool tag]

Syntax Description

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

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco SIP SRST 3.0	This command was introduced.
12.3(4)T	Cisco SIP SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.4(4)T	Cisco CME 3.4 Cisco SIP SRST 3.4	This command was added to Cisco CME.
15.1(2)T	Cisco CME 8.1 Cisco SIP SRST 8.1	This command was modified. The global and pool keywords and <i>tag</i> argument were added. The output display was also modified to show more information about pools in unregistered state and time-stamps of registration event.

Usage Guidelines

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

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

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

Examples

Cisco Unified CME and Cisco Unified SRST

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

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

Cisco Unified CME and Cisco Unified SRST

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

```
Router# show voice register statistics global
Global Statistics:
 Active registrations : 1
 Total SIP phones registered: 2
 Total Registration Statistics
   R egistration requests : 97715
   Registration success : 3
                        : 97712
   Registration failed
   unRegister requests
                          : 1
   unRegister success
                          : 1
   unRegister failed
                          : 0
   Attempts to register
          after last unregister: 97712
   Last register request time : *06:45:11.127 UTC Wed Oct 14 2009
   Last unregister request time: *11:56:22.179 UTC Tue Oct 13 2009
   Register success time : *12:10:37.263 UTC Tue Oct 13 2009
   Unregister success time
                               : *11:56:22.182 UTC Tue Oct 13 2009
 Phones that have attempted registrations and have failed:
MAC address: 001b.535c.d410
IP address : 8.3.3.111
Attempts : 97712
Time of first attempt : *12:20:32.775 UTC Tue Oct 13 2009
Time of latest attempt: *06:46:14.815 UTC Wed Oct 14 2009
 Reason for failure
        Unauthorized registration request
```

Cisco Unified CME and Cisco Unified SRST

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

```
Router# show voice register statistics pool 1
```

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

Table 8 describes significant fields shown in this output.

Table 8 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.
Last Register Request Time	Used with all , pool , and statistics keywords. Shows details such as day, date, and time when the phones requested to register the last time.
Last unRegister Request Time	Used with all, pool, and statistics keywords. Shows details such as day, date, and time when the phones requested to unregister the last time.
Total Registration Statistics	Used with the all , pool , and statistics keywords. Shows the total registration statistics for this pool.
Registration requests	Used with the all , pool , and statistics keywords. Shows the incoming registration requests.
Registration success	Used with the all , pool , and statistics keywords. Shows the successful registrations.
Registration failed	Used with the all , pool , and statistics keywords. Shows the failed registrations.
unRegister requests	Used with the all , pool , and statistics keywords. Shows the incoming unregister/registration expire requests.
unRegister success	Used with the all , pool , and statistics keywords. Reports the number of successful unregisters.
unRegister failed	Used with the all , pool , and statistics keywords. Reports the number of failed unregisters.
Global statistics	Used with the statistics keyword. Details all active registrations.
Register pool <i>number</i> statistics	Used with the statistics keyword. Details specific pool statistics.

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

voice register global

To enter voice register global configuration mode in order to set global parameters for all supported Cisco SIP IP phones in a Cisco Unified CME or Cisco Unified Session Initiation Protocol (SIP) Survivable Remote Site Telephony (SRST) environment, use the **voice register global** command in global configuration mode. To automatically remove the existing DNs, pools, and global dialplan patterns, use the **no** form of this command.

voice register global

no voice register global

Syntax Description

This command has no arguments or keywords.

Command Default

There are no system-level parameters configured for SIP IP phones.

Command Modes

Global configuration (config)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4 Cisco SIP SRST 3.4	This command was introduced.
151(0)T	Cisco Unified CME 8.1 Cisco Unified SRST 8.1	The no form of the command was modified.

Usage Guidelines

Cisco Unified CME

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

Cisco Unified SIP SRST

Use this command to set provisioning parameters for multiple pools; that is, all supported Cisco SIP IP phones in a SIP SRST environment.

Cisco Unified CME 8.1 enhances the **no** form of **voice register global** command. The **no voice register global** command clears global configuration and also removes the configurations for voice register template, voice register dialplan, and voice register session-server.

A confirmation is sought before the cleanup is made.

Examples

Cisco Unified CME

The following is partial sample output from the **show voice register global** command. All of the parameters listed were set under voice register global configuration mode:

Router# show voice register global CONFIG [Version=4.0(0)] Version 4.0(0) Mode is cme Max-pool is 48 Max-dn is 48 Source-address is 10.0.2.4 port 5060 Load 7960-40 is P0S3-07-4-07 Time-format is 12 Date-format is M/D/YTime-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

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 Unified system.	

Feature Information for Cisco Unified SRST 8.1

Table 9 lists the release history for this feature.

Not all commands may be available in your Cisco IOS software release. For release information about a specific command, see the command reference documentation.

Use Cisco Feature Navigator to find information about platform support and software image support. Cisco Feature Navigator enables you to determine which Cisco IOS software images support a specific software release, feature set, or platform. To access Cisco Feature Navigator, go to http://www.cisco.com/go/cfn. An account on Cisco.com is not required.



Table 9 lists only the Cisco IOS software release that introduced support for a given feature in a given Cisco IOS software release train. Unless noted otherwise, subsequent releases of that Cisco IOS software release train also support that feature.

Table 9 Feature Information for Cisco Unified SRST 8.1

Feature Name	Releases	Feature Information
Cisco Unified SRST 8.1	15.1(2)T	Toll Fraud Prevention Enhancement, page 2
		Enhancements to SIP Phone Configuration, page 4

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