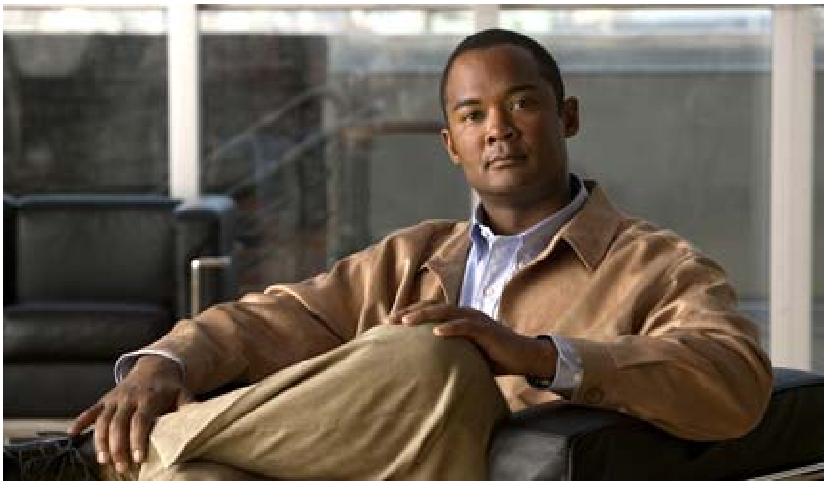

Cisco Unified Communications Manager Express Command Reference

June 20, 2007

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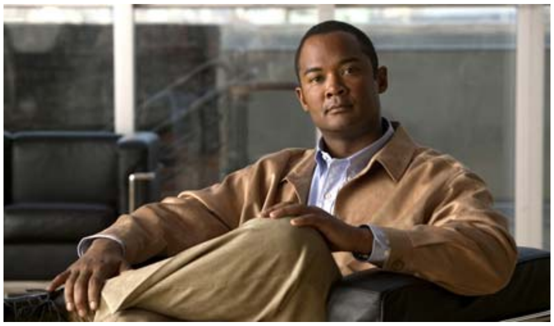
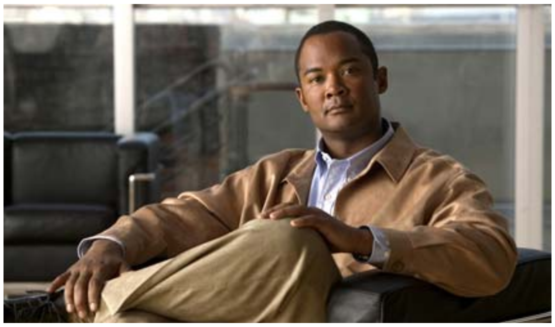
Cisco Unified Communications Manager Express Command Reference
OL-10894-01



CONTENTS



Last Updated: March 8, 2007
First Published: February 27, 2006

Obtaining Documentation, Obtaining Support, and Security Guidelines

For information on obtaining documentation, obtaining support, providing documentation feedback, security guidelines, and also recommended aliases and general Cisco documents, see the monthly *What's New in Cisco Product Documentation*, which also lists all new and revised Cisco technical documentation at:

http://www.cisco.com/en/US/docs/general/whatsnew/whatsnew.html.

Additional References

Cisco Unified Communications Manager Express (formally known as Cisco Unified CallManager Express):

http://www.cisco.com/en/US/products/sw/voicesw/ps4625/products_documentation_roadmap09186a0080189132.html

Cisco Unified IP Phones:

http://www.cisco.com/en/US/products/sw/voicesw/ps4625/products_user_guide09186a008018912b.html

Cisco SIP Configuration Guide:

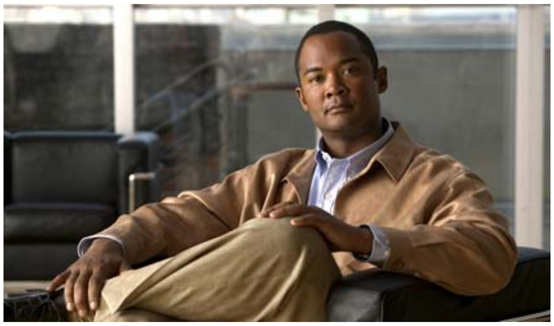
http://www.cisco.com/en/US/products/sw/iosswrel/ps5207/products_installation_and_configuration_guides_list.html

Cisco Unified SRST:

http://www.cisco.com/en/US/products/sw/voicesw/ps2169/products_documentation_roadmap09186a008018912f.html

Cisco 12.4 Voice:

http://www.cisco.com/en/US/products/ps6441/prod_configuration_guide09186a0080565f8a.html



New, Modified, Replaced, and Removed Commands for Cisco Unified CME 4.0 and later

Last Updated: June 18, 2007
First Published: February 27, 2006

New, modified, replaced, and removed commands for Cisco Unified Communications Manager Express 4.0(formally known as Cisco Unified CallManager Express) and later versions are grouped by version and type, and ordered alphabetically within each group. Use the table of contents to go to a group of commands, or use your web browser's Find function to search for a command. To access the documentation for a command, click the highlighted page number following the command name.

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12.4(11)XW1 New Commands

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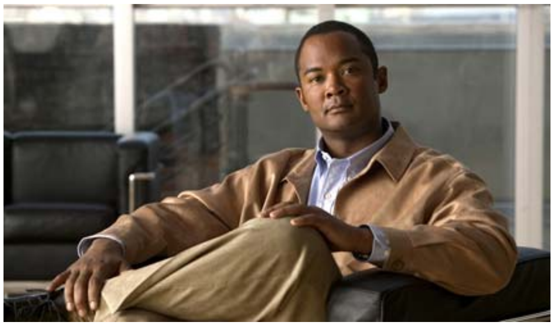
12.4(4)XC Replaced Commands



Cisco Unified CME Commands: A

Last Updated: June 18, 2007
First Published: February 27, 2006

This chapter contains commands to configure and maintain Cisco Unified Communications Manager Express (formally known as Cisco Unified CallManager Express). The commands are presented in alphabetical order. Some commands required for configuring Cisco Unified Communications Manager Express (Cisco Unified CME) may be found in other Cisco IOS command references. Use the command reference master index or search online to find these commands.

after-hour exempt

To specify that an individual IP phone in Cisco Unified CME does not have any of its outgoing calls blocked even though after-hour call blocking has been enabled, use the **after-hour exempt** command in ephone or ephone-template configuration mode. To remove the exemption, use the **no** form of this command.

after-hour exempt

no after-hour exempt

Syntax Description

This command has no arguments or keywords.

Command Default

The SCCP phone is not exempt from call blocking.

Command Modes

Ephone configuration
Ephone-template configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.4(4)XC	Cisco Unified CME 4.0	This command was made available in ephone-template configuration mode.
12.4(9)T	Cisco Unified CME 4.0	This command in the ephone-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Use this command to exempt an individual SCCP phone from call blocking and enable the phone user to place outgoing calls regardless of whether the outgoing called number matches the defined pattern of digits during the call blocking periods.

By default, all IP phones in a Cisco Unified CME system are subject to call blocking if the Call Blocking feature is configured.

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

Examples

The following example shows how to configure this phone so that outgoing calls are not blocked:

```
Router(config)# ephone 23
Router(config-ephone)# mac 00e0.8646.9242
Router(config-ephone)# button 1:33
Router(config-ephone)# after-hour exempt
```

Related Commands

Command	Description
after-hours block pattern	Defines a pattern of digits for blocking outgoing calls from IP phones.
after-hours date	Defines a recurring period based on date during which outgoing calls that match defined block patterns are blocked on IP phones.
after-hours day	Defines a recurring period based on day of the week during which outgoing calls that match defined block patterns are blocked on IP phones.
ephone	Enters ephone configuration mode.

after-hour exempt (voice register)

To specify that call blocking criteria is not applied, enabling the phone user to place outgoing calls on a SIP IP phone or extension even though global system call blocking is enabled, use the **after-hour exempt** command in voice register dn configuration or voice register pool mode. To return to the default, use the **no** form of this command.

after-hour exempt

no after-hour exempt

Syntax Description

This command has no arguments or keywords.

Command Default

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

Command Modes

Voice register dn configuration
Voice register pool configuration

Command History

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

Usage Guidelines

This command in voice register dn configuration exempts an individual SIP extension number or telephone number from call-blocking criteria.

This command in voice register pool configuration exempts all numbers associated with a SIP IP phone from call-blocking criteria.

To define call blocking for IP phones in Cisco Unified CME: First, define one or more patterns of outgoing digits by using the **after-hours block pattern** command in either telephony-service configuration mode for Cisco Unified CME or in call-manager-fallback configuration mode for Cisco Unified SIP SRST. Next, define one or more time periods during which calls that match those patterns are to be blocked by using the **after-hours date** or **after-hours day** command or both.

By default, all Cisco IP phones in a Cisco Unified CME or Cisco Unified SIP SRST system are restricted during the specified time if at least one pattern and at least one time period are defined. A SIP phone extension is exempt as long as the **after-hour exempt** command is configured in voice register dn or in voice register pool configuration mode.

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

Examples

The following example shows how to configure extension 5001, under directory number 2 so that outgoing calls are not blocked:

```
Router(config)# voice register dn 2
```

```
Router(config-register-dn)# number 5001
Router(config-register-dn)# after-hour exempt
```

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

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

Related Commands

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

after-hours block pattern

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

after-hours block pattern *pattern-tag pattern* **[7-24]**

no after-hours block pattern *pattern-tag*

Syntax Description

pattern-tag	Identifier for a call-blocking pattern. Up to 32 call-blocking patterns can be defined in separate commands.
pattern	Outgoing call digits to be matched for blocking.
7-24	(Optional) If the **7-24** keyword is specified, the pattern is always blocked, 7 days a week, 24 hours a day. If the **7-24** keyword is not specified, the pattern is blocked during the days and dates that are defined with the **after-hours day** and **after-hours date** commands.

Command Default

No pattern is defined.

Command Modes

Telephony-service configuration

Command History

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

Usage Guidelines

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

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

Examples

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

```
Router(config)# telephony-service
Router(config-telephony)# after-hours block pattern 1 9011
```

Related Commands

Command	Description
after-hour exempt	Specifies that an IP phone does not have any of its outgoing calls blocked even though call blocking has been defined.
after-hours date	Defines a recurring period based on date during which outgoing calls that match defined block patterns are blocked on IP phones.
after-hours day	Defines a recurring period based on day of the week during which outgoing calls that match defined block patterns are blocked on IP phones.
telephony-service	Enters telephony-service configuration mode.

after-hours date

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

after-hours date *month date start-time stop-time*

no after-hours date *month date*

Syntax Description

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

Command Default

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

Command Modes

Telephony-service configuration

Command History

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

Usage Guidelines

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

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

Examples

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

```
Router(config)# telephony-service
Router(config-telephony)# after-hours date jan 1 00:00 00:00
```

Related Commands

Command	Description
after-hour exempt	Specifies that an IP phone does not have any of its outgoing calls blocked even though call blocking has been defined.
after-hours block pattern	Defines a pattern of digits for blocking outgoing calls from IP phones.
after-hours day	Defines a recurring period based on day of the week during which outgoing calls that match defined block patterns are blocked on IP phones.
telephony-service	Enters telephony-service configuration mode.

after-hours day

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

after-hours day *day start-time stop-time*

no after-hours day *day*

Syntax Description

day	Abbreviated day of the week. The following abbreviations for day of the week are valid: **sun**, **mon**, **tue**, **wed**, **thu**, **fri**, **sat**.
start-time *stop-time*	Beginning and ending times for call blocking, in an HH:MM format using a 24-hour clock. The stop time can be smaller than the start time. The value 24:00 is not valid. If 00:00 is entered as a stop time, it is changed to 23:59. If 00:00 is entered for both start time and stop time, calls are blocked for the entire 24-hour period on the specified day.

Defaults

No time period based on day of the week is defined for call blocking.

Command Modes

Telephony-service configuration

Command History

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

Usage Guidelines

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

Call blocking occurs during the hours between the start time and stop time on the day of the week that is specified in this command. This time period recurs weekly unless it is removed using the **no** form of this command.

Examples

The following example defines the period from Monday night at 7 p.m. to Tuesday morning at 7 a.m. as an after-hours call-blocking period:

```
Router(config)# telephony-service
Router(config-telephony)# after-hours day mon 19:00 07:00
```

Related Commands

Command	Description
after-hour exempt	Specifies that an IP phone does not have any of its outgoing calls blocked even though call blocking has been defined.
after-hours block pattern	Defines a pattern of digits for blocking outgoing calls from IP phones.
after-hours date	Defines a recurring period based on date during which outgoing calls that match defined block patterns are blocked on IP phones.
telephony-service	Enters telephony-service configuration mode.

allow watch

To allow a directory number on a phone registered to Cisco Unified CME to be watched in a presence service, use the **allow watch** command in ephone-dn, ephone-dn-template, or voice register dn configuration mode. To reset to the default condition, use the **no** form of this command.

allow watch

no allow watch

Syntax Description

This command has no arguments or keywords.

Command Default

Watching of the phone line is disabled.

Command Modes

Ephone-dn
Ephone-dn-template
Voice register dn

Command History

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

Usage Guidelines

This command controls whether a phone line associated with a directory number can be watched as part of a presence service. The directory number is enabled as a presentity that can be watched by internal and external watchers. Presence service must be enabled on Cisco Unified CME. Another phone, acting as a watcher, can monitor the status of this phone line when the **blf-speed-dial** or **presence call-list** command is enabled for that phone.

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

Examples

The following example shows that the extension associated with voice register dn 2 can be watched by the phone associated with voice register pool 1:

```
Router(config)# voice register dn 2
Router(config-register-dn)# number 2102
Router(config-register-dn)# allow watch

Router(config)# voice register pool 1
Router(config-register-pool)# id mac 0015.6247.EF90
Router(config-register-pool)# type 7971
Router(config-register-pool)# number 1 dn 2
Router(config-register-pool)# blf-speed-dial 1 2102 label 2102
```

Related Commands

Command	Description
blf-speed-dial	Enables Busy Lamp Field (BLF) monitoring for a speed-dial number on a phone registered to Cisco Unified CME.
presence	Enables presence service and enters presence configuration mode.
presence call-list	Enables BLF monitoring for call lists and directories on phones registered to Cisco Unified CME.
presence enable	Allows the router to accept incoming presence requests.
show presence global	Displays configuration information about the presence service.
show presence subscription	Displays information about active presence subscriptions.

anonymous block

To enable anonymous call blocking in a SIP phone template, use the **anonymous block** command in voice register template configuration mode. To return to the default, use the **no** form of this command.

anonymous block

no anonymous block

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled

Command Modes

Voice register template configuration

Command History

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

Usage Guidelines

This command blocks incoming calls in which the caller is not identified. To apply a template to a SIP phone, use the **template** command in voice register pool configuration mode.

Examples

The following example shows how to set anonymous call blocking in template 1:

```
Router(config)# voice register template 1
Router(config-register-temp)# anonymous block
```

Related Commands

Command	Description
caller-id block (voice register template)	Enables caller-ID blocking for outbound calls from a SIP phone.
template (voice register pool)	Applies a template to a SIP phone.

application (ephone-dn)

To select a session-level application for a specific extension (ephone-dn) in a Cisco CallManager Express (Cisco CME) system, use the **application** command in ephone-dn configuration mode. To disable use of the application, use the **no** form of this command.

application *application-name* [**out-bound**]

no application *application-name* [**out-bound**]

Syntax Description

application-name	Interactive voice response (IVR) application name.
out-bound	(Optional) Application handles the dial peer in outgoing mode.

Defaults

No application is selected for the phone.

Command Modes

Ephone-dn configuration

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.2(2)XT	2.0	This command was introduced on the Cisco 1750, Cisco 1751, Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(8)T	2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725, Cisco 3745, and Cisco MC3810-V3.
12.2(8)T1	2.0	This command was implemented on the Cisco 2600-XM and Cisco 2691.
12.2(11)T	2.01	This command was implemented on the Cisco 1760.

Usage Guidelines

Use this command to assign a Tool Command Language (Tcl) IVR application to a Cisco IP phone extension (ephone-dn).

Use the **show call application voice summary** command to display a list of applications.

Examples

The following example sets the IVR application for directory number 1:

```
Router(config)# ephone-dn 1
Router(config-ephone-dn) application TCL IVR
```

Related Commands

Command	Description
ephone-dn	Enters ephone-dn configuration mode.
show call application voice summary	Displays information about voice applications.

application (telephony-service)

To select a session-level application for all extensions (ephone-dns) in a Cisco CallManager Express (Cisco CME) system, use the **application** command in telephony-service configuration mode. To disable use of an application for all extensions, use the **no** form of this command.

application *application-name*

no application

Syntax Description

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

Defaults

No application is selected for all extensions.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.2(11)YT	2.1	This command was introduced.
12.2(15)T	2.1	This command was integrated into Cisco IOS Release 12.2(15)T.

Usage Guidelines

Use this command to assign a Tool Command Language (Tcl) IVR application to all extensions served by the CME router.

Use the **show call application voice summary** command to display a list of applications.

Examples

The following example sets the IVR application for all phones:

```
Router(config)# telephony-service
Router(config-telephony) application TCL IVR
```

Related Commands

Command	Description
show call application voice summary	Displays information about voice applications.
telephony-service	Enters telephony-service configuration mode.

application (voice register global)

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

application *application-name*

no application

Syntax Description

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

Command Default

Default application on router

Command Modes

Voice register global configuration

Command History

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

Usage Guidelines

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

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

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

Examples

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

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

Related Commands

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

application (voice register pool)

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

application *application-name*

no application

Syntax Description

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

Command Default

Default application on router

Command Modes

Voice register pool configuration

Command History

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

Usage Guidelines

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

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

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

Examples

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

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

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

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

Related Commands

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

authenticate (voice register global)

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

Cisco IOS Release 12.4(11)XJ and later releases

authenticate {**credential** *tag location* | **ood-refer** | **presence** | **realm** *string* | **register**}

no authenticate {**credential** *tag location* | **ood-refer** | **presence** | **realm** *string* | **register**}

Cisco IOS Release 12.4(4)T

authenticate [**all**] [**realm** *string*]

no authenticate [**all**] [**realm** *string*]

Syntax Description

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

Command Default

Authenticate mode is disabled.

Command Modes

Voice register global configuration

Command History

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

Usage Guidelines

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

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

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

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

Examples

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

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

Related Commands

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

auth-mode

To specify the type of authentication to use during CAPF sessions, use the **auth-mode** command in CAPF-server configuration mode. To return to the default, use the **no** form of this command.

auth-mode {**auth-string** | **LSC** | **MIC** | **none** | **null-string**}

no auth-mode

Syntax Description

auth-string	The phone user enters a special authentication string at the phone. The string is entered using the **auth-string** command and is provided to the phone user by the system administrator.
LSC	The phone provides its phone certificate for authentication. Precedence is given to an LSC if one exists.
MIC	The phone provides its phone certificate for authentication. Precedence is given to an MIC if one exists.
none	No certificate upgrade is initiated.
null-string	No authentication is used.

Command Default

No certificate upgrade is initiated (same as the keyword **none**).

Command Modes

CAPF-server configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used with Cisco Unified CME phone authentication.

If you use the **auth-string** keyword with this command, the phone user is required to enter a specified digit string at the phone to be authenticated for CAPF sessions. The digit string is entered into the configuration using the **auth-string** command or the **capf-auth-str** command and must be communicated to the phone user.

Use the **show capf-server** command to display parameters that you have set with this command.

Examples

The following example specifies authentication strings as the method of CAPF authentication. The **auth-string** command specifies that random authentication strings should be generated for all ephones.

```
capf-server
 auth-mode auth-string
 auth-string generate all
```

Related Commands

Command	Description
auth-string	Creates or removes authentication strings for one or all secure ephones.
capf-auth-str	Specifies a string of digits for a user to enter at the phone for CAPF authentication.
show capf-server	Displays configuration and session information for the CAPF server.

auth-string

To generate or remove authentication strings for one or all secure ephones, use the **auth-string** command in CAPF-server configuration mode. To return to the default, use the **no** form of this command.

auth-string {**delete** | **generate**} {**all** | *ephone-tag*} [*digit-string*]

no auth-string {**delete** | **generate**} {**all** | *ephone-tag*} [*digit-string*]

Syntax Description

delete	Remove authentication string(s) for the specified secure device(s).
generate	Create authentication string(s) for the specified secure device(s).
all	All devices.
ephone-tag	Identifier for the ephone to receive the authentication string.
digit-string	(Optional) Digits to use as an authentication string. If this argument is not specified, a random string is generated for each phone.

Command Default

No authentication string exists.

Command Modes

CAPF-server configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used with Cisco Unified CME phone authentication.

This command creates or removes authentication strings for all secure ephones or for a specified secure ephone. Use this command when the **auth-string** keyword is specified in the **auth-mode** command.

Authentication mode for individual ephones can also be set using the **cert-oper (ephone)** command.

CAPF authentication strings for particular ephones can also be entered using the **capf-auth-str** command in ephone configuration mode.

Use the **show capf-server auth-string** command to display configured authentication strings.

When a phone is configured for a certificate upgrade that requires auth-string authentication, then the CAPF initiation needs to be performed manually by the phone user using the following steps:

1. Press the Settings button.
2. If the configuration is locked, press **# (asterisk, asterisk, pound sign) to unlock it.
3. Scroll down the menu and select Security Configuration.
4. Scroll down the next menu to LSC and press the Update soft key.
5. When prompted for the authentication string, enter the string provided by the system administrator.

Examples

The following example specifies authentication strings as the method of CAPF authentication. The **auth-string** command specifies that random authentication strings should be generated for all ephones.

```
capf-server
 auth-mode auth-string
 auth-string generate all
```

Related Commands

Command	Description
auth-mode	Specifies the type of authentication to use during CAPF sessions.
capf-auth-str	Specifies a string of digits for a user to enter at the phone for CAPF authentication.
cert-oper (ephone)	Initiates a certificate activity for an individual ephone and specifies the type of authentication.
show capf-server	Displays configuration and session information for the CAPF server.

auto assign

To automatically assign an already defined telephone or extension number to button 1 of Cisco Unified IP phones as they register for service with a Cisco Unified CME router, use the **auto assign** command in telephony-service configuration mode. To return to the default of not automatically assigning dn-tags, use the **no** form of this command.

auto assign *dn-tag* **to** *dn-tag* [**type** *phone-type*] [**cfw** *extension-number* **timeout** *seconds*]

no auto assign *dn-tag* **to** *dn-tag* [**type** *phone-type*] [**cfw** *extension-number* **timeout** *seconds*]

Syntax Description

dn-tag **to** *dn-tag*	Range of ephone-dn tags for already configured ephone-dns, from which a tag is assigned to the ephone being created. The maximum number of directory numbers supported is version and platform dependent. Type **?** to display the value.
type *phone-type*	(Optional) Type of Cisco Unified IP phone to which to restrict automatic assignment of ephone-dn tags. Valid entries are the following: • **7902**—Cisco Unified IP Phone 7902G. • **7905**—Cisco Unified IP Phone 7905G. • **7910**—Cisco Unified IP Phone 7910 and 7910G. • **7911**—Cisco Unified IP Phone 7911G. • **7912**—Cisco Unified IP Phone 7912G. • **7920**—Cisco Unified Wireless IP Phone 7920. • **7921**—Cisco Unified Wireless IP Phone 7921G. • **7931**—Cisco Unified Wireless IP Phone 7931G. • **7935**—Cisco Unified IP Conference Station 7935. • **7936**—Cisco Unified IP Conference Station 7936. • **7940**—Cisco Unified IP Phones 7940 and 7940G. • **7941**—Cisco Unified IP Phone 7941G. • **7941GE**—Cisco Unified IP Phone 7941G-GE. • **7960**—Cisco Unified IP Phones 7960 and 7960G. • **7961**—Cisco Unified IP Phone 7961G. • **7961GE**—Cisco Unified IP Phone 7961G-GE. • **7970**—Cisco Unified IP Phone 7970G. • **7971**—Cisco Unified IP Phone 7971G-GE. • **ata**—Cisco ATA-186 or Cisco ATA-188.
cfw	(Optional) Automatically assigned ephone-dns are provisioned for call-forward busy and no-answer to the specified extension number.

extension-number	(Optional) Extension number to which calls are to be forwarded on busy and no-answer conditions.
timeout *seconds*	(Optional; required if the **cfw** keyword is used) Amount of time, in seconds, to wait when a call is not being answered before forwarding it. Range: 3 to 60000.

Command Default

Ephone-dn tags are not automatically assigned to registering Cisco Unified IP phones.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.3(7)T	Cisco CME 3.1	The **7920** and **7936** keywords were added.
12.3(11)XL	Cisco CME 3.2.1	The **7970** keyword was added.
12.3(14)T	Cisco CME 3.3	The **7971** keyword was added, and this command was integrated into Cisco IOS Release 12.3(14)T.
12.4(4)XC	Cisco Unified CME 4.0	The **7941**, **7941GE**, **7961**, and **7961GE** keywords were added.
12.4(9)T	Cisco Unified CME 4.0	The **7941**, **7941GE**, **7961**, and **7961GE** keywords were integrated into Cisco IOS Release 12.4(9)T.
12.4(6)XE	Cisco Unified CME 4.0(2)	The **7931** keyword was added.
12.4(4)XC4	Cisco Unified CME 4.0(3)	The **7931** keyword was integrated into Cisco IOS Release 12.4(4)XC4.
12.4(11)T	Cisco Unified CME 4.0(3)	The **7931** keyword was integrated into Cisco IOS Release 12.4(11)T.
12.4(11)XJ2	Cisco Unified CME 4.1	The **7921** keyword was added.
12.4(15)T	Cisco Unified CME 4.1	The **7921** keyword was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

Use this command to create an ephone configuration for a Cisco Unified IP phone whose MAC address is not explicitly configured as it registers in Cisco Unified CME. The system-created ephone configuration includes the MAC address of the Cisco Unified IP phone being registered and an already-defined available ephone-dn assigned to button 1 of this phone.

The **auto-reg-ephone** command must be enabled (default) to use this command. If the autoregistration feature is disabled, a Cisco Unified IP phone whose MAC address is not explicitly configured cannot register in Cisco Unified CME.

Before using this command, configure the ephone-dn tags to be assigned and define at least one primary number for each dn-tag.

All ephone-dns in a specified range should be of the same type, either single-line or dual-line.

Ephone-dn tags to be assigned must belong to normal ephone-dns and cannot belong to paging ephone-dns, intercom ephone-dns, music-on-hold (MOH) ephone-dns, or message-waiting-indication (MWI) ephone-dns.

The **auto assign** command cannot create shared lines.

If an insufficient number of dn-tags is available, some ephone configurations will not include a telephone or extension number.

Use multiple **auto assign** commands to assign discontinuous ranges of ephone-dn tags and to support multiple types of IP phones. Overlapping ranges of dn-tags may be assigned so that they map to more than one type of phone. If no **type** is specified, the values in the range are assigned to phones of any type, and if a specific range is assigned for a specific phone type, the available ephone-dn tag in that range are used first.

If the phone being registered is connected to a Cisco VG200 series analog phone gateway, configuring the **auto assign** command will automatically create one ephone configuration for each configured port, as the port registers with the Cisco Unified CME router. To ensure that the tag-to-port assignment will match the numbering order of the physical ports; for example, dn-tags 1 to 24 assigned to ports 1 to 24 of a Cisco VG224 analog phone gateway, in that order, we recommend that the Cisco Unified CME system be up, running, and configured *before* you boot the analog phone gateway.

The **auto assign** command cannot be used for the Cisco Unified IP Phone 7914 Expansion Module. Phones with one or more expansion modules must be configured manually.

After using this command, reboot the phone for which an ephone is to be configured.

This command is also used by the Cisco Unified CME setup tool to automatically assign ephone-dns after the tool has gathered information about the setup from the user. When lines are assigned by the Cisco Unified CME setup tool in keyswitch mode with two ephone-dn entries created for each individual extension number, the automatic assignment mechanism assigns both ephone-dn entries to an individual ephone associated with an IP phone.

Note Care should be taken when using the **auto assign** command because this command grants telephony service to *any* IP phone that attempts to register. If you use the **auto assign** command, ensure that your network is secure from unauthorized access by unknown IP phones.

Examples

The following examples show how to configure the Auto Assign feature, including prerequisite commands for configuring the **auto assign** command.

The following example shows how to enter the ephone-dn configuration and create ephone-dns configurations, tags 1-4, each having a single primary number:

```
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 2000
Router(config-ephone-dn)# exit
Router(config)# ephone-dn 2
Router(config-ephone-dn)# number 3000
Router(config-ephone-dn)# exit
Router(config)# ephone-dn 3
Router(config-ephone-dn)# number 4000
Router(config-ephone-dn)# exit
Router(config)# ephone-dn 4
Router(config-ephone-dn)# number 4001
Router(config-ephone-dn)# exit
```

The following example shows how to designate ephone-dn tags 1 to 4 for automatic assignment to any type of IP phone and then perform a fast reboot of all phones:

```
Router(config)# telephony-service
Router(config-telephony)# auto assign 1 to 4
Router (config-telephony)# restart all
```

The following example is the partial output from the **show ephone registered** command listing four registered IP phones, to which ephone-dn tags 1 to 4 have been automatically assigned, after the phones were booted:

```
Router# show ephone registered
ephone-1 Mac:0003.E3E7.F627 TCP socket:[2] activeLine:0 REGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
IP:10.0.0.2 51671 Telecaster 7940 keepalive 28 max_line 2
button 1: dn 1 number 2000
ephone-2 Mac:0030.94C3.F43A TCP socket:[1] activeLine:0 REGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
IP:10.0.0.3 50094 Telecaster 7960 keepalive 28 max_line 6
button 1: dn 2 number 3000
ephone-3 Mac:0003.6B40.99DA TCP socket:[3] activeLine:0 REGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
IP:10.0.0.200 51879 Telecaster 7960 keepalive 28 max_line 6
button 1: dn 3 number 4000
ephone-4 Mac:0010.406B.99D9 TCP socket:[4] activeLine:0 REGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
IP:10.0.0.012 51879 Telecaster 7960 keepalive 28 max_line 6
button 1: dn 4 number 4001
.
.
.
```

The following example shows how to designate ephone-dn tags 1 to 12 for automatic assignment to Cisco Unified IP Phone 7910Gs only and ephone-dn tags 13 to 20 for automatic assignment to a Cisco Unified IP Phones 7960 and 7960Gs only, with call forwarding to extension 5001 on busy or after 30 seconds of ringing with no answer:

```
Router(config)# telephony-service
Router(config-telephony)# auto assign 1 to 12 type 7910
Router(config-telephony)# auto assign 13 to 20 type 7960 cfw 5001 timeout 30
```

Related Commands

Command	Description
auto-reg-ephone	Enables registration of Cisco Unified IP phones for which MAC addresses are not explicitly configured.
number	Associates a telephone or extension number with an ephone-dn.
restart (ephone)	Performs a fast reboot of a single phone associated with a Cisco Unified CME router.
restart (telephony-service)	Performs a fast reboot of one or all phones associated with a Cisco Unified CME router.
show ephone	Displays statistical information about registered Cisco Unified IP phones.
show ephone registered	Displays the status of registered phones.

auto logout

To enable the automatic change of an ephone hunt group agent's ephone-dn to not-ready status after a specified number of hunt-group calls are not answered, use the **auto logout** command in ephone-hunt configuration mode. To disable automatic logout, use the **no** form of this command.

auto logout [*number-of-calls*] [**dynamic** | **static**]

no auto logout [*number-of-calls*] [**dynamic** | **static**]

Syntax Description

number-of-calls	(Optional) Number of unanswered hunt-group calls to an ephone-dn before the ephone-dn is automatically changed to not-ready status. Range is from 1 to 20. Default is 1.
dynamic	(Optional) Specifies that this command applies only to dynamic hunt group members (those who are specified by an asterisk (*) wildcard in the hunt group configuration). If neither the **dynamic** nor **static** keyword is used, automatic logout applies to both dynamic and static hunt group members.
static	(Optional) Specifies that this command applies only to static hunt group members (those whose extension numbers are explicitly named in the hunt group configuration). If neither the **dynamic** nor **static** keyword is used, automatic logout applies to both dynamic and static hunt group members.

Command Default

Automatic change of agent status to not-ready is disabled.

Command Modes

Ephone-hunt configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)XL	Cisco CME 3.2.1	This command was introduced.
12.3(14)T	Cisco CME 3.3	This command was integrated into Cisco IOS Release 12.3(14)T.
12.4(4)XC	Cisco Unified CME 4.0	The *number-of-calls* argument and the **dynamic** and **static** keywords were added. The criterion for this command was changed from exceeding the **timeout** command limit to exceeding the number of calls specified in this command.
12.4(9)T	Cisco Unified CME 4.0	The modifications made to this command were integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is valid only for the following Cisco IP phones:

- Cisco Unified IP Phone 7905G
- Cisco Unified IP Phone 7912G
- Cisco Unified IP Phones 7940 and 7940G

- Cisco Unified IP Phones 7960 and 7960G

This command is used with the Automatic Agent Status Not-Ready feature for ephone hunt groups, which automatically puts an agent's phone in not-ready status when it exceeds a specified limit. The limit at which the Automatic Agent Status Not-Ready feature is triggered depends on the Cisco CME version that you are using, as follows:

- Cisco CME 3.3 and earlier versions—Automatic Agent Status Not-Ready is invoked when an ephone-hunt group call rings longer on a member ephone-dn than the period of time configured in the **timeout** command in ephone-hunt configuration mode.
- Cisco Unified CME 4.0 and later versions—Automatic Agent Status Not-Ready is invoked when the specified number of ephone-hunt group calls is unanswered by an agent. The default is one call if the number of calls is not explicitly specified.

When Automatic Agent Status Not-Ready is specified for an ephone hunt group and it is triggered because an ephone-dn member does not answer a specified number of ephone hunt group calls, the following actions take place:

- If the **hunt-group logout HLog** command has been used, the agent is placed in not-ready status. The agent's phone will not receive further hunt-group calls but will receive calls that directly dial the phone's extension numbers. An agent in not-ready status can return to ready status by pressing the HLog soft key or by using the HLog feature access code (FAC).
- If the **hunt-group logout HLog** command has not been used or if the **hunt-group logout DND** command has been used, the phone on which the ephone-dn appears is placed into Do Not Disturb (DND) mode, in which the ephone-dn does not receive any calls at all, including hunt-group calls. The red lamp on the phone lights to indicate DND status. An agent in DND mode can return to ready status by pressing the DND soft key or by using the DND FAC.
- When an agent returns to ready status, the ephone hunt group resumes sending calls to the agent's ephone-dn.

Note When an agent who is a dynamic member of a hunt group is in not-ready status, the agent's slot in the ephone hunt group is not relinquished. It remains reserved by the agent until the agent leaves the group.

You can use the **auto logout** command with any number of ephone hunt groups, but any ephone-dn to which the **auto logout** command applies must belong to only one ephone. Automatic Agent Status Not-Ready is not supported on shared lines.

Examples

This section provides the following examples:

- Cisco CME 3.3 and Earlier Versions
- Cisco Unified CME 4.0 and Later Versions

Cisco CME 3.3 and Earlier Versions

In the following example, ephone hunt group 1 is configured to permit automatic logout. If hunt group calls that are presented to 1001 and 1002 are unanswered (that is, if they ring longer than 40 seconds each), ephone 1 and ephone 2 are automatically put into DND mode. All unanswered calls are sent to voice mail (5000).

```
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 1001

Router(config)# ephone-dn 2
Router(config-ephone-dn)# number 1002
```

```
Router(config)# ephone-hunt 1 peer
Router(config-ephone-hunt)# pilot 1111
Router(config-ephone-hunt)# list 1001, 1002
Router(config-ephone-hunt)# final 5000
Router(config-ephone-hunt)# timeout 40
Router(config-ephone-hunt)# auto logout

Router(config)# ephone 1
Router(config-ephone)# button 1:1
Router(config)# ephone 2
Router(config-ephone)# button 1:2
```

Cisco Unified CME 4.0 and Later Versions

In the following example, Automatic Agent Status Not-Ready is limited to dynamic hunt group members who do not answer two consecutive ephone hunt group calls. Ephone-dn 33, extension 1003, has dynamically joined ephone-hunt group 1. Ephone 3 will be put into DND mode if extension 1003 does not answer two consecutive hunt group calls. Ephones 1 and 2 will not be put into DND if they do not answer hunt-group calls, because the **auto logout** command applies only to dynamic hunt-group agents.

```
Router(config)# telephony-service
Router(config-telephony)# hunt-group logout DND

Router(config)# ephone-dn 11
Router(config-ephone-dn)# number 1001
Router(config)# ephone-dn 22
Router(config-ephone-dn)# number 1002
Router(config)# ephone-dn 33
Router(config-ephone-dn)# number 1003
Router(config-ephone-dn)# ephone-hunt login

Router(config)# ephone-hunt 1 peer
Router(config-ephone-hunt)# pilot 1111
Router(config-ephone-hunt)# list 1001, 1002, *
Router(config-ephone-hunt)# final 5000
Router(config-ephone-hunt)# auto logout 2 dynamic

Router(config)# ephone 1
Router(config-ephone)# button 1:11
Router(config)# ephone 2
Router(config-ephone)# button 1:22
Router(config)# ephone 3
Router(config-ephone)# button 1:33
```

In the following example, Automatic Agent Status Not-Ready cannot be used because all of the ephone-dns are shared.

```
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 1001
Router(config)# ephone-dn 2
Router(config-ephone-dn)# number 1002

Router(config)# ephone-hunt 1 peer
Router(config-ephone-hunt)# pilot 1111
Router(config-ephone-hunt)# list 1001, 1002
Router(config-ephone-hunt)# final 6000

Router(config)# ephone 1
Router(config-ephone)# button 1o1,2
Router(config)# ephone 2
Router(config-ephone)# button 1o1,2
```

Related Commands

Command	Description
ephone-hunt	Enters ephone-hunt configuration mode.
hunt-group logout	Enables separate handling of DND and HLog functionality for hunt-group agents and the display of an HLog soft key on phones.
timeout	Defines the number of seconds after which a call that is not answered is redirected to the next number in a Cisco Unified CME ephone-hunt-group list.

auto-answer

To enable the intercom auto-answer feature on a SIP phone extension, use the **auto-answer** command in voice register dn configuration mode. To return to the default, use the **no** form of this command.

auto-answer

no auto-answer

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled

Command Modes

Voice register dn configuration

Command History

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

Usage Guidelines

This command creates an IP phone line connection that resembles a private line, automatic ring-down (PLAR). The auto-answer causes an extension (directory number) to operate in auto-dial fashion for outbound calls and auto answer-with-mute for inbound calls. If an extension is configured for intercom operation, it can be associated with one Cisco IP phone only.

Any caller can dial an intercom extension, and a call to an intercom extension that is originated by a nonintercom caller triggers an automatic answer exactly like a legitimate intercom call. To prevent nonintercom originators from manually dialing an intercom destination, you can use alphabetic characters when you assign numbers to intercom extensions by using the **number** command. These characters cannot be dialed from a normal phone but can be dialed by preprogrammed intercom extensions when calls are made by the router.

Use the **reset** command to reset an individual SIP phone after you make changes to an extension for a SIP phone in Cisco CME.

Examples

The following example shows how to set the auto-answer feature on SIP phone directory number 1:

```
Router(config)# voice register dn 1
Router(config-register-dn) number A5001
Router(config-register-dn) auto-answer
```

Related Commands

Command	Description
number (voice register dn)	Associates a telephone or extension number with a directory number.

Command	Description
reset (voice register global)	Performs a complete reboot of all SIP phones associated with a Cisco CME router.
reset (voice register pool)	Performs a complete reboot of a single SIP phone associated with a Cisco CME router.

auto-line

To enable automatic line selection on an IP phone in a Cisco CallManager Express (Cisco CME) system, use the **auto-line** command in ephone configuration mode. To disable automatic line selection, use the **no** form of this command.

auto-line [*button-number* [**answer-incoming**] | **incoming**]

no auto-line

Syntax Description

button-number	(Optional) Selects the line associated with the specified button when the handset is lifted.
answer-incoming	(Optional) Enables automatic line selection for incoming calls on the line associated with the *button-number* argument.
incoming	(Optional) Enables automatic line selection for incoming calls only.

Command Default

Automatic line selection is enabled.

Command Modes

Ephone configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.3(7)T	Cisco CME 3.1	The *button-number* argument was added.
12.4(4)XC	Cisco Unified CME 4.0	The **answer-incoming** keyword was added.
12.4(9)T	Cisco Unified CME 4.0	The **answer-incoming** keyword was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Use the **auto-line** command with no keyword or argument enables automatic line selection on the specified ephone. Picking up a handset answers the first ringing line or, if no line is ringing, selects the first idle line. This is also the default behavior if this command is not used.

Use the **auto-line incoming** command enables automatic line selection for incoming calls only. Picking up the handset answers the first ringing line and, if no line is ringing, does not select an idle line for an outgoing call. Pressing a line button selects a line for an outgoing call.

Use the **auto-line** command with the *button-number* argument specifies the line that will automatically be selected when the handset is picked up to make an outgoing call. If a button number is specified and the line associated with that button is unavailable (because it is a shared line in use on another phone), no dial tone is heard when the handset is lifted. You must press an available line button to make an outgoing call. Incoming calls must be answered by pressing the Answer soft key or pressing the ringing line button.

Use the **answer-incoming** keyword with the *button-number* argument enables automatic line selection for incoming calls on the specified button. Picking up the handset answers the incoming call on the line button associated with the *button-number* argument.

Use the **no auto-line** command disables automatic line selection on the ephone that is being configured. Pressing the Answer soft key answers the first ringing line, and pressing a line button selects a line for an outgoing call. Picking up the handset does not answer calls or provide dial tone.

Examples

The following example shows how to disable automatic line selection. The phone user must use the Answer soft key or press a line button to answer calls, or the phone user must press a line button to initiate outgoing calls.

```
Router(config)# ephone 23
Router(config-ephone)# no auto-line
```

The following example shows how to enable automatic line selection for incoming calls only. The phone user picks up the handset to answer the first ringing line. To make outgoing calls, the phone user must press a line button.

```
Router(config)# ephone 24
Router(config-ephone)# auto-line incoming
```

The following example shows how to enable the automatic selection of line button 3 for outgoing calls when the handset is lifted. There is no automatic answering of incoming calls; the user presses the Answer soft key or presses a line button to answer a call.

```
Router(config)# ephone 26
Router(config-ephone)# auto-line 3
```

The following example shows how to enable the automatic selection of line button 3 when the handset is lifted to answer incoming calls or to make outgoing calls.

```
Router(config)# ephone 26
Router(config-ephone)# auto-line 3 answer-incoming
```

Related Commands

Command	Description
ephone	Enters ephone configuration mode.

auto-reg-ephone

To enable automatic registration of ephones with the Cisco Unified CME system, use the **auto-reg-ephone** command in telephony-service configuration mode. To disable automatic registration, use the **no** form of this command.

auto-reg-ephone

no auto-reg-ephone

Syntax Description

This command has no keywords or arguments.

Command Default

Automatic registration is enabled.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is enabled by default and allows automatic registration, in which Cisco Unified CME allocates an ephone slot to any ephone that connects to it, regardless of whether the ephone appears in the configuration or not.

The **no** form of this command blocks the automatic registration of ephones whose MAC addresses are not explicitly listed in the configuration. When automatic registration is blocked, Cisco Unified CME records the MAC addresses of phones that attempt to register but cannot because they are blocked.

Use the **show ephone attempted-registrations** command to view the list of phones that have attempted to register but have been blocked. Use the **clear telephony-service ephone-attempted-registrations** command to clear the list of phones that have attempted to register but have been blocked.

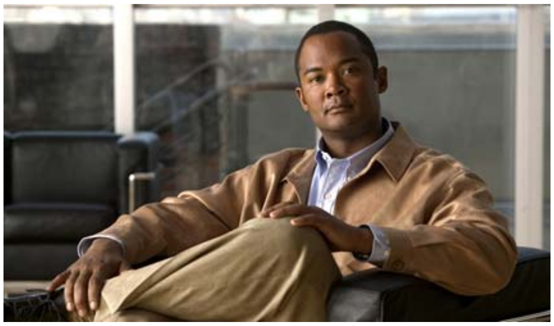
Examples

The following example disables automatic registration of ephones that are not listed in the configuration:

```
Router(config)# telephony-service
Router(config-telephony)# no auto-reg-ephone
```

Related Commands

Command	Description
clear telephony-service ephone-attempted-registrations	Empties the log of ephones that unsuccessfully attempt to register with Cisco Unified CME.
show ephone attempted-registrations	Displays the log of ephones that unsuccessfully attempt to register with Cisco Unified CME.



Cisco Unified CME Commands: B

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This chapter contains commands to configure and maintain Cisco Unified Communications Manager Express (formally known as Cisco Unified CallManager Express). The commands are presented in alphabetical order. Some commands required for configuring Cisco Unified Communications Manager Express (Cisco Unified CME) may be found in other Cisco IOS command references. Use the command reference master index or search online to find these commands.

b2bua

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

b2bua

no b2bua

Syntax Description

This command has no arguments or keywords.

Command Default

Disabled

Command Modes

Dial-peer configuration

Command History

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

Usage Guidelines

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

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

Examples

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

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

Related Commands

Command	Description
allow-connections	Enables calls between SIP endpoints in a VoIP network.
dial-peer voice	Defines a dial peer and enters dial-peer configuration mode.

Command	Description
mode (voice register global)	Enables the mode for provisioning SIP phones in a Cisco Unified CME system.
show dial-peer voice	Displays information for dial peers.
source-address (voice register global)	Identifies the IP address and port through which SIP phones communicate with a Cisco Unified CME router.

blf-speed-dial

To enable Busy Lamp Field (BLF) monitoring for a speed-dial number on a phone registered to Cisco Unified CME, use the **blf-speed-dial** command in ephone or voice register pool configuration mode. To disable BLF monitoring for speed-dial, use the **no** form of this command.

blf-speed-dial *tag number* **label** *string*

no blf-speed-dial *tag*

Syntax Description

tag	Number that identifies the speed-dial index. Range: 1 to 33 (SCCP); 1 to 7 (SIP).
number	Telephone number to speed dial.
label *string*	Alphanumeric label that identifies the speed-dial button. The string can contain a maximum of 30 characters.

Command Default

BLF monitoring is disabled.

Command Modes

Ephone configuration
Voice register pool configuration

Command History

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

Usage Guidelines

This command enables a phone to monitor the status of a line associated with a speed-dial number. The directory number associated with the speed-dial number must have presence enabled with the **allow watch** command.

For information on the BLF status indicators that display on specific types of phones, see the Cisco Unified IP Phone documentation for your phone model.

Examples

The following example shows the BLF speed-dial feature enabled for ephone 1. The line status of extensions 51212 and 51214 displays on the phone 1 provided that presence is enabled for those directory numbers.

```
Router(config)# ephone 1
Router(config-ephone)# blf-speed-dial 1 51212 label sales
Router(config-ephone)# blf-speed-dial 2 51214 label payroll
```

Related Commands

Command	Description
allow watch	Allows a directory number on a phone registered to Cisco Unified CME to be watched in a presence service.
presence	Enables presence service and enters presence configuration mode.
presence call-list	Enables BLF monitoring for call lists and directories on phones registered to Cisco Unified CME.
sccp blf-speed-dial	Sets the retry timeout for BLF notification for speed-dial numbers on SCCP phones registered to an external Cisco Unified CME.
show presence global	Displays configuration information about the presence service.

bulk

To set bulk registration for E.164 numbers that will register with SIP proxy server, use the **bulk** command in voice register global configuration mode. To disable bulk registration, use the **no** form of this command.

bulk *number-pattern*

no bulk

Syntax Description

number-pattern	A sequence of digits including wild card character.

Defaults

.Bulk registration is disabled.

Command Modes

Voice register global configuration.

Command History

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

Usage Guidelines

This command allows you to configure bulk registration for registering a block of phone numbers with an external registrar so that calls can be routed to Cisco CME from the SIP network.

Numbers that match the number pattern defined by using the **bulk** command register with the external registrar. The block of numbers that is registered can include any phone that is attached to Cisco CME using SIP or SCCP, or any analog phone that is directly attached to a Cisco router FXS port.

A number can contain one or more periods (.) as wildcard characters that will match any dialed number in that position. For example, 51.. rings when 5100 is dialed, when 5101 is dialed, and so forth.

The external registrar is configured by using the **registrar server** command under the SIP user-agent configuration mode.

Examples

The following example shows how to specify that numbers matching 1235 and any other dialed number in the next four positions, be routed to the Cisco CME from the SIP network.

```
Router(config)# voice register global
Router(conf-register-global)# mode cme
Router(conf-register-global)# bulk 1235...
```

Related Commands

Command	Description
mode (voice register global)	Enables the mode for provisioning SIP phones in a Cisco CallManager Express (Cisco CME) system.
no reg (voice register dn)	Specifies that a directory number in a SIP Cisco CallManager Express (Cisco CME) system not register with an external proxy server

Command	Description
no reg (voice hunt-group)	Specifies that a pilot number for a voice hunt group not register with an external proxy server
registrar	Enables SIP registrar functionality.

bulk-speed-dial list

To enable use of a bulk speed-dial list, use the **bulk-speed-dial list** command in ephone or telephony-service configuration mode. To remove the list, use the **no** form of this command.

bulk-speed-dial list *list-id location*

no bulk-speed-dial list *list-id*

Syntax Description

list-id	Digit that identifies the list to be used. Range is from 0 to 9.
location	Location of the bulk speed-dial list. Valid storage locations are TFTP, HTTP, and Flash memory.

Command Default

No default behavior or values

Command Modes

Ephone configuration
Telephony-service configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS 12.4(9)T.

Usage Guidelines

This command in telephony-service configuration mode enables a bulk speed-dial list on a per-system basis for all phones in Cisco Unified CME. This command in ephone configuration mode enables a bulk speed-dial list for a particular phone in Cisco Unified CME.

Bulk speed-dial lists must contain only comma-separated data. Typically, speed-dial lists are text files saved with the.txt file extension or Microsoft® Office Excel tables saved as .csv files.

Each list contains entries of speed-dial codes and the associated phone numbers to be dialed. Each entry in a list must appear on a separate line. Fields in each entry are separated by commas (,). A line that begins with a semicolon (;) is a comment and is ignored by Cisco Unified CME. Each entry in a list can include the following fields. For information about each field, see Table 1.

index,*digits*,[*name*],[**hide**],[**append**]

Table 1 explains the fields in a bulk speed-dial list entry.

Table 1 Bulk Speed-Dial List Entry

Field	Description
index	Zero-filled number that uniquely identifies this index entry. Maximum length: 4 digits. All index entries must be the same length.
digits	Telephone number to dialed. Represents a fully qualified E.164 number. Use a comma (,) to represent a one-second pause.
name	(Optional) Alphanumeric string to identify a name, up to 30 characters.
hide	(Optional) Enter **hide** to block the display of the dialed number.
append	(Optional) Enter **append** to allow additional digits to be appended to this number when dialed.

The following is a sample bulk speed-dial list:

```
01,5550140,voicemail,hide,append
90,914085550153,Cisco extension,hide,append
11,9911,emergency,hide,
91,9911,emergency,hide,
08,110,Paging,,append
```

The software does not automatically detect changes to the list files. If you modify a bulk speed-dial list that is enabled at a global or phone level, explicitly disable the list using the **no** form of this command, then enable the modified list. If the same list is enabled for more than one phone, disable the list on each phone using the **no** form of this command in ephone configuration mode, then enable the modified list per phone.

Use the **bulk speed-dial prefix** command to change the prefix code that a phone user must dial to access speed-dial numbers from a bulk speed-dial list. The default prefix is # (pound sign).

If a bulk speed-dial list is enabled using this command in telephony-service configuration mode and is also enable using this command in ephone configuration mode, the list enabled in ephone configuration mode takes precedence over the list at the global level for a given prefix. However, if the prefix used at the global level is different than the prefix used at the phone level, the lists are treated as separate lists - each list being associated with a different prefix, and at the phone level, you can access both lists.

Bulk speed dial is not supported on FXO trunk lines.

Use the **show telephony-service bulk-speed-dial** to display information about bulk speed-dial lists that are configured in Cisco Unified CME.

Examples

The following example shows that the default global bulk speed-dial prefix is changed to #7 and bulk speed-dial list 6 is enabled at a global level for all phones. To place a call to an entry in tise bulk speed-dial list, the phone user must first dial #7, followed by the list-id (6), then the index number for the entry to be called. This example also shows that bulk speed-dial list 7, with the default prefix unchanged, is enabled for ephone 2 only.

```
telephony-service
 bulk-speed-dial list 6 flash:sd_dept_01_1_87.txt
 bulk-speed-dial prefix #7

ephone-dn 3
```

```
 number 2555

ephone-dn 4
 number 2557

ephone 2
 button 1:3 2:4
 bulk-speed-dial list 7 flash:lmi_sd_list_08_24_95.csv
```

Related Commands

Command	Description
bulk-speed-dial prefix	Sets the prefix code that phone users dial to access speed-dial numbers from a bulk speed-dial list that is enabled system-wide.
show telephony-service bulk-speed-dial	Displays information about bulk speed-dial lists that are configured in Cisco Unified CME.

bulk-speed-dial prefix

To set the prefix code that phone users dial to access speed-dial numbers from a global bulk speed-dial list, use the **bulk-speed-dial prefix** command in telephony-service configuration mode. To return the prefix code to the default, use the **no** form of this command.

bulk-speed-dial prefix *prefix-code*

no bulk-speed-dial-prefix

Syntax Description

prefix-code	One to four-character access code for speed dial. Default is #.

Command Default

The default prefix code (pound sign [#]) is used.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS 12.4(9)T.

Usage Guidelines

This command changes the prefix code that a phone user must dial to access speed-dial numbers from a speed-dial list that is enabled using the **bulk-speed-dial list** command in telephony-service configuration mode. The default prefix is # (pound sign).

If a bulk speed-dial list is enabled using this command in telephony-service configuration mode and is also enable using this command in ephone configuration mode, the list enabled in ephone configuration mode takes precedence over the list at the global level for a given prefix. However, if the prefix used at the global level is different than the prefix used at the phone level, the lists are treated as separate lists - each list being associated with a different prefix, and at the phone level, you can access both lists.

Use the **show telephony-service bulk-speed-dial** to display information about bulk speed-dial lists that are configured in Cisco Unified CME.

Examples

The following example changes the default bulk speed-dial prefix to #7 and enables global bulk speed-dial list number 6 for all phones. It also enables a personal bulk speed-dial list for ephone 2. In this example, ephone 2 can access all of the numbers in both lists because each list is assigned a different prefix (# and #7).

```
telephony-service
 bulk-speed-dial list 6 flash:sd_dept_01_1_87.txt
 bulk-speed-dial prefix #7

ephone-dn 3
 number 2555
```

```
ephone-dn 4
 number 2557

ephone 2
 button 1:3 2:4
 bulk-speed-dial list 7 flash:lmi_sd_list_08_24_95.csv
```

Related Commands

Command	Description
bulk-speed-dial list	Enables a bulk speed-dial list.
show telephony-service bulk-speed-dial	Displays information about bulk speed-dial lists that are configured in Cisco Unified CME.

button

To associate ephone-dns with individual buttons on a Cisco Unified IP phone and to specify line type or ring behavior, use the **button** command in ephone configuration mode. To remove an ephone-dn association from a button, use the **no** form of this command.

button *button-number*{*separator*}*dn-tag* [**,***dn-tag*...] [*button-number*{**x**}*overlay-button-number*] [*button-number*...]

no button *button-number*{*separator*}*dn-tag* [**,***dn-tag*...] [*button-number*{**x**}*overlay-button-number*] [*button-number*...]

Syntax Description

button-number	Number of a line button on a Cisco Unified IP phone that is to be associated with an extension (ephone-dn). The maximum number of button–ephone-dn pairs is determined by the phone type. **Note** The Cisco Unified IP Phone 7910G has only one physical line button, but you can assign it two button–ephone-dn pairs.
separator	Single character that denotes the characteristics to be associated with this phone button. Valid entries are as follows: • **:** (colon)—Normal ring. For incoming calls on this extension, the phone produces audible ringing, a flashing ((< icon in the phone display, and a flashing red light on the handset. On the Cisco IP Phone 7914 Expansion Module, a flashing yellow light also accompanies incoming calls. • **b**—Beep but no ring. Audible ring is suppressed for incoming calls, but call-waiting beeps are allowed. Visible cues are the same as those described for a normal ring. • **c**—Call waiting. Provides call waiting for secondary calls to an overlaid ephone-dn. See also the **o** keyword. • **f**—Feature ring. Differentiates incoming calls on a special line from incoming calls on other lines on the phone. The feature-ring cadence is a triple pulse, as opposed to a single pulse for normal internal calls and a double pulse for normal external calls.

	• **m**—Monitor mode for a shared line. Visible line status indicates in-use or not. Line cannot be used on this phone for incoming or outgoing calls. • **o**—Overlay line. Multiple ephone-dns share a single button, up to a maximum of 25 on a button. See also the **c** keyword. • **s**—Silent ring. Audible ring and call-waiting beep are suppressed for incoming calls. The only visible cue is a flashing ((< icon in the phone display. **Note** In Cisco IOS Release 12.4(4)XC and later releases, the silent ringing behavior is overridden during active night-service periods. Silent ringing does not apply during designated night-service periods when the **s** keyword is used. • **w**—Watch mode for all lines on the phone for which this directory number is the primary line. Visible line status indicates whether watched phone is idle or not.
dn-tag	Ephone-dn tag that was previously defined using the **ephone-dn** command. When used with the **c** and **o** keywords, the *dn-tag* argument can contain up to 25 individual dn-tags, separated by commas.
x	Separator that creates an overlay rollover button. When the overlay button specified in this command is occupied by an active call, a second call to one of its ephone-dns will appear on this button. This button is also known as an overlay expansion button.
overlay-button-number	Number of the overlay button that should overflow to this button.

Command Default

No buttons are defined for an ephone.

Command Modes

Ephone configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco ITS 1.0	This command was introduced on the Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(2)XT	Cisco ITS 2.0	This command was implemented on the Cisco 1750 and Cisco 1751.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	Cisco ITS 2.0	This command was implemented on the Cisco 2600-XM and Cisco 2691.
12.2(11)T	Cisco ITS 2.01	This command was implemented on the Cisco 1760.
12.2(11)YT	Cisco ITS 2.1	The **b** and **s** keywords were added.
12.2(15)ZJ	Cisco CME 3.0	The **f**, **m**, and **o** keywords were added.

Cisco IOS Release	Cisco Product	Modification
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.3(11)XL	Cisco CME 3.2.1	The **c** keyword was added and the ability to use the **m** keyword to monitor call-park slots was added.
12.3(14)T	Cisco CME 3.3	This command was integrated into Cisco IOS Release 12.3(14)T.
12.4(4)XC	Cisco Unified CME 4.0	The **x** keyword was added and the number of ephone-dns that can be overlaid on a single button with the **o** or **c** keyword was increased from 10 to 25. The interaction between the keyword and night service was modified; silent ringing is overridden when night service is active.
12.4(9)T	Cisco Unified CME 4.0	The modifications made to this command were integrated into Cisco IOS Release 12.4(9)T.
12.4(11)XJ	Cisco Unified CME 4.1	The **w** keyword was added.
12.4(15)T	Cisco Unified CME 4.1	This command with the **w** keyword was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

The **button** command assigns telephone extensions to Cisco Unified IP phones by associating a button number with one or more directory numbers (ephone-dns).

Note After adding or changing a phone button configuration using this command, you must perform a quick reboot of the phone using the **restart** command.

Telephone services such as call waiting and three-party conferences require a minimum of two phone lines (ephone-dns defined with the **ephone-dn** command) to be available and configured on a Cisco IP phone.

The Cisco Unified IP Phone 7910G has only one physical line button. To support call waiting and three-party conferences on a Cisco Unified IP Phone 7910G, a second (hidden) line is required. This line cannot be selected directly using a line button. You can access the second line when you press the Conference button. You can also support multiple-call services using the **ephone-dn dual-line** configuration option.

Feature Ring (f)

A feature ring is a third type of ring cadence in addition to internal call and external call ring cadences. For example, an internal call in the United States rings for 2 seconds on and 4 seconds off (single-pulse ring), and an external call rings for 0.4 seconds on, 0.2 seconds off, 0.4 seconds on, and 0.2 seconds off (double-pulse ring). A feature ring is a triple-pulse ring. The purpose of associating a feature ring with a line button is to be able to identify from a distance a special line that is ringing on a multiline phone.

Monitor Mode (m)

A line button set in monitor mode on one phone provides visual line status for a line that also appears on another phone. When monitor mode is set for a button with a shared line, the line status indicates that the shared line is either idle or in use. The line and line button are available in monitor mode for visual status only. Calls cannot be made or received using a line button that has been set in monitor mode. Incoming calls on a line button that is in monitor mode do not ring and do not display caller ID or call-waiting caller ID.

Monitor mode is intended to be used only in the context of shared lines so that a receptionist can visually monitor the in-use status of several users' phone extensions (for example, as a busy-lamp field). To monitor all lines on an individual phone so that a receptionist can visually monitor the in-use status of that phone, see the "Watch Mode (w)" section on page 67.

The line button for a monitored line can also be used as a direct-station-select for a call transfer when the monitored line is in an idle state. In this case, the receptionist who transfers a call from a normal line can press the Transfer button and then press the line button of the monitored line, causing the call to be transferred to the phone number of the monitored line.

Overlay (o)

Overlay lines are ephone-dns that share a single button on a multibutton phone. When more than one incoming call arrives on lines that are set on a single button, the line (ephone-dn) that is the leftmost in the **button** command list is the primary line and is given the highest priority. If this call is answered by another phone or if the caller hangs up, the phone selects the next line in its overlay set to present as the ringing call. The caller ID display updates to show the caller ID for the currently presented call.

Ephone-dns that are part of an overlay set can be single-line ephone-dns or dual-line ephone-dns, but the set must contain either all single-line ephone-dns or all dual-line ephone-dns, and not a mixture of the two.

The primary ephone-dn on each phone in a shared-line overlay set should be unique to the phone being configured to guarantee that there is a line available for outgoing calls, and to ensure that the phone user can obtain dial-tone even when there are no idle lines available in the rest of the shared-line overlay set. Use a unique ephone-dn in this manner to provide for a unique calling party identity on outbound calls made by the phone so that the called user can see which specific phone is calling.

The name of the first ephone-dn in the overlay set is not displayed because it is the default ephone-dn for calls to the phone, and the name or number is permanently displayed next to the phone's button. For example, if there are ten ephone-dns in an overlay set, only the last nine ephone-dns are displayed when calls are made to them.

For more information, see the "Configuring Call Coverage Features" module in the *Cisco Unified CME Administrator Guide*.

Overlay Ephone-dns with Call Waiting (c)

The configuration for the overlaid ephone-dns with call waiting (keyword **c**) and without call waiting (keyword **o**) is the same.

Ephone-dns accept call interruptions, such as call waiting, by default. For call waiting to work, the default must be active. To ensure that this is the case, remove the **no call-waiting beep accept** command from the configurations of ephone-dns for which you want to use call waiting.

In Cisco Unified CME 4.0(3), the Cisco Unified IP Phone 7931G cannot support overlays that contain ephone-dn configured for dual-line mode.

Note In general, all the ephone-dns within an overlay must be of the same type (dual-line or single line mode).

For more information, see "Configuring Call Coverage Features" module in the *Cisco Unified CME Administrator Guide*.

Silent Ring (s)

You can configure silent ring on any type of phone. However, you typically set silent ring only on buttons of a phone with multiple lines, such as a Cisco Unified IP Phone 7940, Cisco Unified IP Phones 7960 and 7960G, or a Cisco Unified IP Phone 7914 Expansion Module. The only visible cue is a flashing ((< icon in the phone display.

If you configure a button to have a silent ring, you will not hear a call-waiting beep or call-waiting ring regardless of whether the ephone-dn associated with the button is configured to generate a call-waiting beep or call-waiting ring.

In Cisco IOS Release 12.4(4)XC and later releases, the silent ringing behavior is overridden during active night-service periods. Silent ringing does not apply during designated night-service periods when the **s** keyword is used.

Watch Mode (w)

A line button that is configured for watch mode on one phone provides visual line status for all lines on another phone (watched phone) for which the watched directory number is the primary line. Watched mode allows a phone user, such as a receptionist, to visually monitor the in-use status of an individual phone. The line and line button on the watching phone are available in watch mode for visual status only. Calls cannot be made or received using a line button that has been set in watch mode. Incoming calls on a line button that is in watch mode do not ring and do not display caller ID or call-waiting caller ID.

If any of the following conditions are true, the status of the line button in watch mode indicates that the watched phone is in-use:

- Watched phone is off-hook
- Watched phone is not registered
- Watched phone is in the do-not-disturb (DND) mode
- Watched directory number is not idle

Note If the watched directory number is a shared line and the shared line is not idle on any phone with which it is associated, then in the context of watch mode, the status of the line button indicates that the *watched phone* is in use.

For best results in terms of monitoring the status of an individual phone based on a watched directory number, the directory number to be configured for watch mode should not be a shared line. To monitor a shared line so that a receptionist can visually monitor the in-use status of several users' phone extensions, see the "Monitor Mode (m)" section on page 65.

If the watched directory number is associated with several phones, then the watched phone is the one on which the watched directory number is on button 1 *or* the one on which the watched directory number is on the button that is configured by using the **auto-line** command, with auto-line having priority.

If more than one phone meets the criteria for primary line as described above, then the watched phone is the first phone that that meets the criteria. Typically, that is the phone with the lowest ephone tag value. However, if the watched directory number is configured on button 1 of ephone 1 and the same directory number is also configured on button 3 with "auto-line 3" of ephone 24, then ephone 24 is the watched phone because the auto-line configuration has priority.

The line button for a watched phone can also be used as a direct-station-select for a call transfer when the watched phone is idle. In this case, the phone user who transfers a call from a normal line can press the Transfer button and then press the line button of the watched directory number, causing the call to be transferred to the phone number associated with the watched directory number.

Expansion Buttons for Overlay Ephone-dns (x)

This feature works to expand coverage for an overlay button that has been configured using the **o** separator in the **button** command. Overlay buttons with call waiting that use the **c** separator in the **button** command are not eligible for overlay rollover.

Examples

The following example assigns four button numbers on the phone to ephone-dn tags, and button 4 has a silent ring:

```
ephone-dn 1
 number 233

ephone-dn 4
 number 234

ephone-dn 16
 number 235

ephone-dn 19
 number 236

ephone 1
 button 1:1 2:4 3:16 4s19
```

The following example shows three phones that each have three instances of extension number 1001 overlaid onto a single button, which allows three simultaneous calls to extension 1001. The first call arrives on ephone-dn 1 and rings button 1 on all three phones. The call is answered on ephone 10. A second call for 1001 hunts onto ephone-dn 2 and rings on the remaining two ephones, ephones 11 and 12, and is answered by ephone 12. A third call to 1001 hunts onto ephone-dn 3 and rings on ephone 12, where it is answered. This configuration creates a three-way shared line across three IP phones and can handle three simultaneous calls to the same telephone number. Note that if ephone 12 is busy, the third call will go to voice mail (7000). Note also that if you wanted call waiting, you would use the same configuration, except for the use of the **c** keyword instead of the **o** keyword. Ephone 10 uses call waiting.

```
ephone-dn 1
 number 1001
 no huntstop
!
ephone-dn 2
 number 1001
 no huntstop
 preference 1
!
ephone-dn 3
 number 1001
 preference 2
 call-forward busy 7000
!
! The next ephone configuration includes the first instance of shared line 1001.
ephone 10
 mac-address 1111.2222.3333
 button 1c1,2,3
!
! The next ephone configuration includes the second instance of shared line 1001.
ephone 11
 mac-address 1111.2222.4444
 button 1o1,2,3
!
! The next ephone configuration includes the third instance of shared line 1001.
ephone 12
```

```
 mac-address 1111.2222.555
 button 1o1,2,3
```

The following is an example of unique ephone-dn as the primary dn in a simple shared-line overlay configuration. The no huntstop command is configured for all the ephone-dns except ephone-dn 12, the last one in the overlay set. Because the ephone-dns are dual-line dns, the huntstop-channel command is also configured to ensure that the second channel remains free for outgoing calls and for conferencing.

```
ephone-dn 1 dual-line
 number 101
 huntstop-channel
!
ephone-dn 2 dual-line
 number 102
 huntstop-channel
!
ephone-dn 10 dual-line
 number 201
 no hunsttop
 huntstop-channel
!
ephone-dn 11 dual-line
 number 201
 no hunsttop
 huntstop-channel
!
ephone-dn 12 dual-line
number 201
huntstop-channel
!
!The next ephone configuration includes (unique) ephone-dn 1 as the primary line in a
shared-line overlay

ephone 1
mac-address 1111.1111.1111
button 1o1,10,11,12
!
!The next ephone configuration includes (unique) ephone-dn 2 as the primary line in
another shared-line overlay
ephone 2
mac-address 2222.2222.2222
button 1o2,10,11,12
```

Shared-line overlays can be constructed using the "button o" or "button c" formats depending upon whether call-waiting is desired. The following example shows an ephone configuration that enables call-waiting (c) in a shared-line overlay:

```
ephone 1
 mac-address 1111.1111.1111
 button 1c1,10,11,12
!
 ephone 2
 mac-address 2222.2222.2222
 button 1c2,10,11,12
```

The following example configures a “3x3” shared-line setup for three ephones and nine shared lines (ephone-dns 20 through 28). Each ephone has a unique ephone-dn for each of its three buttons (ephone-dns 1 to 3, ephone-dns 4 to 6, and ephone-dns 7 to 9). The remaining ephone-dns are shared among the three phones. Three phones with three buttons each can take nine calls. The overflow buttons provide the ability for an incoming call to ring on the first available button on each phone.

```
ephone 1
 button 1o1,2,3,20,21,22,23,24,25,26,27,28 2x1 3x1
```

```
ephone 2
 button 1o4,5,6,20,21,22,23,24,25,26,27,28 2x1 3x1

ephone 3
 button 1o7,8,9,20,21,22,23,24,25,26,27,28 2x1 3x1
```

Related Commands

Command	Description
call-waiting beep	Allows phone buttons to accept or generate call-waiting beeps.
restart (ephone)	Performs a fast reboot of a single phone associated with a Cisco Unified CME router.
restart (telephony-service)	Performs a fast reboot of one or all phones associated with a Cisco Unified CME router.
show ephone	Displays information about ephones and the corresponding Cisco Unified IP phones.
show ephone overlay	Displays the configuration and current status of registered overlay ephone-dns.

button-layout

To configure a fixed set of line or feature buttons in an ephone-template which can then be applied to a supported IP phone in Cisco Unified CME, use the **button-layout set** command in ephone-template configuration mode. To disable the feature buttons set and change the action of the buttons on IP phones, use the **no** form of this command.

button-layout *phone-type* **{1** | *2***}**

no button-layout

Syntax Description

phone-type	Type of IP phone. The following choices are valid: • **7931**—Cisco Unified IP Phone 7931.
1	Number of fixed line or feature set containing the following buttons: • Button 24—Menu. • Button 23—Headset.
2	Number of fixed line or feature set containing the following buttons: • Button 24—Menu. • Button 23—Headset. • Button 22—Directories. • Button 21—Messages.

Command Default

No fixed set of line or feature buttons are defined.

Command Modes

Ephone-template configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(6)XE	Cisco Unified CME 4.0(2)	This command was introduced.
12.4(4)XC4	Cisco Unified CME 4.0(3)	This command was introduced.
12.4(11)T	Cisco Unified CME 4.0(3)	This command was integrated into Cisco IOS Release 12.4(11)T.

Usage Guidelines

Use this command to configure either Set 1 or Set 2 in an ephone-template which can then be applied to an individual Cisco Unified IP Phone 7931G in Cisco Unified CME.

After a template has been created, you can apply it to an ephone using the **ephone-template** command in ephone configuration mode. You cannot apply more than one ephone template to an ephone.

To view your ephone-template configurations, use the **show telephony-service ephone-template** command.

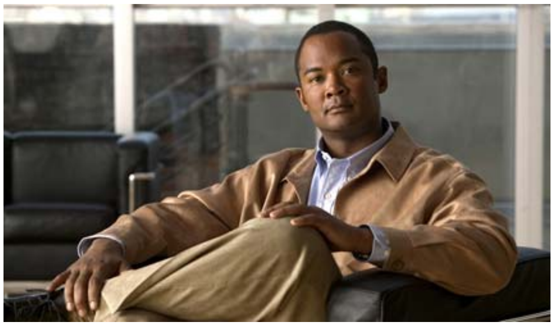
Examples

The following example shows how to create ephone-template 12, containing set 2 feature buttons, and apply the template to ephone 36.

```
Router(config)# ephone-template 12
Router(config-ephone-template)# button-layout set 2
Router(config-ephone-template)# exit
Router(config)# ephone 36
Router(config-ephone)# ephone-template 12
Router(config-ephone)# exit
Router(config)# telephony-service
Router(config-telephony)# create cnf-files
...
```

Related Commands

Command	Description
show telephon-service ephone-template	Displays ephone-template configurations.



Cisco Unified CME Commands: C

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This chapter contains commands to configure and maintain Cisco Unified Communications Manager Express (formally known as Cisco Unified CallManager Express). The commands are presented in alphabetical order. Some commands required for configuring Cisco Unified Communications Manager Express (Cisco Unified CME) may be found in other Cisco IOS command references. Use the command reference master index or search online to find these commands.

call application voice aa-hunt

Effective with Cisco IOS Release 12.3(14)T and later, the **call application voice aa-hunt** command is replaced by the **param aa-hunt** command. See the **param aa-hunt** command for more information.

To declare a Cisco CallManager Express (Cisco CME) basic automatic call distribution (B-ACD) menu number and associate it with the pilot number of an ephone hunt group, use the **call application voice aa-hunt** command in global configuration mode. To remove the menu number and the ephone hunt group pilot number, use the **no** form of this command.

call application voice *application-name* **aa-hunt***menu-number pilot-number*

no call application voice *application-name* **aa-hunt***menu-number pilot-number*

Syntax Description

menu-number	Number that callers must dial to reach the pilot number of an ephone hunt group. The range is from 1to 10. The default is 1.
application-name	Application name given to the call queue script in the **call application voice** command.
pilot-number	Ephone hunt group pilot number.

Command Default

Cisco CME B-ACD menu number 1 is configured, but it has no pilot number.

Command Modes

Global configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)XL	Cisco CME 3.2.1	This command was introduced.
12.3(14)T	Cisco CME 3.2.2	This command was integrated into Cisco IOS Release 12.3(14)T.
12.3(14)T	Cisco CME 3.3	This command was replaced with the **param aa-hunt** command.

Usage Guidelines

This command is used only with with a version of the Cisco CME B-ACD script that is earlier than 2.1.0.0 and is used only with with a version of the Cisco CME B-ACD script that is earlier than 2.1.0.0 and is valid only for the configuration of Cisco CME B-ACD call queue scripts. Up to three menu options are allowed per call queue script. You can use any of the allowable numbers in any order.

The **call application voice aa-hunt** command allows each of the menu options to be associated with the pilot number of an ephone hunt group. The menu options are announced by the en_bacd_options_menu.au audio file, which you can rerecord. When a caller presses a number, the call will go to the pilot number of an ephone hunt group so it can be transferred to one of the ephone hunt group's ephone-dns. It will not go to any other ephone hunt group. The order in which ephone-dns are selected depends on the ephone hunt group's search method, which is configured with the **ephone-hunt** command, and whether an ephone-dn is busy or not.

If only one menu option is configured, callers will hear a greeting and be transferred directly to the pilot number of the corresponding ephone hunt group. They do not have to enter a number.

The highest aa-hunt number will automatically be set to zero (0) for the operator. In the following example, aa-hunt8 supports the menu option of 0 and 8.

```
call application voice queue aa-hunt1 1111
call application voice queue aa-hunt3 3333
call application voice queue aa-hunt8 8888
```

If a phone user presses 0 or 8, their call be sent to pilot number 3333.

For any configuration changes to take effect, you must reload the Cisco CME B-ACD scripts.

Examples

The following example associates three menu numbers with three pilot numbers of three ephone hunt groups. Pilot number 1111 is for ephone hunt group 1 (sales); 2222 is for ephone hunt group 2 (customer service); and 3333 is for ephone hunt group 3 (operator). If sales is selected from the AA menu, the call will be transferred to 1111 and sent to ephone hunt group 1's available ephone-dns (2001, 2002, 2003, 2004, 2005, 2006).

```
Router(config)# ephone-hunt 1 peer
Router(config-ephone-hunt)# pilot 1111
Router(config-ephone-hunt)# list 1001, 1002, 1003, 1004, 1005, 1006, 1007, 1008, 1009,
1010

Router(config)# ephone-hunt 2 peer
Router(config-ephone-hunt)# pilot 2222
Router(config-ephone-hunt)# list 2001, 2002, 2003, 2004, 2005, 2006

Router(config)# ephone-hunt 3 peer
Router(config-ephone-hunt)# pilot 3333
Router(config-ephone-hunt)# list 3001, 3002, 3003, 3004

Router(config)# call application voice queue flash:app-b-acd-x.x.x.x.tcl
Router(config)# call application voice queue aa-hunt1 1111
Router(config)# call application voice queue aa-hunt2 2222
Router(config)# call application voice queue aa-hunt3 3333
```

Related Commands

Command	Description
call application voice	Defines a name for a voice application and specifies the location of the Tcl or VoiceXML document to load for this application.
call application voice aa-pilot	Associates an ephone hunt group with the Cisco CME basic service's AA script by declaring the group's pilot number.
call application voice welcome-prompt	Assigns an audio file that is used by a Cisco CME B-ACD AA script for the welcome greeting.
ephone-dn	Enters ephone-dn configuration mode for the purposes of creating and configuring an extension for a Cisco IP phone line.
ephone-hunt	Enters ephone-hunt configuration mode for the purposes of creating and configuring a hunt group for use in a Cisco CME system.
pilot	Defines the ephone-dn that callers dial to reach a Cisco CME ephone hunt group.

call application voice aa-name

Effective with Cisco IOS Release 12.3(14)T and later, the **call application voice aa-name** command is not available in Cisco IOS software.

To associate the queue script for Cisco CallManager Express (Cisco CME) basic automatic call distribution (B-ACD) with the Cisco CME B-ACD auto-attendant (AA) script, use the **call application voice aa-name** command in global configuration mode. To remove the queue script and AA script association, use the **no** form of this command.

call application voice *application-name* **aa-name** *aa-script-name*

no call application voice *application-name* **aa-name** *aa-script-name*

Syntax Description

application-name	Application name given to the call queue script in the **call application voice** command.
aa-script-name	Application name given to the AA script in the **call application voice** command.

Command Default

No call queue script and AA script association is configured.

Command Modes

Global configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)XL	Cisco CME 3.2.1	This command was introduced.
12.3(14)T	Cisco CME 3.2.2	This command was integrated into Cisco IOS Release 12.3(14)T.
12.3(14)T	Cisco CME 3.3	This command was replaced with the **param aa-name** command.

Usage Guidelines

This command is used only with with a version of the Cisco CME B-ACD script that is earlier than 2.1.0.0 and is valid only for the configuration of Cisco CME B-ACD call queue scripts. Only one AA script can be associated with one call queue script.

For any configuration changes to take effect, you must reload the Cisco CME B-ACD scripts.

Examples

The following example associates a call queue script with an AA script:

```
Router(config)# call application voice aa flash:app-b-acd-aa-x.x.x.x.tcl
Router(config)# call application voice queue flash:app-b-acd-x.x.x.x.tcl
Router(config)# call application voice queue aa-name aa
```

Related Commands

Command	Description
call application voice	Defines a name for a voice application and specifies the location of the Tcl or VoiceXML document to load for this application.
call application voice service-name	Associates a Cisco CME B-ACD AA script with a Cisco CME B-ACD call queue script.

call application voice aa-pilot

Effective with Cisco IOS Release 12.3(14)T and later, the **call application voice aa-pilot** command is replaced by the **param aa-pilot** command. See the **param aa-pilot** command for more information.

To assign a pilot number to the Cisco CallManager Express (Cisco CME) basic automatic call distribution (B-ACD) service, use the **call application voice aa-pilot** command in global configuration mode. To remove the Cisco CME B-ACD pilot number, use the **no** form of this command.

call application voice *application-name* **aa-pilot** *pilot-number*

no call application voice *application-name* **aa-pilot** *pilot-number*

Syntax Description

application-name	Application name given to the auto-attendant (AA) script in the **call application voice** command.
pilot-number	Pilot number for Cisco CME B-ACD.

Command Default

No Cisco CME B-ACD pilot number is configured.

Command Modes

Global configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)XL	Cisco CME 3.2.1	This command was introduced.
12.3(14)T	Cisco CME 3.2.2	This command was integrated into Cisco IOS Release 12.3(14)T.
12.3(14)T	Cisco CME 3.3	This command was replaced by the **param aa-pilot** command.

Usage Guidelines

This command is used only with with a version of the Cisco CME B-ACD script that is earlier than 2.1.0.0 and is valid only for the configuration of Cisco CME B-ACD AA scripts. Only one pilot number can be used for each Cisco CME B-ACD service, and the voice ports handling AA must have dial peers that will send calls to the pilot number.

For any configuration changes to take effect, you must reload the Cisco CME B-ACD scripts.

Examples

The following example assigns 8005550100 as the pilot number to the Cisco CME B-ACD service. Included in this example is the dial-peer configuration for the pilot number.

```
Router(config)# call application voice aa flash:app-b-acd-aa-x.x.x.x.tcl
Router(config)# call application voice aa aa-pilot 8005550100

Router(config)# dial-peer voice 1000 pots
Router(config)# incoming pilot number 8005550100
Router(config)# application aa
Router(config)# direct-inward-dial
```

```
Router(config)# port 1/0:23
Router(config)# forward digits-all
Router(config)# call application voice aa aa-pilot 80055501
```

Related Commands

Command	Description
call application voice	Defines a name for a voice application and specifies the location of the Tcl or VoiceXML document to load for this application.
dial-peer voice	Defines a particular dial peer, specifies the method of voice encapsulation, and enters dial-peer configuration mode.
ephone-dn	Enters ephone-dn configuration mode for the purposes of creating and configuring an extension for a Cisco IP phone line.
ephone-hunt	Enters ephone-hunt configuration mode for the purposes of creating and configuring a hunt group for use in a Cisco CME system.

call application voice call-retry-timer

Effective with Cisco IOS Release 12.3(14)T and later, the **call application call-retry-timer** command is replaced by the **param call-retry-timer** command. See the **param call-retry-timer** command for more information.

To assign the length of time that calls to Cisco CallManager Express (Cisco CME) basic automatic call distribution (B-ACD) must wait before attempting to transfer to an ephone hunt group pilot number, use the **call application voice call-retry-timer** command in global configuration mode. To remove the retry time, use the **no** form of this command.

call application voice *application-name* **call-retry-timer** *seconds*

no call application voice *application-name* **call-retry-timer** *seconds*

Syntax Description

application-name	Application name given to the auto-attendant (AA) script in the **call application voice** command.
seconds	Number of seconds that a call must wait before attempting to transfer an ephone hunt pilot number or voice-mail pilot number. The range is from 5 to 30 seconds. The default is 15 seconds.

Command Default

The retry interval is 15 seconds.

Command Modes

Global configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)XL	Cisco CME 3.2.1	This command was introduced.
12.3(14)T	Cisco CME 3.2.2	This command was integrated into Cisco IOS Release 12.3(14)T.
12.3(14)T	Cisco CME 3.3	This command was replaced by the **param call-retry-timer** command

Usage Guidelines

This command is used only with with a version of the Cisco CME B-ACD script that is earlier than 2.1.0.0 and is valid only for the configuration of Cisco CME B-ACD AA scripts. The following sequence of events might occur:

- An outside call comes into a system configured with Cisco CME B-ACD.
- A menu option is selected that attempts to transfer the call to an ephone hunt group pilot number.
- All of the ephone hunt group's ephone-dns are busy.

In that case, the call will wait in a queue for the period of time set by the **call application voice call-retry-timer** command and retry to the pilot number.

For any configuration changes to take effect, you must reload the Cisco CME B-ACD scripts.

Examples

The following example shows a configuration that allows outside calls to Cisco CME B-ACD to retry an ephone hunt group pilot number every 30 seconds. The example shows the configuration for one ephone hunt group, which is presented by Cisco CME B-ACD menu as the sales department and uses a simple configuration. If a caller selects the sales menu option (**ephone-hunt 1**) and all of the ephone-dns configured in the **list** command (1001, 1002, 1003, 1004) are busy, the call will wait 30 seconds and then retry the pilot number (1111) until either an ephone-dn becomes available or a configured amount of time has elapsed (see the **call application voice max-time-call-retry** command).

```
Router(config)# ephone-hunt 1 peer
Router(config-ephone-hunt)# pilot 1111
Router(config-ephone-hunt)# list 1001, 1002, 1003, 1004

Router(config)# call application voice aa flash:app-b-acd-aa-x.x.x.x.tcl
Router(config)# call application voice aa call-retry-timer 30
```

Related Commands

Command	Description
ephone-dn	Enters ephone-dn configuration mode for the purposes of creating and configuring an extension for a Cisco IP phone line.
ephone-hunt	Enters ephone-hunt configuration mode for the purposes of creating and configuring a hunt group for use in a Cisco CME system.
call application voice	Defines a name for a voice application and specifies the location of the Tcl or VoiceXML document to load for this application.
call application voice aa-hunt	Declares a Cisco CME B-ACD menu number and associates it with the pilot number of an ephone hunt group.
call application voice aa-pilot	Associates an ephone hunt group with the Cisco CME basic service's AA script by declaring the group's pilot number
call application voice max-time-call-retry	Assigns the maximum length of time for which calls to Cisco CME B-ACD can stay in a call queue.

call application voice dial-by-extension-option

Effective with Cisco IOS Release 12.3(14)T and later, the **call application dial-by-extension-option** command is replaced by the **param dial-by-extension-option** command. See the **param dial-by-extension-option** command for more information.

To enable direct extension access and set the access number for Cisco CallManager Express (Cisco CME) basic automatic call distribution (B-ACD), use the **call application voice dial-by-extension-option** command in global configuration mode. To disable direct dial extension access and remove the access number, use the **no** form of this command.

call application voice *application-name* **dial-by-extension** *number*

no call application voice *application-name* **dial-by-extension** *number*

Syntax Description

application-name	Application name given to the auto-attendant (AA) script in the **call application voice** command.
number	The single digit that callers press to be able to enter an extension number from the AA menu. The range is from 1 to 10. There is no default.

Defaults

Direct dial access is disabled. No access number is configured.

Command Modes

Global configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)XL	Cisco CME 3.2.1	This command was introduced.
12.3(14)T	Cisco CME 3.2.2	This command was integrated into Cisco IOS Release 12.3(14)T.
12.3(14)T	Cisco CME 3.3	This command was replaced by the **param dial-by-extension-option** command.

Usage Guidelines

This command is used only with with a version of the Cisco CME B-ACD script that is earlier than 2.1.0.0 and is valid only for the configuration of Cisco CME B-ACD AA scripts. It enables the en_bacd_enter_dest.au audio file. The default announcement says, “Please enter the extension number you want to reach.” The **call application voice dial-by-extension-option** command also allows for the configuration of the number that callers must press before they can enter the extension number that they want to call.

Callers who select the extension access option can then dial any extension. If they dial an ephone hunt group ephone-dn or pilot number, their call will not be sent to the ephone hunt-group call queue.

Examples

The following example configures Cisco CME B-ACD to include an option that allows callers to press the number 4 so they can dial an extension number.

```
Router(config)# call application voice aa flash:app-b-acd-aa-x.x.x.x.tcl
Router(config)# call application voice aa dial-by-extension 4
```

Related Commands

Command	Description
call application voice	Defines a name for a voice application and specifies the location of the Tcl or VoiceXML document to load for this application.

call application voice drop-through-option

Cisco IOS Release 12.3(14)T and later releases support Cisco CME Basic Automatic Call Distribution (B-ACD) and Auto-Attendant (AA) Tcl scripts version 2.1.0.0 and greater. In these releases, the **call application voice drop-through-option** command has been replaced by the **param drop-through-option** command.

call application voice drop-through-prompt

Cisco IOS Release 12.3(14)T and later releases support Cisco CME Basic Automatic Call Distribution (B-ACD) and Auto-Attendant (AA) Tcl scripts version 2.1.0.0 and greater. In these releases, the **call application voice drop-through-prompt** command has been replaced by the **param drop-through-prompt** command.

call application voice handoff-string

Cisco IOS Release 12.3(14)T and later releases support Cisco CME Basic Automatic Call Distribution (B-ACD) and Auto-Attendant (AA) Tcl scripts version 2.1.0.0 and greater. In these releases, the **call application voice handoff-string** command has been replaced by the **param handoff-string** command.

call application voice max-extension-length

Cisco IOS Release 12.3(14)T and later releases support Cisco CME Basic Automatic Call Distribution (B-ACD) and Auto-Attendant (AA) Tcl scripts version 2.1.0.0 and greater. In these releases, the **call application voice max-extension-length** command has been replaced by the **param max-extension-length** command.

call application voice max-time-call-retry

Effective with Cisco IOS Release 12.3(14)T and later, the **call application voice max-time-call-retry** command is replaced by the **param max-time-call-retry** command. See the **param max-time-call-retry** command for more information.

To assign the maximum length of time for which calls to Cisco CallManager Express (Cisco CME) basic automatic call distribution (B-ACD) can stay in a call queue, use the **call application voice max-time-call-retry** command in global configuration mode. To remove the maximum length of time, use the **no** form of this command.

call application voice *application-name* **max-time-call-retry** *seconds*

no call application voice *application-name* **max-time-call-retry** *seconds*

Syntax Description

application-name	Application name given to the auto attendant (AA) script in the **call application voice** command.
seconds	Maximum length of time that the Cisco CME B-ACD AA script can keep redialing an ephone hunt group pilot number. The range is from 0 to 3600 seconds. The default is 600 seconds.

Command Default

The default maximum length of time that calls can stay in a call queue is 600 seconds.

Command Modes

Global configuration

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.3(11)XL	Cisco CME 3.2.1	This command was introduced.
12.3(14)T	Cisco CME 3.2.2	This command was integrated into Cisco IOS Release 12.3(14)T.
12.3(14)T	Cisco CME 3.3	This command was replaced by the **param max-time-call-retry** command.

Usage Guidelines

This command is used only with with a version of the Cisco CME B-ACD script that is earlier than 2.1.0.0 and is valid only for the configuration of Cisco CME B-ACD AA scripts. The **call application voice max-time-retry** command allows you set a time limit for the redialing of pilot numbers under the following circumstances:

- An outside call comes into a system configured with Cisco CME B-ACD.
- A menu option is selected that transfers the call to an ephone hunt-group pilot number.
- All of the ephone hunt group's ephone-dns are busy.
- The call is sent to a queue and tries the pilot number at intervals of time set by the **call application voice call-retry-timer** command.

When the time period set by the **call application voice max-call-retry** command expires, one of the following two events will occur:

- If a voice-mail pilot number has been configured in Cisco CME and mail boxes for hunt group pilot numbers have been configured in a voice-mail application, calls will be transferred to voice mail.
- If voice mail has not been configured, a default message will be played that says, “We are unable to take your call at this time. Please try again at a later time. Thank you for calling.”

Examples

In the following example, the length of time for which calls can try to reach ephone hunt group 1 and ephone hunt group 2 is 90 seconds. If a caller selects the AA menu option for either hunt group and all of its ephone-dns configured in the **list** command are busy, the call will keep retrying the ephone hunt group’s pilot number until one of the ephone-dns is available or 90 seconds has elapsed. When 90 seconds elapses, the call will go to voice mail.

```
Router(config)# ephone-hunt 1 peer
Router(config-ephone-hunt)# pilot 1111
Router(config-ephone-hunt)# list 1001, 1002, 1003, 1004

Router(config)# ephone-hunt 2peer
Router(config-ephone-hunt)# pilot 2222
Router(config-ephone-hunt)# list 2001, 2002, 2003, 2004

Router(config)# call application voice aa flash:app-b-acd-aa-x.x.x.x.tcl
Router(config)# call application voice aa max-call-retry-timer 90
```

Related Commands

Command	Description
call application voice	Defines a name for a voice application and specifies the location of the Tcl or VoiceXML document to load for this application.
call application voice call-retry-timer	Assigns the length of time that calls to Cisco CME B-ACD must wait before attempting to transfer to an ephone hunt-group pilot number or to voice mail.
call application voice max-time-vm-retry	Assigns the maximum number of times that calls to Cisco CME B-ACD can attempt to reach voice mail.
ephone-dn	Enters ephone-dn configuration mode for the purposes of creating and configuring an extension for a Cisco IP phone line.
ephone-hunt	Enters ephone-hunt configuration mode for the purposes of creating and configuring a hunt group for use in a Cisco CME system.

call application voice max-time-vm-retry

Cisco IOS Release 12.3(14)T and later releases support Cisco CME Basic Automatic Call Distribution (B-ACD) and Auto-Attendant (AA) Tcl scripts version 2.1.0.0 and greater. In these releases, the **call application voice max-time-vm-retry** command has been replaced by the **param max-time-vm-retry** command.

call application voice number-of-hunt-grps

Effective with Cisco IOS Release 12.3(14)T and later, the **call application voice number-of-hunt-grps** command is replaced by the **param number-of-hunt-grps** command. See the **param number-of-hunt-grps** command for more information.

To declare the maximum number of ephone hunt-group menus supported by Cisco CallManager Express (Cisco CME) basic automatic call distribution (B-ACD), use the **call application voice number-of-hunt-grps** command in global configuration mode. To remove the maximum number of ephone hunt-group menus supported by Cisco CME B-ACD, use the **no** form of this command.

call application voice *application-name* **number-of-hunt-grps** *number*

no call application voice *application-name* **number-of-hunt-grps** *number*

Syntax Description

application-name	Application name given to the auto-attendant (AA) script in the **call application voice** command.
number	Number of hunt groups used by the Cisco CME B-ACD AA script and call queue script. The range is from 1 to 3. The default is 3.

Command Default

Three ephone hunt-group menus are supported by Cisco CME B-ACD.

Command Modes

Global configuration

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.3(11)XL	Cisco CME 3.2.1	This command was introduced.
12.3(14)T	Cisco CME 3.2.2	This command was integrated into Cisco IOS Release 12.3(14)T.
12.3(14)T	Cisco CME 3.3	This command was replaced by the **param number-of-hunt-grps** command.

Usage Guidelines

This command is used only with with a version of the Cisco CME B-ACD script that is earlier than 2.1.0.0 and is valid only for the configuration of Cisco CME B-ACD AA scripts. The *number* argument declares the number of ephone hunt groups only. The menu option for direct extension access (see the **call application voice dial-by-extension-option** command) is not included.

Examples

The following example configures a Cisco CME B-ACD call queue script to use three ephone hunt groups and one direct extension access number, making the *number* argument in the **call application voice number-of-hunt-grps** equal to 3. The **ephone-hunt** command is used to configure the three ephone hunt groups. The **call application voice dial-by-extension-option** command is used to enable direct extension access and set the access number to 1.

```
Router(config)# ephone-hunt 1 peer
Router(config-ephone-hunt)# pilot 1111
```

```
Router(config-ephone-hunt)# list 1001, 1002, 1003, 1004, 1005, 1006, 1007, 1008, 1009,
1010

Router(config)# ephone-hunt 2 peer
Router(config-ephone-hunt)# pilot 2222
Router(config-ephone-hunt)# list 2001, 2002, 2003, 2004, 2005, 2006
Router(config-ephone-hunt)# final 9000

Router(config)# ephone-hunt 3 peer
Router(config-ephone-hunt)# pilot 3333
Router(config-ephone-hunt)# list 3001, 3002, 3003, 3004

Router(config)# call application voice aa flash:app-b-acd-aa-x.x.x.x.tcl
Router(config)# call application voice aa dial-by-extension 1
Router(config)# call application voice aa number-of-hunt-grps 3
```

Related Commands

Command	Description
call application voice	Defines a name for a voice application and specifies the location of the Tcl or VoiceXML document to load for this application.
call application voice dial-by-extension-option	Enables direct extension access and sets the access number for Cisco CME B-ACD.
ephone-hunt	Enters ephone-hunt configuration mode for the purposes of creating and configuring a hunt group for use in a Cisco CME system.

call application voice queue-len

Effective with Cisco IOS Release 12.3(14)T and later, the **call application queue-len** command is replaced by the **param queue-len** command. See the **param queue-len** command for more information.

To set the maximum number of calls allowed for each ephone hunt group's call queue that is used by Cisco CallManager Express (Cisco CME) basic automatic call distribution (B-ACD), use the **call application voice queue-len** command in global configuration mode. To remove the queue-length setting, use the **no** form of this command.

call application voice *application-name* **queue-len** *number*

no call application voice *application-name* **queue-len** *number*

Command Default

application-name	Application name given to the call queue script in the **call application voice** command.
number	Number of calls that can be waiting in each ephone hunt group's queue. The range is dependent on your hardware configuration. The range is from 1 to 30. The default is 10.

Defaults

Thirty calls are allowed in each call queue.

Command Modes

Global configuration

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.3(11)XL	3.2.1	This command was introduced.
12.3(14)T	Cisco CME 3.2.2	This command was integrated into Cisco IOS Release 12.3(14)T.
12.3(14)T	3.3	This command was replaced by the **param queue-len** command.

Usage Guidelines

This command is used only with with a version of the Cisco CME B-ACD script that is earlier than 2.1.0.0 and is valid only for the configuration of Cisco CME B-ACD call queue scripts. The following sequence of events might occur:

- An outside call comes into a system configured with Cisco CME B-ACD.
- A menu option is selected that transfers the call to an ephone hunt-group pilot number.
- All of the ephone hunt group's ephone-dns are busy.

In that case, the call will be sent to a queue for that individual hunt group. The number of calls that each ephone hunt group can hold in its queue is configured by the **call application voice queue-len** command.

In the following configuration example, ephone hunt group 1 supports two ephone-dns; ephone hunt group 2 supports three ephone-dns; and the queue length is 10 for both ephone hunt groups:

```
ephone-hunt 1 peer
 pilot 1111
 list 1001, 1002

ephone-hunt 2 peer
 pilot 2222
 list 2001, 2002, 2003
call application voice queue flash:app-b-acd-x.x.x.x.tcl
call application voice callqueuescriptfilename queue-len 10
```

If ephone hunt group 1's ephone-dns are busy, ten more calls can be made to ephone hunt group 1. During that time, the calls in the queue would periodically retry the pilot numbers (**call application voice max-time-retry-timer** command) and receive secondary greetings (**call application voice second-greeting-time** command). If none of the calls has hung up or connected to an ephone-dn, the eleventh caller would hear the en_bacd_disconnect.au message and a busy signal. The default message is, "We are unable to take your call at this time. Please try again at a later time. Thank you for calling." Includes a four-second pause after the message.

For ephone hunt group 2, three calls can be connected to ephone-dns 2001, 2002, and 2003, and ten calls can be waiting in ephone hunt group 2's queue. If the status remains unchanged, the fourteenth caller hears the disconnect message and a busy signal. But if one of the earlier calls disconnects (either by leaving the queue or by ending a call), the fourteen call enters the queue.

The maximum number of calls allowed in the queues of ephone hunt groups must be based on the number of ports or trunks available. For example, if you had 20 foreign exchange office (FXO) ports and two ephone hunt groups, you could configure a maximum of ten calls per ephone hunt-group queue with the **call application voice queue-len 10** command. You could use the same configuration if you had a single T1 trunk, which supports 23 channels.

Examples

The following example configures a Cisco CME B-ACD call queue script to allow a maximum of 12 calls to wait in each ephone hunt group's calling queue for ephone-dns to become available:

```
Router(config)# call application voice queue flash:app-b-acd-x.x.x.x.tcl
Router(config)# call application voice queue queue-len 12
```

Related Commands

Command	Description
call application voice	Defines a name for a voice application and specifies the location of the Tcl or VoiceXML document to load for this application.
call application voice call-retry-timer	Assigns the length of time that calls to Cisco CME B-ACD must wait before attempting to transfer to an ephone hunt-group pilot number or to voice mail.
ephone-dn	Enters ephone-dn configuration mode for the purposes of creating and configuring an extension for a Cisco IP phone line.
ephone-hunt	Enters ephone-hunt configuration mode for the purposes of creating and configuring a hunt group for use in a Cisco CME system.

call application voice queue-manager-debugs

Effective with Cisco IOS Release 12.3(14)T and later, the **call application queue-manager-debugs** command is replaced by the **param queue-manager-debugs** command. See the **param aa-hunt** command for more information.

To enable or disable the collection of call queue debug information from Cisco CallManager Express (Cisco CME) basic automatic call distribution (B-ACD), use the **call application voice queue-manager-debugs** command in global configuration mode. To remove the current setting, use the **no** form of this command with the keyword that was used in the previous occurrence of the **call application voice queue-manager-debugs** command.

call application voice *application-name* **queue-manager-debugs** [**0** | **1**]

no call application voice *application-name* **queue-manager-debugs** [**0** | **1**]

Syntax Description

application-name	Application name given to the call queue script in the **call application voice** command.
0	Disables debugging.
1	Enables debugging.

Command Default

The collection of call queue debug information from Cisco CME B-ACD is disabled.

Command Modes

Global configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)XL	Cisco CME 3.2.1	This command was introduced.
12.3(14)T	Cisco CME 3.2.2	This command was integrated into Cisco IOS Release 12.3(14)T.
12.3(14)T	Cisco CME 3.3	This command was replaced by the **param queue-manager-debugs** command.

Usage Guidelines

This command is used only with with a version of the Cisco CME B-ACD script that is earlier than 2.1.0.0 and is valid only for the configuration of Cisco CME B-ACD call queue scripts. It enables the collection of data regarding call queue activity. It is used in conjunction with the **debug voip ivr script** command. Both commands must be enabled at the same time.

For any configuration changes to take effect, you must reload the Cisco CME B-ACD scripts.

Examples

The following example configures a Cisco CME B-ACD call queue script to enable debugging for the collection of data for the **debug voip ivr script** command:

```
Router(config)# call application voice queue flash:app-b-acd-x.x.x.x.tcl
Router(config)# call application voice queue queue-manager-debugs 1
```

Related Commands

Command	Description
call application voice	Defines a name for a voice application and specifies the location of the Tcl or VoiceXML document to load for this application.
debug voip ivr script	Display debugging messages for IVR scripts.

call application voice second-greeting-time

Effective with Cisco IOS Release 12.3(14)T and later, the **call application voice second-greeting-time** command is replaced by the **param second-greeting-time** command. See the **param second-greeting-time** command for more information.

To set the delay before the second greeting is played after a caller joins a Cisco CallManager Express (Cisco CME) basic automatic call distribution (B-ACD) calling queue and set the interval of time at which the second-greeting message is repeated, use the **call application voice second-greeting-time** command in global configuration mode. To remove the second-greeting time, use the **no** form of this command.

call application voice *application-name* **second-greeting-time** *seconds*

no call application voice *application-name* **second-greeting-time** *seconds*

Syntax Description

application-name	Application name given to the auto-attendant (AA) script in the **call application voice** command.
seconds	Amount of time that second-greeting message must wait before it can be played. The range is from 30 to 120 seconds. The default is 60 seconds.

Command Default

The second-greeting delay time is 60 seconds.

Command Modes

Global configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)XL	Cisco CME 3.2.1	This command was introduced.
12.3(14)T	Cisco CME 3.2.2	This command was integrated into Cisco IOS Release 12.3(14)T.
12.3(14)T	Cisco CME 3.3	This command was replaced by the **param second-greeting-time** command.

Usage Guidelines

This command is used only with with a version of the Cisco CME B-ACD script that is earlier than 2.1.0.0 and is valid only for the configuration of Cisco CME B-ACD AA scripts. A second greeting is an audio message of up to 15 seconds in length. The default announcement is, “All agents are currently busy assisting other customers. Continue to hold for assistance. Someone will be with you shortly.” The second-greeting message is only presented to callers waiting in a CME B-ACD call queue.

The second-greeting time is clocked when the second-greeting message begins, not after it ends. For example, if the second greeting were 15 seconds in length and the configured second-greeting time were 70 seconds, the greeting would begin every 70 seconds, not 85 seconds as if to allow for the 15-second message.

For any configuration changes to take effect, you must reload the Cisco CME B-ACD scripts.

Examples

The following example configures a Cisco CME B-ACD AA script to allow a second-greeting message to be repeated every 50 seconds as long as a call is in a call queue.

```
Router(config)# call application voice aa flash:app-b-acd-aa-x.x.x.x.tcl
Router(config)# call application voice AAscriptfilename second-greeting-time 50
```

Related Commands

Command	Description
call application voice	Defines a name for a voice application and specifies the location of the Tcl or VoiceXML document to load for this application.
ephone-dn	Enters ephone-dn configuration mode for the purposes of creating and configuring an extension for a Cisco IP phone line.
ephone-hunt	Enters ephone-hunt configuration mode for the purposes of creating and configuring a hunt group for use in a Cisco CME system.

call application voice voice-mail

Effective with Cisco IOS Release 12.3(14)T and later, the **call application voice voice-mail** command is replaced by the **param voice-mail** command. See the **param voice-mail** command for more information.

To assign a pilot number for the Cisco CallManager Express (Cisco CME) basic automatic call distribution (B-ACD) service's voice mail, use the **call application voice voice-mail** command in global configuration mode. To remove the voice-mail pilot number, use the **no** form of the command.

call application voice *application-name* **voice-mail** *number*

no call application voice *application-name* **voice-mail** *number*

Syntax Description

application-name	Application name given to the auto attendant (AA) script in the **call application voice** command.
number	Pilot number of the voice mail to which calls to Cisco CME B-ACD will be transferred.

Command Default

No voice-mail pilot number is configured for Cisco CME B-ACD.

Command Modes

Global configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)XL	Cisco CME 3.2.1	This command was introduced.
12.3(14)T	Cisco CME 3.2.2	This command was integrated into Cisco IOS Release 12.3(14)T.
12.3(14)T	Cisco CME 3.3	This command was replaced by the **param voice-mail** command.

Usage Guidelines

This command is used only with with a version of the Cisco CME B-ACD script that is earlier than 2.1.0.0 and is valid only for the configuration of Cisco CME B-ACD AA scripts. Only one pilot number is allowed per Cisco CME B-ACD service. Calls to the service will be sent to this voice mail number.

For any configuration changes to take effect, you must reload the Cisco CME B-ACD scripts.

Examples

The following example configures a Cisco CME B-ACD voice-mail pilot number as 5000.

```
Router(config)# call application voice aa flash:app-b-acd-aa-x.x.x.x.tcl
Router(config)# call application voice aa voice-mail 5000
```

Related Commands

Command	Description
call application voice	Defines a name for a voice application and specifies the location of the Tcl or VoiceXML document to load for this application.

call application voice welcome-prompt

Effective with Cisco IOS Release 12.3(14)T and later, the **call application voice welcome-prompt** command is replaced by the **param welcome-prompt** command. See the **param welcome-prompt** command for more information.

To assign an audio file that is used by the Cisco CallManager Express (Cisco CME) basic automatic call distribution (B-ACD) auto-attendant (AA) script for the welcome greeting, use the **call application welcome-prompt** command in global configuration mode. To remove the audio file assignment, use the **no** form of this command.

call application voice *application-name* **welcome-prompt** *_audio-filename*

no call application voice *application-name* **welcome-prompt** *_audio-filename*

Syntax Description

application-name	Application name given to the AA script in the **call application voice** command.
_audio-filename	Filename of the welcome greeting to be played when callers first reach the Cisco CME B-ACD, preceded by the underscore (_) character. The filename must not have a language code prefix, such as "en," for English.

Command Default

The welcome audio file downloaded with Cisco CME B-ACD is used for the welcome prompt.

Command Modes

Global configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)XL	Cisco CME 3.2.1	This command was introduced.
12.3(14)T	Cisco CME 3.2.2	This command was integrated into Cisco IOS Release 12.3(14)T.
12.3(14)T	Cisco CME 3.3	This command was replaced by the **param welcome-prompt** command.

Usage Guidelines

This command is used only with with a version of the Cisco CME B-ACD script that is earlier than 2.1.0.0 and is valid only for the configuration of Cisco CME B-ACD AA scripts. The welcome greeting is the initial AA response to a caller. The default audio file used is en_bacd_welcome.au, which is is downloaded with Cisco CME B-ACD and announces, "Thank you for calling," and includes a two-second pause after the message.

The filename must be preceded by an underscore (_) character. In addition, it must not contain a language-code prefix, such as "en" for English. For example, for en_bacd_welcome.au, you must configure **welcome-prompt _bacd_welcome.au** instead of **welcome-prompt _en_bacd_welcome.au**.

For any configuration changes to take effect, you must reload the Cisco CME B-ACD scripts.

Examples

The following example sets file name en_welcome.au as the welcome greeting for Cisco CME B-ACD:

```
Router(config)# call application voice aa flash:app-b-acd-aa-x.x.x.x.tcl
Router(config)# call application voice aa welcome-prompt _bacd_welcome_2.au
```

Related Commands

Command	Description
call application voice	Defines a name for a voice application and specifies the location of the Tcl or VoiceXML document to load for this application.
call application voice aa-name	Associates a Cisco CME B-ACD call queue script with a Cisco CME B-ACD AA script
call application voice service-name	Associates a Cisco CME B-ACD AA script with a Cisco CME B-ACD call queue script.

caller-id

To specify whether to pass the local caller ID or the original caller ID with calls from a Cisco CallManager Express extension (ephone-dn) that is using loopback, use the **caller-id command in** ephone-dn configuration mode. To return to the default, use the **no** form of this command.

caller-id {**local** | **passthrough**}

no caller-id {**local** | **passthrough**}

Syntax Description

local	Passes the local caller ID for redirected calls. For transferred calls, caller ID is provided by the original caller-ID information source (for example, from a separate loopback-dn that handles inbound calls or from a public switched telephone network interface. For forwarded calls, caller ID is provided by the original caller-ID information source or, for local IP phones, is extracted from the redirected information associated with the call.
passthrough	Passes the original caller ID for redirected calls. For transferred calls, the caller ID is provided by the original caller-ID information that is obtained from the inbound side of the loopback-dn. For forwarded calls, the caller ID is provided by the original caller-ID information of the incoming call.

Defaults

For transferred calls, caller ID is provided by the number and name fields from the outbound side of the loopback-dn. For forwarded calls, caller ID is provided by the original caller ID of the incoming call. Settings for the **caller-id block** command and translation rules on the outbound side are executed.

Command Modes

Ephone-dn configuration

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.2(15)ZJ3	3.0	This command was introduced.
12.3(4)T	3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Usage Guidelines

This command is valid only for ephone-dns that are being used for loopback.

Examples

The following example selects local caller ID for redirected calls:

```
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 5001
Router(config-ephone-dn)# loopback-dn 15 forward 4
Router(config-ephone-dn)# caller-id local
Router(config-ephone-dn)# no huntstop
```

Related Commands

Command	Description
ephone-dn	Enters ephone-dn configuration mode.
loopback-dn	Creates a virtual loopback voice port (loopback-dn) to establish a demarcation point for VoIP voice calls and supplementary services.

caller-id block (voice register template)

Note Effective with Cisco IOS Release 12.4(9)XJ, the **caller-id block (voice register template)** command is not available in Cisco IOS software.

To enable caller-ID blocking for outbound calls from a specific SIP phone, use the **caller-id block** command in voice register template configuration mode. To disable caller-ID blocking, use the **no** form of this command.

caller-id block

no caller-id block

Syntax Description

This command has no arguments or keywords.

Command Default

Caller ID blocking is disabled.

Command Modes

Voice register template configuration

Command History

Cisco IOS Release	Version	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.
12.4(11)XJ	Cisco Unified CME 4.1	This command was removed.
12.4(15)T	Cisco Unified CME 4.1	This command was removed in Cisco IOS Release 12.4(15)T.

Usage Guidelines

This command sets caller-ID blocking for outbound calls originating from any SIP phone that uses the specified template. This command requests the far-end gateway device to block the display of the calling party information for calls received from the specified SIP phone. This command does not affect the calling party information displayed for inbound calls received by the SIP phone. To apply a template to a SIP phone, use the **template** command in voice register pool configuration mode.

Examples

The following example shows how to enable caller-ID blocking in template 1:

```
Router(config)# voice register template 1
Router(config-register-temp)# caller-id block
```

Related Commands

Command	Description
anonymous block (voice register template)	Enables anonymous call blocking in a SIP phone template.
template (voice register pool)	Applies a template to a SIP phone.
voice register template	Enters voice register template configuration mode and defines a template of common parameters for SIP phones.

caller-id block code (telephony-service)

To set a code for a user to dial to block the display of caller ID on selected outgoing calls from Cisco IP phones, use the **caller-id block code** command in telephony-service configuration mode. To remove the code, use the **no** form of this command.

caller-id block code *code-string*

no caller-id block code

Syntax Description

code-string	Character string to dial to enable blocking of caller ID display on selected outgoing calls. The first character must be an asterisk (*) and the remaining characters must be digits. The string can contain a maximum of 16 characters.

Defaults

No caller-ID blocking code is defined.

Command Modes

Telephony-service configuration

Command History

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

Usage Guidelines

Once the caller-ID blocking code has been defined using this command, phone users should enter the caller-ID blocking code before dialing any call on which they want their caller ID not to display.

Examples

The following example sets a caller-ID blocking code of *4321:

```
Router(config)# telephony-service
Router(config-telephony)# caller-id block code *4321
```

Related Commands

Command	Description
telephony-service	Enters telephony-service configuration mode.

call-feature-uri

To specify the uniform resource identifier (URI) for soft keys on SIP phones registered to a Cisco Unified CME router, use the **call-feature-uri** command in voice register global configuration mode. To remove a URI association, use the **no** form of this command.

call-feature-uri cfwdall *service-uri*

no call-feature-uri cfwdall

Syntax Description

cfwdall *service-uri*	URI that is requested when the call forward all (CfwdAll) soft key is pressed.

Command Default

No URI is associated with the soft key.

Command Modes

Voice register global configuration

Command History

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

Usage Guidelines

This command updates the service URI for call forward all in the configuration file downloaded from the Cisco Unified CME router to the SIP phones during phone registration. The configuration is updated when Call Forward All is enabled from the phone using the CfwdAll soft key.

After you configure this command, restart the phone by using the **reset** or **restart** command.

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

Examples

The following example shows how to specify the URI for the call forward all soft key:

```
Router(config)# voice register global
Router(config-register-global)# call-feature-uri cfwdall http://1.4.212.11/cfwdall
```

Related Commands

Command	Description
call-forward b2bua all	Enables call forwarding for a SIP back-to-back user agent (B2BUA) so that all incoming calls are forwarded to another extension.
reset (voice register pool)	Performs a complete reboot of one phone associated with a Cisco Unified CME router.
reset (voice register global)	Performs a complete reboot of all SIP phones associated with a Cisco Unified CME router.

call-forward

To globally apply dialplan-pattern expansion to redirecting numbers for extension numbers associated with SCCP IP phones in Cisco Unified CME, use the **call-forward system** command in telephony-service configuration mode. To disable the **call-forward system** command, use the **no** form of this command.

call-forward system redirecting-expanded

no call-forward system redirecting-expanded

Syntax Description

system	Call forward system parameter.
redirecting-expanded	Expand redirecting extensions to an E.164 number.

Command Default

The redirecting number is not expanded.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.

Usage Guidelines

Use this command to apply dialplan-pattern expansion on a per-system basis to individual nonSIP redirecting numbers, including original called and last reroute numbers, in a Cisco Unified CME system.

When A calls B, and B forwards the call to C; B is the original called number and the last reroute number. If C then forwards or transfers the call to another number, C becomes the original called number and the last reroute number. The dial-plan pattern expansion is applied to both redirecting numbers. Once the number is expanded, it remains expanded during the entire call instance.

The dial-plan pattern to be matched must be configured using the **dialplan-pattern** command.

Examples

The following example shows how to create a dialplan-pattern for expanding calling numbers to an E.164 number and to also apply the expansion globally to redirecting numbers.

```
Router(config)# voice register global
Router(config-register-global)# dialplan-pattern 1 5105550... extension-length 5
Router(config-register-global)# call-forward system redirecting-expanded
```

Related Commands

Command	Description
dialplan-pattern	Create global prefix for expanding extension numbers of forward-to and transfer-to targets.
show telephony-service dial-peer	Displays dial peer information for extensions in a Cisco Unified CME system.

call-forward (voice register)

To globally apply dialplan-pattern expansion to redirecting numbers for extension numbers associated with SIP IP phones in Cisco Unified CME, use the **call-forward system** command in voice register global configuration mode. To disable the **call-forward system** command, use the **no** form of this command.

call-forward system redirecting-expanded

no call-forward system redirecting-expanded

Syntax Description

system	Call forward system parameter.
redirecting-expanded	Redirecting extension is to be expanded to an E.164 number.

Command Default

The redirecting number is not expanded.

Command Modes

Voice register global configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.

Usage Guidelines

Use this command to apply dialplan-pattern expansion on a per-system basis to individual SIP redirecting numbers, including original called and last reroute numbers, in Cisco Unified CME.

When A calls B, and B forwards the call to C; B is the original called number and the last reroute number. If C then forwards or transfers the call to another number, C becomes the original called number and the last reroute number. The dial-plan pattern expansion is applied to both redirecting numbers. Once the number is expanded, it remains expanded during the entire call instance.

This command supports call forward using B2BUA only.

The dial-plan pattern to be matched must be configured using the **dialplan-pattern** command.

Examples

The following example shows how to create a dialplan-pattern for expanding calling numbers of SIP phones to an E.164 number and to also apply the expansion globally to SIP redirecting numbers.

```
Router(config)# voice register global
Router(config-register-global)# dialplan-pattern 1 5105550... extension-length 5
Router(config-register-global)# call-forward system redirecting-expanded
```

Related Commands

Command	Description
dialplan-pattern (voice register)	Create global prefix for expanding extension numbers of forward-to and transfer-to targets if the target is an extension on a SIP phone.
show voice register dial-peer	Displays dial peer information for extensions in a Cisco Unified CME system.

call-forward all

To configure call forwarding so that all incoming calls to a directory number are forwarded to another directory number, use the **call-forward all** command in ephone-dn or ephone-dn-template configuration mode. To disable call forwarding, use the **no** form of this command.

call-forward all *directory-number*

no call-forward all

Syntax Description

directory-number	Directory number to which calls are forwarded. Represents a fully qualified E.164 number.

Command Default

Call forwarding for all calls is not set.

Command Modes

Ephone-dn configuration
Ephone-dn-template configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco ITS 1.0	This command was introduced on the following platforms: Cisco 2600 series and Cisco 3600 series, and Cisco IAD2420 series.
12.2(2)XT	Cisco ITS 2.0	This command was implemented on the Cisco 1750 and Cisco 1751.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	Cisco ITS 2.0	This command was implemented on the Cisco 2600-XM and Cisco 2691.
12.2(11)T	Cisco ITS 2.01	This command was implemented on the Cisco 1760.
12.4(4)XC	Cisco Unified CME 4.0	This command was made available in ephone-dn-template configuration mode.
12.4(9)T	Cisco Unified CME 4.0	This command in ephone-dn-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

The call forwarding mechanism applies to the individual directory number and cannot be configured for individual Cisco Unified IP phones.

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

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

Examples

The following example shows how to set call forwarding of all calls on directory number 5001 to directory number 5005. All incoming calls destined for extension 5001 are forwarded to another Cisco IP phone with the extension number 5005:

```
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 5001
Router(config-ephone-dn)# call-forward all 5005
```

The following example uses an ephone-dn template to forward all calls for extension 5001 to extension 5005.

```
Router(config)# ephone-dn-template 3
Router(config-ephone-dn-template)# call-forward all 5005
Router(config-ephone-dn-template)# exit
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 5001
Router(config-ephone-dn)# ephone-dn-template 3
```

Related Commands

Command	Description
call-forward busy	Configures call forwarding to another number when a Cisco Unified IP phone is busy.
call-forward noan	Configures call forwarding to another number when no answer is received from a Cisco Unified IP phone.
ephone	Enters ephone configuration mode.
ephone-dn	Enters ephone-dn configuration mode.

call-forward b2bua all

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

call-forward b2bua all *directory-number*

no call-forward b2bua all

Syntax Description

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

Command Default

Disabled.

Command Modes

Voice register dn configuration (Cisco Unified SIP SRST only)
Voice register pool configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4 Cisco SIP SRST 3.4	This command was introduced.
12.4(11)XJ	Cisco Unified CME 4.1	This command was removed from voice register pool configuration mode for Cisco Unified CME only.
12.4(15)T	Cisco Unified CME 4.1	This command with modifications was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

This command in voice register dn configuration mode applies the call forward mechanism to a individual SIP extension in Cisco Unified CME or Cisco Unified SIP SRST. This command in voice register pool configuration mode is for Cisco Unified SIP SRST only and applies to SIP IP phones on which the extension appears.

If this command is configured in both the voice register dn and voice register pool configuration modes, the configuration under voice register dn takes precedence.

We recommend that you do not use this command to configure a SIP extension or SIP IP phone that is a member of a hunt group. If this command is configured for a member of a hunt group, remove the phone from any hunt group to which it is assigned to avoid forwarding calls to all phones in the hunt group.

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

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

Examples

Cisco Unified CME and Cisco Unified SIP SRST

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

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

Cisco Unified SIP SRST

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

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

Related Commands

Command	Description
call-forward b2bua busy	Enables call forwarding for a SIP B2BUA so that incoming calls to a busy extension are forwarded to another extension.
call-forward b2bua mailbox	Controls the specific voice-mail box selected in a voice-mail system at the end of a call forwarding exchange.
call-forward b2bua noan	Enables call forwarding for a SIP B2BUA so that incoming calls to an extension that does not answer after a configured amount of time are forwarded to another extension.
call-waiting (voice register pool)	Enables call waiting on a SIP phone.

call-forward b2bua busy

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

call-forward b2bua busy *directory-number*

no call-forward b2bua busy

Syntax Description

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

Command Default

Disabled.

Command Modes

Voice register dn configuration (Cisco Unified SIP SRST only)
Voice register pool configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4 Cisco SIP SRST 3.4	This command was introduced.
12.4(11)XJ	Cisco Unified CME 4.1	This command was removed from voice register pool configuration mode for Cisco Unified CME only.
12.4(15)T	Cisco Unified CME 4.1	This command with modifications was removed in Cisco IOS Release 12.4(15)T.

Usage Guidelines

This command in voice register dn configuration mode applies the call forward mechanism to a individual SIP extension in Cisco Unified CME or Cisco Unified SIP SRST that is off-hook. This command in voice register pool configuration mode is for Cisco Unified SIP SRST only and applies to SIP IP phones on which the extension appears.

In Cisco Unified CME, call forward busy is also invoked when a call arrives for a destination that is configured but unregistered. A destination is considered to be configured if its number is listed under the voice register dn configuration.

If this command is configured in both voice register dn and voice register pool configuration modes, the configuration under voice register dn takes precedence.

We recommend that you do not use this command to configure a SIP extension or SIP IP phone that is a member of a hunt group. If this command is configured for a member of a hunt group, remove the phone from any hunt group to which it is assigned to avoid forwarding calls to all phones in the hunt group.

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

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

Examples

Cisco Unified CME and Cisco Unified SIP SRST

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

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

Cisco Unified SIP SRST

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

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

Related Commands

Command	Description
call-forward b2bua all	Enables call forwarding for a SIP B2BUA so that all incoming calls are forwarded to another extension.
call-forward b2bua mailbox	Controls the specific voice-mail box selected in a voice-mail system at the end of a call forwarding exchange.
call-forward b2bua noan	Enables call forwarding for a SIP B2BUA so that incoming calls to an extension that does not answer after a configured amount of time are forwarded to another extension.
call-waiting (voice register pool)	Enables call waiting on a SIP phone.

call-forward b2bua mailbox

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

call-forward b2bua mailbox *directory-number*

no call-forward b2bua mailbox

Syntax Description

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

Command Default

Disabled.

Command Modes

Voice register dn configuration (Cisco Unified SIP SRST only)
Voice register pool configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4 Cisco SIP SRST 3.4	This command was introduced.
12.4(11)XJ	Cisco Unified CME 4.1	This command was removed from voice register pool configuration mode for Cisco Unified CME only.
12.4(15)T	Cisco Unified CME 4.1	This command with modifications was integrated into Cisco IOS Release 12.4(15)T

Usage Guidelines

This command is used to denote the voice-mail box to use at the end of a chain of call forwards to busy or no answer destinations. It can be used to forward calls to a voice-mail box that has a different number than the forwarding extension, such as a shared voice-mail box.

This command in voice register dn configuration mode applies the call forward mechanism to a individual SIP extension in Cisco Unified CME or Cisco Unified SIP SRST. This command in voice register pool configuration mode is for Cisco Unified SIP SRST only and applies to SIP IP phones on which the extension appears.

If this command is configured in both the voice register dn and voice register pool configuration modes, the configuration under voice register dn takes precedence.

We recommend that you do not use this command to configure a SIP extension or SIP IP phone that is a member of a hunt group. If this command is configured for a member of a hunt group, remove the phone from any hunt group to which it is assigned to avoid forwarding calls to all phones in the hunt group.

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

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

Examples

Cisco Unified CME and Cisco Unified SIP SRST

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

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

Cisco Unified SIP SRST

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

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

Related Commands

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

call-forward b2bua noan

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

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

no call-forward b2bua noan

Syntax Description

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

Command Default

Disabled.

Command Modes

Voice register dn configuration (Cisco Unified SIP SRST only)
Voice register pool configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4 Cisco SIP SRST 3.4	This command was introduced.
12.4(11)XJ	Cisco Unified CME 4.1	This command was removed from voice register pool configuration mode for Cisco Unified CME only.
12.4(15)T	Cisco Unified CME 4.1	This command with modifications was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

This command in voice register dn configuration mode applies the call forward mechanism to a individual SIP extension in Cisco Unified CME or Cisco Unified SIP SRST that remains unanswered after a specified length of time. This command in voice register pool configuration mode is for Cisco Unified SIP SRST only and applies to SIP IP phones on which the extension appears.

If this command is configured in both the voice register dn and voice register pool configuration modes, the configuration under voice register dn takes precedence.

We recommend that you do not use this command to configure a SIP extension or SIP IP phone that is a member of a hunt group. If this command is configured for a member of a hunt group, remove the phone from any hunt group to which it is assigned to avoid forwarding calls to all phones in the hunt group.

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

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

Examples

Cisco Unified CME and Cisco Unified SIP SRST

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

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

Cisco Unified SIP SRST

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

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

Related Commands

Command	Description
call-forward b2bua all	Enables call forwarding for a SIP B2BUA so that all incoming calls are forwarded to another extension.
call-forward b2bua busy	Enables call forwarding for a SIP B2BUA so that incoming calls to a busy extension are forwarded to another extension.
call-forward b2bua mailbox	Controls the specific voice-mail box selected in a voice-mail system at the end of a call forwarding exchange.
call-waiting (voice register pool)	Enables call waiting on a SIP phone.

call-forward b2bua unreachable

Note Effective with Cisco IOS Release 12.4(11)XJ, the **call-forward b2bua unreachable** command is not available in Cisco IOS software.

To forward calls to a phone that is not registered to Cisco Unified CME, use the **call-forward b2bua unreachable** command in voice register dn or voice register pool configuration mode. To disable call forwarding, use the **no** form of this command.

call-forward b2bua unreachable *directory-number*

no call-forward b2bua unreachable

Syntax Description

directory-number	Telephone number to which calls are forwarded. Represents a fully qualified E.164 number.

Defaults

Disabled

Command Modes

Voice register dn configuration
Voice register pool configuration

Command History

Cisco IOS Release	Version	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.
12.4(11)XJ	Cisco Unified CME 4.1	This command was removed.
12.4(15)T	Cisco Unified CME 4.1	This command was removed in Cisco IOS Release 12.4(15)T.

Usage Guidelines

Call forward unreachable is triggered when a call arrives for a destination that is configured but unregistered with Cisco CME. A destination is considered to be configured if its number is listed under the voice register pool or voice register dn configurations.

If call forward unreachable is not configured for a pool or directory number (DN) register, any calls that match the numbers in that pool or DN register will use call forward busy instead.

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

Examples

The following example shows how to forward calls to extension 5005 when extension 5001 on directory number 4 is unreachable, either because it is unplugged or the network between the Cisco router and the extension is nonfunctional. The timeout before the call is forwarded to extension 5005 is 10 seconds.

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

```
Router(config-register-dn)# number 5001
Router(config-register-dn)# call-forward b2bua unreachable 5005 timeout 10
```

Related Commands

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

call-forward busy

To configure call forwarding so that incoming calls to a busy extension (ephone-dn) are forwarded to another extension, use the **call-forward busy** command in ephone-dn or ephone-dn-template configuration mode. To disable call forwarding, use the **no** form of this command.

call-forward busy *target-number* [**primary** | **secondary**] [**dialplan-pattern**]

no call-forward busy

Syntax Description

target-number	Phone number to which calls are forwarded.
primary	(Optional) Call forwarding is selectively applied only to the dial peer created for the primary number for this ephone-dn.
secondary	(Optional) Call forwarding is selectively applied only to the dial peer created for the secondary number for this ephone-dn.
dialplan-pattern	(Optional) Call forwarding is selectively applied only to dial peers created for this ephone-dn by the dial-plan pattern.

Command Default

Call forwarding for a busy extension is not enabled.

Command Modes

Ephone-dn configuration
Ephone-dn-template configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco ITS 1.0	This command was introduced on the following platforms: Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(2)XT	Cisco ITS 2.0	This command was implemented on the Cisco 1750 and Cisco 1751.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	Cisco ITS 2.0	This command was implemented on the Cisco 2600-XM and Cisco 2691.
12.2(11)T	Cisco ITS 2.01	This command was implemented on the Cisco 1760.
12.4(4)XC	Cisco Unified CME 4.0	The **primary**, **secondary**, and **dialplan-pattern** keywords were added, and this command was made available in ephone-dn-template configuration mode.
12.4(9)T	Cisco Unified CME 4.0	This command with the **primary**, **secondary**, and **dialplan-pattern** keywords added, and this command in ephone-dn-template configuration mode was integrated into Cisco IOS 12.4(9)T.

Usage Guidelines

The call forwarding mechanism is applied to an individual extension (ephone-dn) and is not applied to the phone on which the extension appears.

Normally, call forwarding is applied to all dial peers that are created by the ephone-dn. An ephone-dn can create up to four dial peers:

- A dial peer for the primary number
- A dial peer for the secondary number
- A dial peer for the primary number as expanded by the **dialplan-pattern** command
- A dial peer for the secondary number as expanded by the **dialplan-pattern** command

The **primary**, **secondary**, and **dialplan-pattern** keywords allow you to apply call forwarding selectively to one or more dial peers based on the exact called number that was used to route the call to the ephone-dn. If none of the optional keywords is used, call forwarding applies to all dial-peers that are defined for the ephone-dn.

An ephone-dn can have all four types of call forwarding defined at the same time: all-calls, no-answer, busy, and night-service. Each type of call forwarding can have a different forwarding target defined in its *target-number* argument. If more than one type of call forwarding is in effect (is active) at one time, the precedence order for evaluating the different types is as follows:

1. call forward night service
2. call forward all
3. call forward busy and call forward no answer

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

Examples

The following example forwards all calls for the ephone-dn 2345 when it is busy.

```
Router(config)# ephone-dn 236
Router(config-ephone-dn)# number 2345
Router(config-ephone-dn)# call-forward busy 2000
```

The following example uses an ephone-dn template to forward calls for extension 2345 when it is busy.

```
Router(config)# ephone-dn-template 6
Router(config-ephone-dn-template)# call-forward busy 2000
Router(config-ephone-dn-template)# exit
Router(config)# ephone-dn 236
Router(config-ephone-dn)# number 2345
Router(config-ephone-dn)# ephone-dn-template 6
```

The following example creates a dial-plan pattern to expand extension numbers into E.164 numbers. It then sets call forwarding of incoming calls to directory number 5005. In this example, call forwarding on busy is applied only when callers dial the primary number for this ephone-dn, 5001.

```
Router(config)# telephony-service
Router(config-telephony)# dialplan-pattern 1 40855501.. extension-length 4
extension-pattern 50..
Router(config-telephony)# exit
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 5001 secondary 5002
Router(config-ephone-dn)# call-forward busy 5005 primary
```

Related Commands

Command	Description
call-forward all	Configures call forwarding for all incoming calls to an ephone-dn.
call-forward night-service	Configures call forwarding for all incoming calls to an ephone-dn during the hours defined for night service.
call-forward noan	Configures call forwarding to another number when no answer is received from an ephone-dn.
ephone-dn	Enters ephone-dn configuration mode.
ephone-dn-template	Enters ephone-dn-template configuration mode.

call-forward max-length

To restrict the number of digits that can be entered using the CfwdALL soft key on an IP phone, use the **call-forward max-length command in** ephone-dn or ephone-dn-template configuration mode. To remove a restriction on the number of digits that can be entered, use the **no** form of this command.

call-forward max-length *length*

no call-forward max-length

Syntax Description

length	Number of digits that can be entered using the CfwdAll soft key on an IP phone.

Command Default

There is no restriction on the number of digits that can be entered.

Command Modes

Ephone-dn configuration
Ephone-dn-template configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(7)T	Cisco CME 3.1	This command was introduced.
12.4(4)XC	Cisco Unified CME 4.0	This command was made available in ephone-dn-template configuration mode.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.3(9)T.

Usage Guidelines

This command can be used to prevent a phone user from using the CfwdALL soft key on an IP phone to forward calls to numbers that will incur toll charges when they receive forwarded calls.

If the *length* argument is set to 0, the CfwdALL soft key is completely disabled. If the ephone-dn associated with the first line button has an active call forward number when this command is used to set the *length* argument to 0, the CfwdALL soft key will be disabled after the next phone restart.

The restriction created by this command does not apply to destinations that are entered using the Cisco IOS command-line interface (CLI) or the Cisco Unified CME GUI.

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

Examples

The following example restricts the number of digits that a phone user can enter using the CfwdALL soft key to four. In this example, extensions in the phone user's Cisco Unified CME system have four digits, so that means the user can use the IP phone to forward all calls to any extension in the system, but not to any number outside the system.

```
Router(config)# ephone-dn 1
```

```
Router(config-ephone-dn)# number 5001
Router(config-ephone-dn)# call-forward max-length 4
```

The following example uses an ephone-dn-template to restrict the number of digits that a phone user can enter using the CfwdALL soft key to four.

```
Router(config)# ephone-dn-template 4
Router(config-ephone-dn-template)# call-forward max-length 4
Router(config-ephone-dn-template)# exit
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 5001
Router(config-ephone-dn)# ephone-dn-template 4
```

Related Commands

Command	Description
call-forward all	Configures call forwarding for all incoming calls on one of the lines of a Cisco Unified IP phone.
ephone-dn	Enters ephone-dn configuration mode.
ephone-dn-template	Enters ephone-dn-template configuration mode.
ephone-dn-template (ephone-dn)	Applies an ephone-dn template to an ephone-dn.

call-forward night-service

To automatically forward calls to another number during night-service hours, use the **call-forward night-service** command in ephone-dn or ephone-dn-template configuration mode. To disable automatic call forwarding during night service, use the **no** form of this command.

call-forward night-service *target-number*

no call-forward night-service

Syntax Description

target-number	Phone number to which calls are forwarded.

Command Default

Calls are not forwarded during night-service hours.

Command Modes

Ephone-dn configuration
Ephone-dn-template

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.

Usage Guidelines

You must also configure the **night-service bell** command for this ephone-dn.

Night-service hours are defined using the **night-service date** and **night-service day** commands.

An ephone-dn can have all four types of call forwarding defined at the same time: all-calls, no-answer, busy, and night-service. Each type of call forwarding can have a different forwarding destination defined in its *target-number* argument. If more than one type of call forwarding is in effect (is active) at one time, the precedence order for evaluating the different types is as follows:

1. call forward night-service
2. call forward all
3. call forward busy and call forward no answer

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

Examples

The following example establishes night-service hours from 1 p.m. Saturday until 8 a.m. Monday. During that time, calls to extension 1000 (ephone-dn 1) are forwarded to extension 2346. Note that the **night-service bell** command has also been used for ephone-dn 1.

```
telephony-service
 night-service day sat 13:00 12:00
 night-service day sun 12:00 08:00
 night-service code *1234
!
```

```
ephone-dn 1
 number 1000
 night-service bell
 call-forward night-service 2346
!
ephone-dn 2
 number 2346

ephone 12
 button 1:1

ephone 13
 button 1:2
```

The following example uses an ephone-dn template to apply call forwarding for extension 2876 during the night service hours established in the previous example.

```
ephone-dn-template 2
 call-forward night-service 2346

ephone-dn 25
 number 2876
 ephone-dn-template 2
```

Related Commands

Command	Description
call-forward all	Configures call forwarding for all incoming calls to an ephone-dn.
call-forward busy	Configures call forwarding to another number when an ephone-dn is busy.
call-forward noan	Configures call forwarding to another number when no answer is received from an ephone-dn.
night-service bell (ephone-dn)	Marks an ephone-dn for night-service treatment.
night-service date	Defines a recurring time period associated with a month and day during which night service is active.
night-service day	Defines a recurring time period associated with a day of the week during which night service is active.

call-forward noan

To configure call forwarding so that incoming calls to an extension (ephone-dn) that does not answer are forwarded to another number, use the **call-forward noan command in** ephone-dn or ephone-dn-template configuration mode. To disable call forwarding, use the **no** form of this command.

call-forward noan *target-number* **timeout** *seconds* [**primary** | **secondary**] [**dialplan-pattern**]

no call-forward noan

Syntax Description

target-number	Phone number to which calls are forwarded.
timeout *seconds*	Sets the duration that a call can ring with no answer before the call is forwarded to the target number. Range is from 3 to 60000. There is no default value.
primary	(Optional) Call forwarding is selectively applied only to the dial peer created for the primary number for this ephone-dn.
secondary	(Optional) Call forwarding is selectively applied only to the dial peer created for the secondary number for this ephone-dn.
dialplan-pattern	(Optional) Call forwarding is selectively applied only to dial peers created for this ephone-dn by the dial-plan pattern.

Command Default

Call forwarding for an extension that does not answer is not enabled.

Command Modes

Ephone-dn configuration
Ephone-dn-template configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco ITS 1.0	This command was introduced on the following platforms: Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(2)XT	Cisco ITS 2.0	This command was implemented on the Cisco 1750 and Cisco 1751.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	Cisco ITS 2.0	This command was implemented on the Cisco 2600-XM and Cisco 2691.
12.2(11)T	Cisco ITS 2.01	This command was implemented on the Cisco 1760.
12.4(4)XC	Cisco Unified CME 4.0	The **primary**, **secondary**, and **dialplan-pattern** keywords were added, and this command was made available in ephone-dn-template configuration mode.

Usage Guidelines

The call forwarding mechanism is applied to an individual extension (ephone-dn) and is not applied to the phone on which the extension appears.

Normally, call forwarding is applied to all dial peers that are created by the ephone-dn. An ephone-dn can create up to four dial peers:

- A dial peer for the primary number
- A dial peer for the secondary number
- A dial peer for the primary number as expanded by the **dialplan-pattern** command
- A dial peer for the secondary number as expanded by the **dialplan-pattern** command

The **primary**, **secondary**, and **dialplan-pattern** keywords allow you to apply call forwarding selectively to one or more dial peers based on the exact called number that was used to route the call to the ephone-dn. If none of the optional keywords is used, call forwarding applies to all dial-peers that are defined for the ephone-dn.

An ephone-dn can have all four types of call forwarding defined at the same time: all-calls, no-answer, busy, and night-service. Each type of call forwarding can have a different forwarding target defined in its *target-number* argument. If more than one type of call forwarding is in effect (is active) at one time, the precedence order for evaluating the different types is as follows:

1. call forward night service
2. call forward all
3. call forward busy and call forward no answer

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

Examples

The following example forwards calls for the ephone-dn 2345 when it does not answer.

```
Router(config)# ephone-dn 236
Router(config-ephone-dn)# number 2345
Router(config-ephone-dn)# call-forward busy 2000
```

The following example uses an ephone-dn-template to forward calls for the ephone-dn 2345 when it does not answer.

```
Router(config)# ephone-dn-template 8
Router(config-ephone-dn-template)# call-forward busy 2000
Router(config-ephone-dn-template)# exit
Router(config)# ephone-dn 236
Router(config-ephone-dn)# number 2345
Router(config-ephone-dn)# ephone-dn-template 8
```

The following example creates a dial-plan pattern to expand extension numbers into E.164 numbers. It then sets call forwarding of incoming calls to directory number 5005. In this example, call forwarding on no answer is applied only when callers dial the primary number for this ephone-dn, 5001.

```
Router(config)# telephony-service
Router(config-telephony)# dialplan-pattern 1 40855501.. extension-length 4
extension-pattern 50..
Router(config-telephony)# exit
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 5001 secondary 5002
Router(config-ephone-dn)# call-forward noan 5005 primary
```

Related Commands

Command	Description
call-forward all	Configures call forwarding for all incoming calls for an ephone-dn.
call-forward busy	Configures call forwarding to another number when an ephone-dn is busy.
call-forward night-service	Configures call forwarding for all incoming calls to an ephone-dn during the hours defined for night service.
ephone-dn	Enters ephone-dn configuration mode.
ephone-dn-template	Enters ephone-dn-template configuration mode.
ephone-dn-template (ephone-dn)	Applies an ephone-dn-template to an ephone-dn.

call-forward pattern

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

call-forward pattern *pattern*

no call-forward pattern *pattern*

Syntax Description

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

Command Default

No call-forward pattern is defined.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.2(11)YT	2.1	This command was introduced.
12.2(15)T	2.1	This command was integrated into Cisco IOS Release 12.2(15)T.

Usage Guidelines

Use this command with Cisco IOS Telephony Services (ITS) V2.1, Cisco CallManager Express 3.0, or a later version.

When H.450.3 call forwarding is selected, the router must be configured with a Tool Command Language (Tcl) script that supports the H.450.3 protocol. The Tcl script is loaded on the router by using the **call application voice** command.

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

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

Examples

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

```
Router(config)# telephony-service
Router(config-telephony)# call-forward pattern 4...
```

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

```
Router(config)# telephony-service
Router(config-telephony)# call-forward pattern .T
```

Related Commands

Command	Description
call application voice	Defines an application, indicates the location of the corresponding Tcl files that implement the application, and loads the selected Tcl script.
telephony-service	Enters telephony-service configuration mode.

calling-number local

To replace a calling-party number and name with the forwarding-party number and name (the local number and name) in calls forwarded using local hairpin call routing, use the **calling-number local** command in telephony-service configuration mode. To reset to the default, use the **no** form of this command.

calling-number local [**secondary**]

no calling-number local

Syntax Description

secondary	(Optional) Uses the secondary number associated with the forwarding party instead of the primary number. The primary number is the default if this keyword is not used.

Defaults

Calling-party numbers and names are used in forwarded calls.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ3	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.3(15)ZJ4	Cisco CME 3.0	The **secondary** keyword was introduced.
12.3(14)T	Cisco CME 3.3	Support was added to the default IOS voice application framework and dependency on the TCL script was removed.

Usage Guidelines

In Cisco CME 3.2 and earlier versions, this command is used with the Tool Command Language (Tcl) script app-h450-transfer.2.0.0.7 or a later version.

In Cisco CME 3.3 and later versions, this command can be used without the TCL script because the functionality is integrated into the default IOS voice application framework.

If the ephone-dn used by a forwarding party has a secondary number in addition to its primary number and neither number is registered with the gatekeeper, the primary number is the number that appears as the calling number on hairpin-forwarded calls when the **calling-number local** command is used. If only one of the numbers is registered with the gatekeeper, the registered number is the number that appears as the calling number. If both numbers are registered with the gatekeeper, the primary number is the number that appears as the calling number.

If the ephone-dn used by a forwarding party has a secondary number in addition to its primary number and the **calling-number local secondary** command is used, the secondary number is the number that appears as the calling number on hairpin-forwarded calls if both numbers are registered with the gatekeeper or if both numbers are not registered. If only one number is configured to register with the gatekeeper, the number that is registered appears as the calling number.

Examples

The following example specifies use of the name and number of the local forwarding party in hairpin-forwarded calls:

```
Router(config)# telephony-service
Router(config-telephony)# calling-number local
```

The following examples demonstrate the use of the **calling-number local** command without the **secondary** keyword.

- The calling number for hairpin calls forwarded from ephone-dn 1 is 1234 in the following example:

```
calling-number local

ephone-dn 1
 number 1234 secondary 4321 no-reg
```

- The calling number for hairpin calls forwarded from ephone-dn 1 is 4321 in the following example:

```
calling-number local

ephone-dn 1
 number 1234 secondary 4321 no-reg primary
```

- The calling number for hairpin calls forwarded from ephone-dn 1 is 1234 in the following example:

```
calling-number local

ephone-dn 1
 number 1234 secondary 4321 no-reg both
```

or

```
 number 1234 secondary 4321
```

The following examples demonstrate the use of the **calling-number local secondary** command.

- The calling number for hairpin calls forwarded from ephone-dn 1 is 1234 in the following example:

```
calling-number local secondary

ephone-dn 1
 number 1234 secondary 4321 no-reg
```

- The calling number for hairpin calls forwarded from ephone-dn 1 is 4321 in the following example:

```
calling-number local secondary

ephone-dn 1
 number 1234 secondary 4321 no-reg primary
```

- The calling number for hairpin calls forwarded from ephone-dn 1 is 4321 in the following example:

```
calling-number local secondary

ephone-dn 1
 number 1234 secondary 4321 no-reg both
```

or

```
number 1234 secondary 4321
```

call-park system

To specify system parameters for the call-park feature, use the **call-park system redirect** command in telephony-service configuration mode. To return to the default, use the **no** form of this command.

call-park system {**redirect**}

no call-park system

Syntax Description

redirect	H.323 and SIP calls will use H.450 or the SIP Refer method of call forwarding or transfer to park calls and to pick up calls from park

Command Default

H.323 and SIP calls use hairpin call forwarding or transfer to park calls and to pick up calls from park.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.

Examples

The following example specifies that H.323 and SIP calls should use the H.450 or SIP Refer method of call forwarding or transfer to park calls and pick up calls from park:

```
Router(config)# telephony-service
Router(config-telephony)# call-park system redirect
```

call-waiting (voice register pool)

To enable call-waiting option on a SIP phone, use the **call-waiting** command in voice register pool configuration mode. To disable call waiting, use the **no** form of this command.

call-waiting

no call-waiting

Syntax Description

This command has no arguments or keywords.

Defaults

Enabled

Command Modes

Voice register pool configuration mode

Command History

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

Usage Guidelines

The call waiting feature is enabled by default on SIP phones. To disable call waiting, use the **no call-waiting** command.

Examples

The following example shows how to disable call waiting on SIP phone 1:

```
Router(config)# voice register pool 1
Router(config-register-pool)# no call-waiting
```

Related Commands

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

call-waiting beep

To allow call-waiting beeps to be accepted by or generated from an ephone-dn, use the **call-waiting beep** command in ephone-dn or ephone-dn-template configuration mode. To disable the acceptance and generation of call-waiting beeps by an ephone-dn, use the **no** form of this command.

call-waiting beep [**accept** | **generate**]

no call-waiting beep [**accept** | **generate**]

Syntax Description

accept	(Optional) Allows call-waiting beeps to be accepted by an ephone-dn.
generate	(Optional) Allows call-waiting beeps to be generated by an ephone-dn.

Command Default

Call-waiting beeps are accepted and generated.

Command Modes

Ephone-dn configuration
Ephone-dn-template configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)T	Cisco CME 3.2	This command was introduced.
12.4(4)XC	Cisco Unified CME 4.0	This command was made available in ephone-dn-template configuration mode.
12.4(9)T	Cisco Unified CME 4.0	This command in ephone-dn-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

The **call-waiting beep** command must be used with the **ephone-dn** command. The **call-waiting beep** command is used like a toggle and can be switched on and off for each ephone-dn.

A beep can be heard only if both sending and receiving ephone-dns are configured to accept call-waiting beeps.

To display how call-waiting beeps are configured, use the **show running-config** command in the privileged EXEC configuration mode. If the **no call-waiting beep generate** and **no call-waiting beep accept** commands are configured, the **show running-config** output will display the **no call-waiting beep** command.

If you configure a button to have a silent ring using the **s** option of the **button** command, you will not hear a call-waiting beep regardless of whether the ephone-dn associated with the button is configured to generate a call-waiting beep.

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

Examples

The following example configures ephone-dn 1 and ephone-dn 2 not to accept and not to generate call-waiting beeps:

```
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 2588
Router(config-ephone-dn)# no call-waiting beep accept
Router(config-ephone-dn)# no call-waiting beep generate
Router(config-ephone-dn)# exit

Router(config)# ephone-dn 2
Router(config-ephone-dn)# number 2589
Router(config-ephone-dn)# no call-waiting beep accept
Router(config-ephone-dn)# no call-waiting beep generate
Router(config-ephone-dn)# exit
```

The following example uses an ephone-dn template to set the same attributes as in the previous example:

```
Router(config)# ephone-dn-template 5
Router(config-ephone-dn-template)# no call-waiting beep accept
Router(config-ephone-dn-template)# no call-waiting beep generate
Router(config-ephone-dn-template)# exit

Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 2588
Router(config-ephone-dn)# ephone-dn-template 5
Router(config-ephone-dn)# exit

Router(config)# ephone-dn 2
Router(config-ephone-dn)# number 2589
Router(config-ephone-dn)# ephone-dn-template 5
Router(config-ephone-dn)# exit
```

The following example configures ephone-dn 1 and ephone-dn 2 to switch back to accept call-waiting beeps. Ephone-dn 1 and ephone-dn 2 now accept but do not generate call-waiting beeps.

```
Router(config)# ephone-dn 1
Router(config-ephone-dn)# call-waiting beep accept
Router(config)# ephone-dn 2
Router(config-ephone-dn)# call-waiting beep accept
```

Related Commands

Command	Description
show running-config	Displays the contents of the currently running configuration file or the configuration for a specific interface, or map class information.

call-waiting ring

To allow an ephone-dn to use a ring sound for call-waiting notification, use the **call-waiting ring** command in ephone-dn or ephone-dn-template configuration mode. To disable the ring notification, use the **no** form of this command.

call-waiting ring

no call-waiting ring

Syntax Description

This command has no arguments or keywords.

Command Default

The ephone-dn accepts call waiting and uses beeps for notification.

Command Modes

Ephone-dn configuration
Ephone-dn-template configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)XL	Cisco CME 3.2.1	This command was introduced.
12.3(14)T	Cisco CME 3.3	This command was integrated into Cisco IOS Release 12.3(14)T.
12.4(4)XC	Cisco Unified CME 4.0	This command was made available in ephone-dn-template configuration mode.
12.4(9)T	Cisco Unified CME 4.0	This command in ephone-dn-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

To use a ring sound for call-waiting notification on an ephone-dn, you must ensure that the ephone-dn will accept secondary calls while it is connected to another line. The acceptance of call waiting is the default ephone-dn behavior. However, the **no call-waiting beep accept** command can change this default so an ephone-dn does not accept call waiting. This command must be removed for ringing notification to work.

The **call-waiting ring** command will automatically disable a call-waiting beep configuration.

If you configure a button to have a silent ring using the `s` option of the **button** command, you will not hear a call-waiting ring regardless of whether the ephone-dn associated with the button is configured to generate a call-waiting ring.

Note The call-waiting ring option cannot be used on the Cisco Unified IP Phone 7902, Cisco Unified IP Phone 7905, or Cisco Unified IP Phone 7912. Do not use the **call-waiting ring** command for ephone-dns associated with these types of phones.

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

Examples

The following example configures ephone-dn 1 and ephone-dn 2 to use ringing for their call-waiting notification:

```
Router(config)# ephone-dn 1
Router(config-ephone-dn)# call-waiting ring

Router(config)# ephone-dn 2
Router(config-ephone-dn)# no call-waiting ring
```

The following example uses an ephone-dn template to set the same attributes as in the previous example:

```
Router(config)# ephone-dn-template 9
Router(config-ephone-dn-template)# call-waiting ring
Router(config-ephone-dn-template)# exit

Router(config)# ephone-dn-template 10
Router(config-ephone-dn-template)# no call-waiting ring
Router(config-ephone-dn-template)# exit

Router(config)# ephone-dn 1
Router(config-ephone-dn)# ephone-dn-template 9
Router(config-ephone-dn)# exit

Router(config)# ephone-dn 2
Router(config-ephone-dn)# ephone-dn-template 10
Router(config-ephone-dn)# exit
```

Related Commands

Command	Description
call-waiting beep	Allows call-waiting beeps to be accepted by or generated from an ephone-dn.

capf-auth-str

To define a string of digits that a user enters at the phone for CAPF authentication, use the **capf-auth-str** command in ephone configuration mode. To return to the default, use the **no** form of this command.

capf-auth-str *digit-string*

no capf-auth-str

Syntax Description

digit-string	String of digits that a phone user enters at the phone for CAPF authentication.

Command Default

No authentication string exists for the phone.

Command Modes

Ephone configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.

Usage Guidelines

This command is used with Cisco Unified CME phone authentication to create or remove an authentication string (Personal Identification Number or PIN) for the specified secure ephone. Use this command if the **auth-string** keyword is specified in the **auth-mode** command. Once you specify a CAPF authentication string, it becomes part of the ephone configuration. This value can also be set globally or per ephone using the **auth-string** command in CAPF configuration mode.

Use the **show capf-server auth-str** command to display configured authentication strings.

When a phone is configured for a certificate upgrade that requires auth-string authentication, the CAPF initiation needs to be performed manually by the phone user using the following steps:

1. Press the Settings button.
2. If the configuration is locked, press **# (asterisk, asterisk, pound sign) to unlock it.
3. Scroll down the menu and select Security Configuration.
4. Scroll down the next menu to LSC and press the Update soft key.
5. When prompted for the authentication string, enter the string provided by the system administrator.

Examples

The following example specifies the type of authentication for ephone 392 is an authentication string that is entered from the phone, and then defines the string as 38593.

```
ephone 392
 button 1:23 2:24 3:25
 device-security-mode authenticated
 cert-oper upgrade auth-mode auth-string
 capf-auto-str 38593
S
```

Related Commands

Command	Description
auth-mode	Specifies the type of authentication to use during CAPF sessions.
auth-string	Generates or removes authentication strings for one or all secure ephones.
show capf-server	Displays configuration and session information for the CAPF server.

capf-server

To enter CAPF-server configuration mode to set CAPF server parameters, use the **capf-server** command in global configuration mode. To remove the CAPF server configuration, use the **no** form of this command.

capf-server

no capf-server

Syntax Description

This command has no keywords or arguments.

Command Default

No CAPF server configuration is present.

Command Modes

Global configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.

Usage Guidelines

This command is used with Cisco Unified CME phone authentication.

Examples

The following example sets parameters for the CAPF server:

```
Router(config)# capf-server
Router(config-capf-server)# source address 10.10.10.1
Router(config-capf-server)# trustpoint-label server25
Router(config-capf-server)# cert-oper upgrade all
Router(config-capf-server)# cert-enroll-trustpoint server12 password 0 x8oWiet
Router(config-capf-server)# auth-mode auth-string
Router(config-capf-server)# auth-string generate all
Router(config-capf-server)# port 3000
Router(config-capf-server)# keygen-retry 5
Router(config-capf-server)# keygen-timeout 45
Router(config-capf-server)# phone-key-size 2048
```

cert-enroll-trustpoint

To enroll the CAPF with the CA or RA, use the **cert-enroll-trustpoint** command in CAPF-server configuration mode. To remove an enrollment, use the **no** form of this command.

cert-enroll-trustpoint *ca-label* **password** {**0** | **1**} *password-string*

no cert-enroll-trustpoint

Syntax Description

ca-label	PKI trustpoint label for the CA or for the RA if an RA is being used.
password	Values that follow apply to the password.
0 | **1**	Encryption status of the password string that follows. • **0**—Encrypted. • **1**—Clear text. **Note** This option refers to the way that you want the password to appear in show command output and not to the way that you enter the password in this command.
password-string	Alphanumeric challenge password that is required for certificate enrollment.

Command Default

The CAPF server is not enrolled with the CA or RA.

Command Modes

CAPF-server configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.

Usage Guidelines

This command is used with Cisco Unified CME phone authentication.

Examples

The following example specifies that the CAPF server should enroll with the trustpoint named server12 (the CA) using the password x8oWiet, which should be encrypted:

```
Router(config)# capf-server
Router(config-capf-server)# source address 10.10.10.1
Router(config-capf-server)# trustpoint-label server25
Router(config-capf-server)# cert-oper upgrade all
Router(config-capf-server)# cert-enroll-trustpoint server12 password 0 x8oWiet
Router(config-capf-server)# auth-mode auth-string
Router(config-capf-server)# auth-string generate all
Router(config-capf-server)# port 3000
Router(config-capf-server)# keygen-retry 5
Router(config-capf-server)# keygen-timeout 45
Router(config-capf-server)# phone-key-size 2048
```

cert-oper (CAPF-server)

To initiate the specified certificate operations for all ephones, use the **cert-oper** command in CAPF-server configuration mode. To return to the default, use the **no** form of this command.

cert-oper {**delete all** | **fetch all** | **upgrade all**}

no cert-oper

Syntax Description

delete all	Remove all phone certificates.
fetch all	Retrieve all phone certificates for troubleshooting.
upgrade	Install or upgrade all phone certificates.

Command Default

A certificate operation is not specified.

Command Modes

CAPF-server configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.

Usage Guidelines

This command is used with Cisco Unified CME phone authentication. The **cert-oper** command in ephone configuration mode can also be used to specify certificate operations for individual ephones. Note that the keywords for that command are different than for this command.

Examples

The following example instructs the CAPF server to upgrade all phone certificates.

```
Router(config)# capf-server
Router(config-capf-server)# source address 10.10.10.1
Router(config-capf-server)# trustpoint-label server25
Router(config-capf-server)# cert-oper upgrade all
Router(config-capf-server)# cert-enroll-trustpoint server12 password 0 x8oWiet
Router(config-capf-server)# auth-mode auth-string
Router(config-capf-server)# auth-string generate all
Router(config-capf-server)# port 3000
Router(config-capf-server)# keygen-retry 5
Router(config-capf-server)# keygen-timeout 45
Router(config-capf-server)# phone-key-size 2048
```

Related Commands

Command	Description
cert-oper (ephone)	Configures an individual ephone for certificate activity.

cert-oper (ephone)

To initiate a certificate activity for an individual ephone and specify the type of authentication, use the **cert-oper** command in ephone configuration mode. To return to the default, use the **no** form of this command.

cert-oper {**delete** | **fetch** | **upgrade**} {**auth-string** | **LSC** | **MIC** | **null-string**}

no cert-oper

Syntax Description

delete	Remove phone certificate.
fetch	Retrieve phone certificate for troubleshooting.
upgrade	Install or upgrade phone certificate.
auth-string	The phone user enters a special authentication string at the phone. See the "Usage Guidelines" section.
LSC	The phone provides its phone certificate for authentication. Precedence is given to a Locally Significant Certificate (LSC) if one exists.
MIC	The phone provides its phone certificate for authentication. Precedence is given to a Manufacturing Inserted Certificate (MIC) if one exists.
null-string	No authentication is used.

Command Default

No certificate activity is specified.

Command Modes

Ephone configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.

Usage Guidelines

This command is used with Cisco Unified CME phone authentication.

Phones require digitally signed certificates to participate in secure communications. In most cases, IP phones are shipped with MICs. At times, it may become necessary to replace an expired or revoked MIC. The CAPF server acts as a proxy for fetching a new certificate for the IP phone. The certificate thus issued through CAPF is an LSC.

When a phone is configured for certificate upgrade with auth-string authentication, the password string is entered into the ephone configuration using the **capf-auth-str** command. CAPF initiation is then manually performed at the phone using the following steps:

1. Press the Settings button.
2. If the configuration is locked press **# (asterisk, asterisk, pound sign) to unlock it.
3. Scroll down the menu and select Security Configuration.
4. Scroll down the next menu to LSC and press the Update soft key.

5. When prompted for the authentication string, enter the string provided by the system administrator.

To initiate certificate operations for all phones, use the **cert-oper** command in CAPF-server configuration mode. Note that the keywords for that command are different than for this command.

Examples

The following example specifies the type of authentication for ephone 392 is an authentication string that is entered from the phone, and then defines the string as 38593.

```
ephone 392
 button 1:23 2:24 3:25
 device-security-mode authenticated
 cert-oper upgrade auth-mode auth-string
 capf-auto-str 38593
```

Related Commands

Command	Description
capf-auth-str	Defines a string of digits that a user enters at the phone for CAPF authentication
cert-oper (CAPF server)	Initiates certificate operations for all ephones.

clear telephony-service ephone-attempted-registrations

To empty the log of ephones that unsuccessfully attempt to register with Cisco Unified CME, use the **clear telephony-service ephone-attempted-registrations** command in privileged EXEC configuration mode.

clear telephony-service ephone-attempted-registrations

Syntax Description

This command has no keywords or arguments.

Command Default

The log continues to accumulate attempted ephone registrations.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

The **no auto-reg-ephone** command blocks the automatic registration of ephones whose MAC addresses are not explicitly listed in the configuration. When automatic registration is blocked, Cisco Unified CME records the MAC addresses of phones that attempt to register but cannot because they are blocked.

Use the **show ephone attempted-registrations** command to view the list of phones that have attempted to register but have been blocked. The **clear telephony-service ephone-attempted-registrations** command clears the list.

Examples

The following example clears the attempted-registrations log.

```
Router# clear telephony-service ephone-attempted-registrations
```

Related Commands

Command	Description
auto-reg-ephone	Enables automatic registration of ephones with Cisco Unified CME.
show ephone attempted-registrations	Displays the log of ephones that unsuccessfully attempt to register with Cisco CME.

clear telephony-service conference hardware number

To drop all conference parties and clear the conference call, use the **clear telephony-service conference hardware number** command in privileged EXEC mode.

clear telephony-service conference hardware number *number*

Syntax Description

number	Conference telephone or extension number.

Command Default

The conference call continues with all current parties.

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

Use the **show telephony-service conference hardware** command to display the active hardware conferences. Use the **clear telephony-service conference hardware number** command to clear the desired conference.

Examples

The following example clears the conference number 1111 and drops all its parties:

```
Router# clear telephony-service conference hardware number 1111
```

Related Commands

Command	Description
show telephony-service conference hardware	Displays information about hardware conferences in a Cisco CME system.

clear telephony-service xml-event-log

To clear the event table used for the Cisco Unified CME XML application, use the **clear telephony-service xml-event-log** command in privileged EXEC mode.

clear telephony-service xml-event-log

Syntax Description

This command has no keywords or arguments.

Command Default

The XML event table is not cleared.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

The **show fb-its-log** command displays the contents of the XML event table.

Examples

The following example clears the entries from the XML event table:

```
Router# clear telephony-service xml-event-log
```

Related Commands

Command	Description
show fb-its-log	Displays Cisco Unified CME XML API information.

cnf-file

To specify the generation of different phone configuration files by type of phone or by individual phone, use the **cnf-file** command in telephony-service configuration mode. To return to the default, use the **no** form of this command.

cnf-file {**perphonetype** | **perphone**}

no cnf-file {**perphonetype** | **perphone**}

Syntax Description

perphonetype	A separate configuration file is generated for each type of phone.
perphone	A separate configuration file is generated for each phone.

Command Default

A single configuration file is used for all phones.

Command Modes

Telephony-service

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Configuration files can be applied in the following ways:

- Per system—This is the default. All phones use a single configuration file. The default user and network locale in a single configuration file are applied to all phones in the Cisco Unified CME system. Alternative and user-defined user and network locales are not supported. To use the per-system option, either do not use the **cnf-file** command or use the **no cnf-file** command to reset the option from a different configuration.
- Per phone type—This setting creates separate configuration files for each phone type. For example, all Cisco Unified IP Phone 7960s use XMLDefault7960.cnf.xml, and all Cisco Unified IP Phone 7905s use XMLDefault7905.cnf.xml. All phones of the same type use the same configuration file, which is generated using the default user and network locale. To create configuration files per phone type, use the **cnf-file perphonetype** command. This option is not supported if the location option is system.
- Per phone—This setting creates a separate configuration file for each phone, by MAC address. For example, a Cisco Unified IP Phone 7960 with the MAC address 123.456.789 creates the per-phone configuration file SEP123456789.cnf.xml. The configuration file for a phone is generated with the default user and network locale unless a different user and network locale is applied to the phone using an ephone template. To create configuration files per phone type, use the **cnf-file perphone** command. This option is not supported if the location option is system.

To reset the type of configuration file to the default, use the **no** form of this command and the keyword that you previously used to set the type.

This feature is supported only on the following phones:

- Cisco Unified IP Phones 7940 and 7940G
- Cisco Unified IP Phones 7960 and 7960G
- Cisco Unified IP Phone 7970G
- Cisco Unified IP Phone 7971G-GE

Examples

The following example selects flash memory as the configuration file storage location and per-phone as the type of configuration files that the system should generate.

```
telephony-service
 cnf-file location flash:
 cnf-file perphone
```

Related Commands

Command	Description
cnf-file location	Specifies a storage location for phone configuration files.

cnf-file location

To specify a storage location for phone configuration files, use the **cnf-file location** command in telephony-service configuration mode. To return to the default, use the **no** form of this command.

cnf-file location {**flash:** | **slot0:** | **tftp** *tftp-url*}

no cnf-file location {**flash:** | **slot0:** | **tftp** *tftp-url*}

Syntax Description

flash:	Router flash memory.
slot0:	Router slot 0 memory.
tftp *tftp-url*	External TFTP server at the specified URL.

Command Default

A single phone configuration file is stored in system memory and is used by all phones.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

You can specify any of the following four locations to store configuration files:

- System—This is the default. When the system is the storage location, there can be only one default configuration file and it is used for all phones in the system. All phones, therefore, use the same user locale and network locale. User-defined user and network locales are not supported. To use the system location, either do not use the **cnf-file location** command to specify a location or use the **no cnf-file location** {**flash:** | **slot0:** | **tftp** *url*} command to reset the option from a previous, different location.
- Flash or slot 0—When flash or slot 0 memory on the router is the storage location, you can create additional configuration files that can be applied per phone type or per individual phone. Up to five user-defined user and network locales can be used in these configuration files. To store configuration files in flash or slot 0, use the **cnf-file location flash:** or **cnf-file location slot0:** command. The generation of configuration files on flash or slot 0 can take up to a minute, depending on the number of files that are being generated.

Note When the storage location chosen is flash and the file system type on this device is Class B(LEFS), make sure to check free space on the device periodically and use the **squeeze** command to free the space used up by deleted files. Unless you use the **squeeze** command, the space used by the moved or deleted configuration files cannot be used by other files.

- TFTP—When an external TFTP server is the storage location, you can create additional configuration files that can be applied per phone type or per individual phone. Up to five user-defined user and network locales can be used in these configuration files. To store configuration files on an external TFTP server, use the **cnf-file location tftp** *url* command.

TFTP does not support file deletion. When configuration files are updated, they overwrite any existing configuration files with the same name. If you change the configuration file location, files are not deleted from the TFTP server.

To reset the location to the default, use the **no** form of this command and the keyword that you previously used to set the location.

This feature is supported only on the following phones:

- Cisco Unified IP Phones 7940 and 7940G
- Cisco Unified IP Phones 7960 and 7960G
- Cisco Unified IP Phone 7970G
- Cisco Unified IP Phone 7971G-GE

Examples

The following example selects flash memory as the configuration file storage location and per-phone as the type of configuration files that the system should generate.

```
telephony-service
 cnf-file location flash:
 cnf-file perphone
```

Related Commands

Command	Description
cnf-file	Specifies the use of different phone configuration files by type of phone or by individual phone.

codec (ephone)

To select a preferred codec for Cisco Unified CME to use when setting up calls for a phone, use the **codec** command in ephone or ephone-template configuration mode. To return to the default, use the **no** form of this command.

codec {**g711ulaw** | **g729r8** [**dspfarm-assist**]}

no codec

Syntax Description

g711ulaw	Selects G.711 mu-law codec.
g729r8	Selects G.729r8 codec.
dspfarm-assist	Attempts to use DSP-farm resources for transcoding the segment between the phone and the Cisco Unified CME router if G.711 is negotiated for the call. **Note** The **dspfarm-assist** keyword is ignored if the SCCP endpoint type is ATA, VG224, or VG248.

Command Default

G.711 mu-law

Command Modes

Ephone configuration
Ephone-template configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command can be used to help save network bandwidth for a remote IP phone.

When you use the **codec** command without the **dspfarm-assist** keyword, you only affect calls between two phones on the Cisco Unified CME router (such as between an IP phone and another IP phone or between an IP phone and an FXS analog phone). The command has no effect on a call directed through a VoIP dial peer unless you use the **dspfarm-assist** keyword.

For calls to other phones in the same Cisco Unified CME system, an IP phone that is configured to use G.729 will always have its calls set up to use G.729. If the phone participates in a call on a line that is shared with a phone that is configured for G.729 or is paged together with another phone that is configured for G.729, it must use G.729.

For calls to phones that are not in the same Cisco Unified CME system (such as VoIP calls), the codec is negotiated based on the protocol that is used for the call (such as H.323). The Cisco Unified CME system plays no part in the negotiation.

When you use the **g729r8** keyword to select the G.729r8 codec for the RTP segment between the IP phone and the Cisco Unified CME router and you also use the **dspfarm-assist** keyword, the router attempts to use DSP-farm resources in the following way. If the IP phone is in a VoIP call (H.323 or SIP) or a Cisco Unified CME conference in which the codec must be set to G.711, the router uses configured DSP-farm resources to attempt to return the segment between the phone and the Cisco Unified CME router to G.729. Note that adequate DSP resources must be appropriately configured separately.

You should consider your options carefully when deciding to use the **dspfarm-assist** keyword with the **codec** command. The benefit is that it allows calls to use the G.729r8 codec on the call leg between the IP phone and the Cisco Unified CME router, which saves network bandwidth. The disadvantage is that for situations requiring G.711 codecs, such as conferencing and Cisco Unity Express, DSP resources that are possibly scarce will be used to transcode the call, and delay will be introduced while voice is shuttled to and from the DSP. In addition, the overuse of this feature can mask configuration errors in the codec selection mechanisms involving dial peers and codec lists.

Therefore, it is recommended that the **dspfarm-assist** keyword be used sparingly and only when absolutely required for bandwidth savings or when you know the phone will be participating very little, if at all, in calls that require a G.711 codec.

If the **dspfarm-assist** keyword has been configured for a phone and a DSP resource is not available when needed for transcoding, a phone registered to the local Cisco Unified CME router will use G.711 instead of G.729r8. This is not true for non-SCCP call legs; if no DSP resource is available for the transcoding required for a conference, for example, the conference will not be created.

Note The **dspfarm-assist** keyword is ignored if the SCCP endpoint type is ATA, VG224, or VG248.

This command can be part of an ephone template that is applied to several ephones. If you use an ephone template to apply a command to a phone and you also use the same command in ephone configuration mode for the same phone, the value that you set in ephone configuration mode has priority.

Examples

The following example selects the G.729 codec with DSP farm assist for calls that are being set up for ephone 25:

```
ephone 25
 button 1:37
 codec g729r8 dspfarm-assist
```

The following example uses ephone template 1 to apply the G.729 codec preference to ephone 25:

```
ephone-template 1
 codec g729r8

ephone 25
 button 1:37
 ephone-template 1
```

codec (voice register pool)

To specify the codec to be used when setting up a call for a SIP phone or group of SIP phones in Cisco Unified CME or Cisco Unified SIP SRST, use the **codec** command in voice register pool configuration mode. To disable a specified codec, use the **no** form of this command.

codec *codec-type* [*bytes*]

no codec

Syntax Description

codec-type	Specifies the preferred codec: • **g711alaw**—G.711 a–law 64,000 bps • **g711ulaw**—G.711 mu–law 64,000 bps. • **g729r8—**G.729 8000 bps (this is the default).
bytes	(Optional) Specifies the number of bytes in the voice payload of each frame.

Command Default

g729r8

Command Modes

Voice register pool configuration

Command History

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

Usage Guidelines

When you use the codec command, you affect calls between two phones on the same Cisco Unified CME or Cisco Unified SRST router (such as between an IP phone and another IP phone or between an IP phone and an FXS analog phone).

Use this command to change the automatically selected default codec for the dial peer dynamically created when the SIP phone registers.

If codec values for the dial peers of a connection do not match, the call fails. The default codec for the POTS dial peer for a SCCP phone is G.711 and the default codec for a VoIP dial peer for a SIP phone is G.729. If neither the SCCP phone nor the SIP phone has been specifically configured to change the codec, calls between the two IP phones on the same router will produce a busy signal caused by the mismatched default codecs. For calls to other phones on the same Cisco router, a SIP phone that is configured to use G.711 will always have its calls set up to use G.711. If the phone participates in a call on a line that is shared with a phone that is configured for G.711, it must use G.711.

For calls to external phones; that is, phones that are not in the same Cisco Unified CME (such as VoIP calls), the codec is negotiated based on the protocol that is used for the call (such as H.323). Cisco Unified CME plays no part in the negotiation.

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

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

Examples

The following example shows how to set codec complexity to g711 for a SIP phone in pool 1:

```
Router(config)# voice register pool 1
Router(config-register-pool)# codec g711ulaw
...
```

The following partial sample from the **show voice register pool** command shows the configuration for voice register pool 1:

```
pool tag 1
Config
 MAC address is 0012.DABF.26BE
 Type is 7960
 Number list 1: dn 1
 Proxy Ip address is 0.0.0.0
 Codec is g711ulaw
...
Dialpeers created
dial-peer voice 4003 voip
 destination-pattern 6667
 session target ipv
 session protocol sip2v
 codec g711ulaw
```

Related Commands

Command	Description
id (voice register pool)	Explicitly identifies a locally available individual SIP phone, or when running Cisco Unified SIP SRST, set of SIP phones.
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.
voice-class codec	Assigns a previously configured codec selection preference list.
voice register pool	Enters voice register pool configuration mode for SIP phones.

conference (ephone-dn)

To configure a conference associated with a directory number, use the **conference** command in ephone-dn configuration mode. To disable a conference associated with a directory number, use the **no** form of this command.

conference {**ad-hoc** | **meetme**}

no conference {**ad-hoc** | **meetme**}

Syntax Description

ad-hoc	Configures ad hoc conferences.
meetme	Configures meet-me conferences.

Command Default

No conference is associated with the directory number.

Command Modes

Ephone-dn configuration

Command History

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

Usage Guidelines

Ad hoc conferences are those which begin as a call between the conference creator and another party. The creator then calls other parties and adds them to the original call creating a conference.

Meet-me conferences have a designated meet-me telephone or extension number that all parties call to join the conference. The conference creator initiates the meet-me conference by pressing the MeetMe soft key, then dialing the meet-me number. Other parties join the conference by dialing the meet-me number.

Use the **ephone-dn** command to configure enough extensions for your conference needs. Each extension can handle two conference parties if the **dual-line** keyword is used with he **ephone-dn** command as shown in the example below. Use the **show ephone-dn** command to display phone information for the extension.

Examples

The following example configures extension 9001 as a four-party meet-me conference number.

```
Router(config)# ephone-dn 1 dual-line
Router(config-ephone-dn)# number 9001
Router(config-ephone-dn)# conference meetme

Router(config)# ephone-dn 2 dual-line
Router(config-ephone-dn)# number 9001
Router(config-ephone-dn)# conference meetme
```

You must configure additional directory numbers to add more parties to the conference.

Related Commands

Command	Description
ephone-dn	Enters ephone-dn configuration mode for the purposes of creating and configuring an extension (ephone-dn) for a Cisco Unified IP phone line.
show ephone-dn	Displays phone information for specified dn-tag or for all dn-tags.

conference (voice register template)

To enable the soft key for conference in a SIP phone template, use the **conference** command in voice register template configuration mode. To disable the soft key, use the **no** form of this command.

conference

no conference

Syntax Description

This command has no arguments or keywords.

Defaults

Enabled

Command Modes

Voice register template configuration

Command History

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

Usage Guidelines

This command enables a soft key for conference in the specified template which can then be applied to SIP phones. The conference soft key is enabled by default. To disable the conference soft key, use the **no conference** command. To apply a template to a SIP phone, use the **template** command in voice register pool configuration mode.

Examples

The following example shows how to disable the conference soft key in template 1:

```
Router(config)# voice register template 1
Router(config-register-temp)# no conference
```

Related Commands

Command	Description
template (voice register pool)	Applies a template to a SIP phone.
transfer-attended (voice register template)	Enables a soft key for attended transfer in a SIP phone template.
transfer-blind (voice register template)	Enables a soft key for blind transfer in a SIP phone template.
voice register pool	Enters voice register pool configuration mode for SIP phones.
voice register template	Enters voice register template configuration mode and defines a template of common parameters for SIP phones.

conference add-mode

To configure the mode for adding parties to ad hoc hardware conferences, use the **conference add-mode** command in ephone or ephone-template configuration mode. To return to the default, use the **no** form of this command.

conference add-mode [**creator**]

no conference add-mode [**creator**]

Syntax Description

creator	Specifies that only the creator can add parties.

Command Default

Any party can add other parties provided the creator remains in the conference.

Command Modes

Ephone configuration
Ephone-template configuration

Command History

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

Usage Guidelines

For more control of conference participation, use this command to specify that only the creator can add new parties. This configuration ensures that no one can add parties to the conference without the creator's knowledge.

Use this command to configure an ephone directly in ephone configuration mode, or use it to configure an ephone template in ephone-template configuration mode. Use the **ephone-template** command in ephone configuration mode to apply the ephone template to one or more ephones. Use the **show telephony-service ephone** command to display the add and drop modes for the ephone. Use the **show telephony-service ephone-template** command to display the ephone template.

Examples

The following example configures ad hoc hardware conferences so that only the creator can add participants.

```
Router(config)# ephone 1
Router(config-ephone)#  conference add-mode creator
```

Related Commands

Command	Description
ephone-template	Creates an ephone template to configure a set of phone features and to enter ephone-template configuration mode.
ephone-template (ephone)	Applies an ephone template to an ephone.
show telephony-service ephone	Displays configuration for the Cisco IP phones.
show telephony-service ephone-template	Displays the contents of ephone-templates.

conference admin

To configure the ephone as the ad hoc and meet-me hardware conference administrator, use the **conference admin** command in ephone or ephone-template configuration mode. To return to the defaults, use the **no** form of this command.

conference admin

no conference admin

Syntax Description

This command has no arguments or keywords.

Command Default

This ephone is not the ad hoc and meet-me hardware conference administrator.

Command Modes

Ephone configuration
Ephone-template configuration

Command History

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

Usage Guidelines

Use this command to configure an ad hoc and meet-me hardware conference administrator. The administrator can:

- Dial in to any conference directly through the conference number
- Use the ConfList soft key to list conference parties
- Remove any party from any conference

The administrator can control the use of conference bridges by enforcing time limits and making sure conference bridges are available for scheduled meetings.

Use this command to configure an ephone directly in ephone configuration mode, or use it to configure an ephone template in ephone-template configuration mode. Use the **ephone-template** command in ephone configuration mode to apply the ephone template to one or more ephones. Use the **show telephony-service ephone** command to display the add and drop modes for the ephone. Use the **show telephony-service ephone-template** command to display the ephone template.

Examples

The following example configures ephone 1 as the ad hoc and meet-me hardware conference administrator.

```
Router(config)# ephone 1
Router(config-ephone)#  conference admin
```

Related Commands

Command	Description
ephone-template	Creates an ephone template to configure a set of phone features and to enter ephone-template configuration mode.
ephone-template (ephone)	Applies an ephone template to an ephone.
show telephony-service ephone	Displays configuration for the Cisco IP phones.
show telephony-service ephone-template	Displays the contents of ephone-templates.

conference drop-mode

To configure the mode for terminating ad hoc hardware conferences when parties drop out, use the **conference drop-mode** command in ephone or ephone-template configuration mode. To return to the default, use the **no** form of this command.

conference drop-mode [**creator** | **local**]

no conference drop-mode [**creator** | **local**]

Syntax Description

creator	Specifies that the active conference terminates when the creator hangs up.
local	Specifies that the active conference terminates when the last local party in the conference hangs up or drops out of the conference.

Command Default

The conference is not dropped, regardless of whether the creator hangs up, provided three parties remain in the conference.

Command Modes

Ephone configuration
Ephone-template configuration

Command History

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

Usage Guidelines

For more control of conference participation, use this command to specify that the conference drops when the creator hangs up (see the example). This configuration ensures that the conference cannot continue without the creator’s presence.

Use this command to configure an ephone directly in ephone configuration mode, or use it to configure an ephone template in ephone-template configuration mode. Use the **ephone-template** command in ephone configuration mode to apply the ephone template to one or more ephones. Use the **show telephony-service ephone** command to display the add and drop modes for the ephone. Use the **show telephony-service ephone-template** command to display the ephone template.

Examples

The following example configures ad hoc hardware conferences so that only the creator can add participants and the active conference terminates when the creator hangs up.

```
Router(config)# ephone 1
Router(config-ephone)#  conference drop-mode creator
```

Related Commands

Command	Description
ephone-template	Creates an ephone template to configure a set of phone features and to enter ephone-template configuration mode.
ephone-template (ephone)	Applies an ephone template to an ephone.
show telephony-service ephone	Displays configuration for the Cisco IP phones.
show telephony-service ephone-template	Displays the contents of ephone-templates.

conference hardware

To configure a Cisco Unified CallManager Express system for hardware conferencing only, use the **conference hardware** command in telephony-service configuration mode. To return to the default, three-party software conferencing, use the **no** form of this command.

conference hardware

no conference hardware

Syntax Description

This command has no arguments or keywords.

Command Default

Three-party ad hoc software conferencing.

Command Modes

Telephony-service configuration

Command History

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

Usage Guidelines

Software conferencing allows a maximum of three parties in a conference. Use this command to take advantage of DSP farm resources for hardware conferencing which allows ad hoc conferences with more than three parties.

If you need ad hoc hardware conferences, you must use this command to configure DSP farm hardware conferencing. You can configure other conferencing features using the **conference-join custom-cptone**, **conference-leave custom-cptone**, and **maximum conference-party** commands in DSP farm profile configuration mode. Use the **show dspfarm profile** command to display the DSP farm profile.

Examples

The following example configures hardware conferencing as the default for ad hoc conferences on this Cisco Unified CallManager Express system:

```
Router(config)# telephony-service
Router(config-telephony)# conference hardware
```

Related Commands

Command	Description
conference-join custom-cptone	Associates a custom call-progress tone to indicate joining a conference with a DSP farm profile.
conference-leave custom-cptone	Associates a custom call-progress tone to indicate leaving a conference with a DSP farm profile.

Command	Description
maximum conference-party	Configures the maximum number of conference participants allowed in each conference.
show dspfarm profile	Display configured digital signal processor (DSP) farm profile information.

cor (ephone-dn)

This command is now documented as the **corlist** command. For complete command information, see the **corlist** command page.

cor (voice register pool)

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

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

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

Syntax Description

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

Command Default

None

Command Modes

Voice register pool configuration

Command History

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

Usage Guidelines

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

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

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

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

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

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

Examples

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

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

Related Commands

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

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

corlist

This command was previously documented as the **cor** command.

To apply a class of restriction (COR) to the dial peers associated with a Cisco CME extension (ephone-dn), use the **corlist command in** ephone-dn configuration mode. To disable the COR associated with an extension, use the **no** form of this command.

corlist {**incoming** | **outgoing**} *corlist-name*

no corlist {**incoming** | **outgoing**}

Syntax Description

incoming	Specifies a COR list to be used by incoming dial peers.
outgoing	Specifies a COR list to be used by outgoing dial peers.
corlist-name	COR list name.

Command Default

No COR is used by the dial peers associated with the extension that is being configured.

Command Modes

Ephone-dn configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(2)XT	Cisco ITS 2.0	This command was introduced on the Cisco 1750, Cisco 1751, Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	Cisco ITS 2.0	This command was implemented on the Cisco 2600-XM and Cisco 2691.
12.2(11)T	Cisco ITS 2.01	This command was implemented on the Cisco 1760.

Usage Guidelines

COR is used to specify which incoming ephone-dn dial peer can use which outgoing ephone-dn dial peer to make a call. COR denies certain call attempts on the basis of the incoming and outgoing class of restrictions that have been provisioned on the dial peers. This functionality provides flexibility in network design, allows administrators to block calls (for example, calls to 900 numbers), and applies different restrictions to call attempts from different originators.

Each dial peer can be provisioned with an incoming and an outgoing COR list.

The **corlist incoming** and **corlist outgoing** commands in dial-peer configuration mode perform these functions for dial peers that are not associated with ephone-dns. The **dial-peer cor list** and **member** commands define the sets of capabilities, or COR lists, that are referred to in the **corlist** commands.

Examples

The following example shows how to set a COR parameter for incoming calls to dial peers associated with the extension that has the dn-tag 1:

```
Router(config)# ephone-dn 1
Router(config-ephone-dn)# corlist incoming corlist1
```

Related Commands

Command	Description
corlist incoming	Specifies the COR list to be used when a specified dial peer acts as the incoming dial peer.
corlist outgoing	Specifies the COR list to be used by an outgoing dial peer.
dial-peer cor list	Defines a COR list name.
ephone-dn	Enters ephone-dn configuration mode.

create cnf-files

To build the eXtensible Markup Language (XML) configuration files that are required for IP phones used with Cisco ITS V2.1, Cisco CME 3.0, Cisco Unified CME 4.0 or later versions, use the **create cnf-files** command in telephony-service configuration mode. To remove the configuration files and disable the automatic generation of configuration files, use the **no** form of this command.

create cnf-files

no create cnf-files

Syntax Description

This command has no arguments or keywords.

Command Default

Required XML configuration files are not built.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(11)YT	Cisco ITS 2.1	This command was introduced.
12.2(15)T	Cisco ITS 2.1	This command was integrated into Cisco IOS Release 12.2(15)T.
12.4(4)XC	Cisco Unified CME 4.0	The "Usage Guidelines" section was updated to describe the interaction of this command with new features.

Usage Guidelines

Use this command to build XML configuration files for Cisco IP phones during initial system setup. The XML files created by this command are located in an in-RAM file system at system:/its.

The **no** form of this command removes configuration files and disables automatic configuration file generation.

This command must be used after any of the following actions:

- Using the **cnf-file location** command to change the configuration file location.
- Using the **cnf-file** command to change the type of configuration files.
- Using the **user-locale** command to change the user locale.
- Using the **network-locale** command to change the network locale.
- Using the **user-locale (ephone-template)** or **network-locale (ephone-template)** command to change the user locale or network locale selection in an ephone template.
- Using the **ephone-template (ephone)** command to apply or remove an ephone template from an ephone.

Examples

The following example builds the necessary XML configuration files on the Cisco Unified CME router:

```
Router(config)# telephony-service
Router(config-telephony)# create cnf-files
```

Related Commands

Command	Description
telephony-service	Enters telephony-service configuration mode.

create profile (voice register global)

To generate the configuration profile files required for SIP phones, use the **create profile** command in voice register global configuration mode. To return to the default, use the **no** form of this command.

create profile

no create profile

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled

Command Modes

Voice register global configuration

Command History

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

Usage Guidelines

This command generates the configuration files used for provisioning SIP phones and writes the files to the location specified with the **tftp-path** command.

Examples

The following example shows how to create the configuration profile:

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

Related Commands

Command	Description
file text (voice register global)	Generates ASCII text files for SIP phones.
mode (voice register global)	Enables the mode for provisioning SIP phones in a Cisco CallManager Express (Cisco CME) system.
reset (voice register global)	Performs a complete reboot of all SIP phones associated with a Cisco CME router.
source-address (voice register global)	Identifies the IP address and port through which SIP phones communicate with a Cisco CME router.
tftp-path (voice register global)	Specifies the directory to which the provisioning file for SIP phones in a Cisco CallManager Express (Cisco CME) system will be written.
voice register global	Enters voice register global configuration mode in order to set global parameters for all supported Cisco SIP phones in a Cisco CME or Cisco SIP SRST environment.

credentials

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

credentials

no credentials

Syntax Description

This command has no arguments or keywords.

Command Default

Credentials are not provided.

Command Modes

Global configuration

Command History

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

Usage Guidelines

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

Cisco Unified CME

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

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

Cisco Unified SRST

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

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

Activate this command on all SRST routers.

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

Examples

Cisco Unified CME

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

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

Cisco Unified SRST

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

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

Related Commands

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

ctl-client

To enter CTL-client configuration mode to set parameters for the CTL client, use the **ctl-client** command in global configuration mode. To return to the default, use the **no** form of this command.

ctl-client

no ctl-client

Syntax Description

This command has no keywords or arguments.

Command Default

No CTL-client parameters are set.

Command Modes

Global configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used with Cisco Unified CME phone authentication.

Examples

The following example defines server IP addresses and trustpoints for the CAPF server, the Cisco Unified CME router, and the TFTP server, as well as trustpoints for SAST1 and SAST2. It also specifies that a new CTL file should be generated.

```
Router(config)# ctl-client
Router(config-ctl-client)# server capf 10.2.2.2 trustpoint capftrust
Router(config-ctl-client)# server cme 10.2.2.3 trustpoint cmetp
Router(config-ctl-client)# server tftp 10.2.2.4 trustpoint tftptp
Router(config-ctl-client)# sast1 trustpoint sast1tp
Router(config-ctl-client)# sast2 trustpoint sast2tp
Router(config-ctl-client)# regenerate
```

ctl-service admin

To specify a user name and password to authenticate the client during the CTL protocol, use the **ctl-service admin** command in credentials configuration mode. To return to the default, use the **no** form of this command.

ctl-service admin *username* **secret** {**0** | **1**} *password-string*

no ctl-service admin

Syntax Description

username	Defines the name that will be used to authenticate the client.
secret {**0** | **1**}	Defines a character string for login authentication and whether it will be encrypted when it is stored in the running configuration. • **0**—Not encrypted. • **1**—Encrypted using Message Digest 5 (MD5).
password-string	Character string for login authentication

Command Default

No user name or password is defined for authentication.

Command Modes

Credentials configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used with Cisco Unified CME phone authentication to define a user who will be used to authenticate the CTL client with a CTL provider.

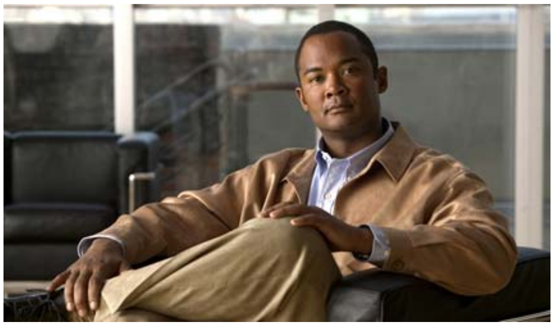
Examples

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

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

Related Commands

Command	Description
credentials	Enters credentials configuration mode to configure a Cisco Unified CME CTL provider certificate or an SRST router certificate.
debug credentials	Sets debugging on the credentials service that runs between a Cisco Unified CME CTL provider and the CTL client or between an SRST router and Cisco Unified CallManager.
show credentials	Displays the credentials settings on a Cisco Unified CME or SRST router.
trustpoint (credentials)	Specifies the name of the trustpoint to be associated with a Cisco Unified CME CTL provider certificate or with an SRST router certificate.



Cisco Unified CME Commands: D

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This chapter contains commands to configure and maintain Cisco Unified Communications Manager Express (formally known as Cisco Unified CallManager Express). The commands are presented in alphabetical order. Some commands required for configuring Cisco Unified Communications Manager Express (Cisco Unified CME) may be found in other Cisco IOS command references. Use the command reference master index or search online to find these commands.

date-format (telephony-service)

To set the date display format on the Cisco IP phones in a Cisco CallManager Express (Cisco CME) system, use the **date-format** command in telephony-service configuration mode. To display the date in the default format, use the **no** form of this command.

date-format {**dd-mm-yy** | **mm-dd-yy** | **yy-dd-mm** | **yy-mm-dd**}

no date-format

Syntax Description

dd-mm-yy **mm-dd-yy** **yy-dd-mm** **yy-mm-dd**	Format in which dates are displayed on the IP phone: • **dd**—Two-digit day. • **mm**—Two-digit month. • **yy**—Two-digit year.

Defaults

mm-dd-yy

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(2)XT	Cisco ITS 2.0	This command was introduced on the Cisco 1750, Cisco 1751, Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745 routers.
12.2(8)T1	Cisco ITS 2.0	This command was implemented on the Cisco 2600-XM and Cisco 2691.
12.2(11)T	Cisco ITS 2.01	This command was implemented on the Cisco 1760.

Examples

The following example sets the date format to day, month, and year, so that December 17, 2004 is represented as 17-12-04.

```
Router(config)# telephony-service
Router(config-telephony)# date-format dd-mm-yy
```

Related Commands

Command	Description
telephony-service	Enters telephony-service configuration mode.

date-format (voice register global)

To set the date display format on SIP phones in a Cisco CallManager Express (Cisco CME) system, use the **date-format** command in voice register global configuration mode. To display the date in the default format, use the **no** form of this command.

date-format {`d/m/d` | `m/d/y` | *`y-d-m`* | `y/d/m` | `y/m/d` | *`yy-m-d`*}

no date-format

Syntax Description

d/m/d **m/d/y** **y-d-m** **y/d/m** **y/m/d** **yy-m-d**	Format in which dates are displayed on the IP phone: • **d**—Day • **m**—Month • **y**—Year

Defaults

mm-dd-yy

Command Modes

Voice register global configuration

Command History

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

Examples

The following example shows how to set the date format to day, month, and year, so that December 17, 2004 is represented as 17-12-04.

```
Router(config)# voice register global
Router(config-register-global)# date-format dd-mm-yy
```

Related Commands

Command	Description
dst auto-adjust (voice register global)	Enables automatic adjustment of daylight saving time on SIP phones.
time-format (voice register global)	Selects a 12-hour clock or a 24-hour clock for the time display format on Cisco IP phones in a SIP CME system.
voice register global	Enters voice register global configuration mode in order to set global parameters for all supported Cisco SIP phones in a Cisco CME or Cisco SIP SRST environment.

default (voice hunt-group)

To set a command to its defaults, use the **default** command in voice hunt-group configuration mode.

default *default-value*

Syntax Description

default-value	One of the voice hunt group configuration commands. Valid choices are as follows: • hops (Peer or longest-idle voice hunt group only) • preference • timeout

Defaults

No default behaviors or values

Command Modes

Voice hunt-group configuration

Command History

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

Usage Guidelines

Use this command to configure the default value for a voice hunt group command.

The default command instructs the voice hunt group to use the default value of the specified command whenever the hunt group is called. This has the same effect as using the no form of the specified command, but the default command clearly specifies which commands are using their default values.

To use the default values for more than one command, enter each command on a separate line.

Examples

The following example shows how to set the default values for two separate voice hunt-group commands:

```
Router(config)# voice hunt-group 4 peer
Router(config-voi-hunt-group)# default hops
Router(config-voi-hunt-group)# default timeout
```

Related Commands

Command	Description
voice hunt-group	Defines a hunt group for SIP phones in a Cisco CallManager Express (Cisco CME) system.

description (ephone)

To provide ephone descriptions for network management systems using an eXtensible Markup Language (XML) query, use the **description** command in ephone configuration mode. To remove a description, use the **no** form of this command.

description *string*

no application

Syntax Description

string	Allows for a maximum of 128 characters, including spaces. There are no character restrictions.

Defaults

No ephone description is configured.

Command Modes

Ephone configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)XL	Cisco CME 3.2.1	This command was introduced
12.3(14)T	Cisco CME 3.3	This command was integrated into Cisco IOS Release 12.3(14)T.

Usage Guidelines

The descriptions configured with this command will appear neither on phone displays nor in show command output. Instead, they are sent to network management systems, such as CiscoView. Network management systems obtain **description** command data by sending an XML ISgetDevice request to a Cisco CME system. The Cisco CME system responds by sending ISDevDesc field data to the network management system, which uses the data to perform such tasks as printing descriptions on screen.

For more information about ISgetDevice requests and the ISDevDesc field, refer to the *XML Provisioning Guide for Cisco CME/SRST*.

Examples

The following example provides a description for ephone 1:

```
Router(config)# ephone 1
Router(config-ephone) description S/N:SK09456FPH3, Location:SJ21- 2nd Floor E5-9, User:
Smith, John
```

Related Commands

Command	Description
ephone	Enters ephone configuration mode for an IP phone for the purposes of creating and configuring an ephone.

description (ephone-dn and ephone-dn-template)

To display a custom text-string description in the header bar of all supported Cisco Unified IP phones, use the **description** command in ephone-dn or ephone-dn-template configuration mode. To return to the default, use the **no** form of this command.

description *string*

no description

Syntax Description

string	Alphanumeric characters to be displayed in the header bar of the phone display. If spaces appear in the string, enclose the string in quotation marks. The maximum string length is 40 characters. **Note** Display behavior depends on phone firmware version.

Command Default

The extension number of the first line on the phone appears in the header bar.

Command Modes

Ephone-dn configuration
Ephone-dn-template configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(11)T	Cisco ITS 2.0.1	This command was introduced.
12.2(11)YT	Cisco ITS 2.1	The number of characters in the string was modified.
12.2(15)T	Cisco ITS 2.1	This command was integrated into Cisco IOS Release 12.2(15)T.
12.4(4)XC	Cisco Unified CME 4.0	This command was made available in ephone-dn-template configuration mode.
12.4(9)T	Cisco Unified CME 4.0	This command in ephone-dn-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Use this command under the ephone-dn that is associated with the first line button on a Cisco Unified IP phone. A typical use for the **description** command is to display in the header bar the entire E.164 telephone number associated with the first line button rather than just the extension number, which is the default.

This command is supported by the following Ip phones:

- Cisco Unified IP Phone 7940 and 7940G
- Cisco Unified IP Phones 7960 and 7960G
- Cisco Unified IP Phone 7970
- Cisco Unified IP Pone 7971

For Cisco Unified IP Phone 7940s and 7940Gs or Cisco Unified IP Phone 7960s and 7960Gs, the *string* is truncated to 14 characters if the text string is greater than 14 characters.

For Cisco Unified IP Phone 797x, all characters in the *string* appear alternately with time and date, each for 5 seconds.

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

Examples

The following example shows how to define a header bar display for a phone on which the first line button is the extension number 50155:

```
Router(config)# ephone-dn 4
Router(config-ephone-dn)# number 50155
Router(config-ephone-dn)# description 888-555-0155
```

The following example shows how to use an ephone-dn template to define a header bar display for a phone on which the first line button is the extension number 50155:

```
Router(config)# ephone-dn-template 3
Router(config-ephone-dn-template)# description "888 555-0155"
Router(config-ephone-dn-template)# exit
Router(config)# ephone-dn 4
Router(config-ephone-dn)# number 50155
Router(config-ephone-dn)# ephone-dn-template 3
```

Related Commands

Command	Description
ephone-dn	Enters ephone-dn configuration mode.
ephone-dn-template	Enters ephone-dn-template configuration mode.
number	Configures a valid number for a Cisco Unified IP phone.

description (ephone-hunt)

To create a label for an ephone hunt group, use the **description** command in ephone-hunt configuration mode. To return this value to the default, use the **no** form of this command.

description *string*

no description

Syntax Description

string	Character string that identifies a hunt group.

Command Default

No description exists for the ephone hunt group.

Command Modes

Ephone-hunt configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS 12.4(9)T.

Usage Guidelines

This command creates a label to identify the ephone-hunt group. This label helps make the configuration more readable.

Examples

The following example shows how to identify a hunt group for technical support agents.

```
ephone-hunt 3 peer
 pilot 4200
 list 1001, 1002, 1003
 description Tech Support Hunt Group
 hops 3
 timeout 7, 10, 15
 max-timeout 25
 final 4500
```

Related Commands

Command	Description
ephone-hunt	Defines an ephone hunt group and enters ephone-hunt configuration mode.

description (voice register pool)

To display a custom description in the header bar of Cisco IP Phone 7940 and 7940G or a Cisco IP Phone 7960 and 7960G, use the **description** command in voice register pool configuration mode. To return to the default, use the **no** form of this command.

description *string*

no description

Syntax Description

string	Alphanumeric characters that appear in the header bar of the phone display. If spaces appear in the string, enclose the string in quotation marks. The maximum string length is 40 characters, but the string is truncated to 14 characters in the display of Cisco IP Phone 7940s, Cisco IP Phone 7940Gs, Cisco IP Phone 7960s, and Cisco IP Phone 7960Gs.

Defaults

The extension number of the first line on the phone appears in the header bar.

Command Modes

Voice register pool configuration

Command History

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

Usage Guidelines

Use this command to display a customized description in the header bar of a SIP phone instead of the extension number, which is the default. For example, you can display the entire E.164 telephone number associated with the first phone line.

Examples

The following example shows how to define a header bar display for a SIP phone on which the extension number is 50155:

```
Router(config)# voice register pool 4
Router(config-register-pool)# number 1 50155
Router(config-register-pool)# description 888-555-0155
```

Related Commands

Command	Description
voice register pool	Enters voice register pool configuration mode for SIP phones.
number (voice register pool)	Configures a valid number for a SIP phone.

device-security-mode

To set the security mode for SCCP signaling for devices communicating with the Cisco Unified CME router globally or per ephone, use the **device-security-mode** command in telephony-service or ephone configuration mode. To return to the default, use the **no** form of this command.

device-security-mode {**authenticated** | **none** | **encrypted**}

no device-security-mode

Syntax Description

authenticated	SCCP signaling between a device and Cisco Unified CME through the secure TLS connection on TCP port 2443.
none	SCCP signaling is not secure.
encrypted	SCCP signaling between a device and Cisco Unified CME through the secure TLS connection on TCP port 2443, and the media uses Secure Real-Time Transport Protocol (SRTP).

Command Default

Device signaling is not secure.

Command Modes

Telephony-service configuration
Ephone configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.
12.4(11)XW1	Cisco Unified CME 4.1	The **encrypted** keyword was added.
12.4(15)T	Cisco Unified CME 4.1	The **encrypted** keyword was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

Use this command with Cisco Unified CME phone authentication and encryption.

Set the SCCP signaling security mode globally using this command in telephony-service configuration mode or per ephone using this command in ephone configuration mode. If you use both commands, the per-phone setting overrides the global setting.

Examples

The following example selects secure SCCP signaling for all ephones.

```
Router(config)# telephony-service
Router(config-telephony)# device-security-mode authenticated
```

The following example selects secure SCCP signaling for ephone 28:

```
Router(config)# ephone 28
```

```
Router(config-ephone)# button 1:14 2:25
Router(config-ephone)# device-security-mode authenticated
```

The following example selects secure SCCP signaling for all ephones and then disables it for ephone 36:

```
Router(config)# telephony-service
Router(config-telephony)# device-security-mode authentication

Router(config)# ephone 36
Router(config-ephone)# button 1:15 2:16
Router(config-ephone)# device-security-mode none
```

The following example selects encrypted secure SCCP signaling and encryption through SRTP for all ephones:

```
Router(config)# telephony-service
Router(config-telephony)# device-security-mode encrypted
```

dialplan

To assign a dial plan to a SIP phone, use the **dialplan** command in voice register pool or voice register template configuration mode. To remove the dial plan from the phone, use the **no** form of this command.

dialplan *dialplan-tag*

no dialplan *dialplan-tag*

Syntax Description

dialplan-tag	Number that identifies the dial plan to use for this SIP phone. This is the *dialplan-tag* argument that was assigned to the dial plan with the **voice register dialplan** command. Range: 1 to 24.

Command Default

No dial plan is assigned to the phone.

Command Modes

Voice register pool configuration
Voice register template configuration

Command History

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

Usage Guidelines

You apply a dial plan to a SIP phone with this command after you create the dial plan with the **voice register dialplan** command. When the phone is reset or restarted, the dial plan file specified with this command is loaded to the phone. A phone can use only one dial plan.

A dial plan assigned to a SIP phone has priority over Key Press Markup Language (KPML), which is enabled by default on the phone.

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

After using the **no dialplan** command to remove a dial plan from a phone, use the **restart** command after creating a new configuration profile if the dial plan was defined with the **pattern** command. If the dial plan was defined using a custom XML file with the **filename** command, you must use the **reset** command for the change to take effect.

Examples

The following example shows that dial plan 5 is assigned to the SIP phone identified by pool 1:

```
Router(config)# voice register pool 1
Router(config-register-pool)# dialplan 5
```

The following example shows that dial plan 5 is assigned to voice register template 10:

```
Router(config)# voice register template 10
Router(config-register-temp)# dialplan 5
```

Related Commands

Command	Description
digit collect kpml	Enables KPML digit collection on a SIP phone.
filename	Specifies a custom XML file that contains the dial patterns to use for a SIP dial plan.
pattern	Defines a dial pattern for a SIP dial plan.
show voice register dialplan	Displays all configuration information for a specific SIP dial plan.
show voice register pool	Displays all configuration information associated with a particular voice register pool.
voice register dialplan	Enters voice register dialplan configuration mode to define a dial plan for SIP phones.

dialplan-pattern

To define a pattern that is used to expand extension numbers in Cisco Unified CME into fully qualified E.164 numbers, use the **dialplan-pattern** command in telephony-service configuration mode. To disable the **dialplan-pattern** command settings, use the **no** form of this command.

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

no dialplan-pattern *tag*

Syntax Description

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

Defaults

No expansion pattern exists.

Command Modes

Telephony-service configuration

Command History

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

Cisco IOS Release	Cisco Product	Modification
12.2(8)T1	Cisco ITS 2.0	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
12.2(11)T	Cisco ITS 2.01	This command was implemented on the Cisco 1760.
12.2(11)YT	Cisco ITS 2.1	The **extension-pattern** keyword was added.
12.2(15)T	Cisco ITS 2.1	This command was integrated into Cisco IOS Release 12.2(15)T.

Usage Guidelines

This command creates a pattern for expanding individual abbreviated extension numbers of calling numbers into fully qualified E.164 numbers.

Use this command when configuring a network with multiple Cisco Unified CMEs to ensure that the appropriate calling number, extension or E.164 number, is provided to the target Cisco Unified CME, and appears on the phone display of the called phone. In networks that have a single Cisco Unified CME, this command is not needed.

If multiple dial-plan patterns are defined, the system matches extension numbers against the patterns in sequential order, starting with with the lowest numbered dial-plan pattern tag first. Once a pattern matches an extension number, the pattern is used to generate an expanded number. If additional patterns subsequently match the extension number, they are not used.

The **dialplan-pattern** command builds additional dial peers for the expanded numbers it creates. For example, when the ephone-dn with the number 1001 was defined, the following POTS dial peer was automatically created for it:

```
dial-peer voice 20001 pots
 destination-pattern 1001
 voice-port 50/0/2
```

When you define a dial-plan pattern that 1001 will match, such as 40855510.., a second dial peer is created so that calls to both the 1001 and 4085551001 numbers will be completed. In our example, the additional dial peer that is automatically created looks like the following:

```
dial-peer voice 20002 pots
 destination-pattern 4085551001
 voice-port 50/0/2
```

Both numbers are recognized by Cisco Unified CME as being associated with a SCCP phone.

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

In networks with multiple routers, you may need to use the **dialplan-pattern** command to expand extensions to E.164 numbers because local extension numbering schemes can overlap each other. Networks with multiple routers have authorities such as gatekeepers that route calls through the network. These authorities require E.164 numbers so that all numbers in the network will be unique. Use the **dialplan-pattern** command to expand extension numbers into unique E.164 numbers for registering with a gatekeeper.

Ephone-dn numbers for the Cisco IP phones must match the number in the *extension-length* argument; otherwise, the extension number cannot be expanded. For example, the following command maps all 3-digit extension numbers to the telephone number 40855501xx, so that extension 111 is expanded but the 4-digit extension 1011 is not.

```
dialplan-pattern 1 40855501.. extension-length 3
```

Using the **dialplan-pattern** command to expand extension numbers can sometimes result in the improper matching of numbers with dial peers. For example, the expanded E.164 number 2035550134 can match dial-peer destination-pattern 203, not 134, which would be the correct destination pattern for the desired extension. If it is necessary for you to use the **dialplan-pattern** command and you know that the expanded numbers might match destination patterns for other dial peers, you can manually configure the E.164 expanded number for an extension as its secondary number using the **number** command, as shown in the following example.

```
ephone-dn 23
 number 134 secondary 2035550134
```

The pattern created by the **dialplan-pattern** command is also used to enable distinctive ringing for inbound calls. If a calling-party number matches a dial-plan pattern, the call is considered an internal call and has a distinctive ring that identifies the call as internal. Any call with a calling-party number that does not match a dial-plan pattern is considered an external call and has a distinctive ring that is different from the internal ringing.

When the **extension-pattern** keyword and *extension-pattern* argument are used, the leading digits of an extension pattern are stripped and replaced with the corresponding leading digits of the dial plan. For example, the following command maps all 4xx extension numbers to the E.164 number 40855501xx, so that extension 412 corresponds to 4085550112.

```
dialplan-pattern 1 4085550100 extension-length 3 extension-pattern 4..
```

Examples

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

```
Router(config)# telephony-service
Router(config-telephony)# dialplan-pattern 1 40855550.. extension-length 4
extension-pattern 50..
```

In the following example, the **dialplan-pattern** command creates dial-plan pattern 1 for extensions 800 to 899 with the telephone prefix starting with 4085559. As each number in the extension pattern is declared with the **number** command, two POTS dial peers are created. In the example, they are 801 (an internal office number) and 4085579001 (an external number).

```
Router(config)# telephony-service
Router(config-telephony)# dialplan-pattern 1 40855590.. extension-length 3
extension-pattern 8..
```

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

```
Router(config)# telephony-service
Router(config-telephony)# dialplan-pattern 1 40855550.. extension-length 4
extension-pattern 50..
Router(config-telephony)# dialplan-pattern 2 51055560.. extension-length 4
extension-pattern 60..
```

Related Commands

Command	Description
show telephony-service dial-peer	Displays dial peer information for extensions in a Cisco CME system.

dialplan-pattern (voice register)

To define a pattern that is used to expand extension numbers in Cisco Unified CME into fully qualified E.164 numbers, use the **dialplan-pattern** command in voice register global configuration mode. To disable the **dialplan-pattern** command settings, use the **no** form of this command.

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

no dialplan-pattern *tag*

Syntax Description

tag	Unique number for identifying this dial-plan pattern. Range: 1 to 5.
pattern	Dial-plan pattern to be matched, such as the area code, the prefix, and the first one or two digits of the extension number, plus wildcard markers or dots (.) for the remainder of the extension number digits.
extension-length	Number of extension digits that will appear as a caller ID.
extension-length	Number of digits in an extension. This variable must match the length of the directory numbers configured for SIP extensions in Cisco Unified CME. Range: 1 to 32.
extension-pattern	(Optional) Leading digit pattern to be configured for an extension when it is different from the leading digit pattern of the E.164 telephone number, as defined in the *extension-pattern* argument.
extension-pattern	(Optional) Leading digit pattern to be stripped from extension number when expanding an extension to an E.164 telephone number. Consists of one or more digits and wildcard markers or dots (.). For example, 5.. would include extension 500 to 599, and 5... would include 5000 to 5999. The length of the extension pattern must equal the value configured for the *extension-length* argument.
no-reg	(Optional) Prevents the E.164 numbers in the dial peer from registering with the gatekeeper.

Command Default

No expansion pattern exists.

Command Modes

Voice register global configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS 12.4(9)T.

Usage Guidelines

This command creates a pattern for expanding individual abbreviated SIP extension numbers of calling numbers into fully qualified E.164 numbers.

Use this command when configuring a network with multiple Cisco Unified CMEs to ensure that the appropriate calling number, extension or E.164 number, is provided to the target Cisco Unified CME, and appears on the phone display of the called phone. In networks that have a single Cisco Unified CME, this command is not needed.

Up to five dial-plan patterns can be configured. If multiple dial-plan patterns are defined, the system matches extension numbers against the patterns in sequential order, starting with the lowest numbered dial-plan pattern tag first.

Dial peers for directory numbers are automatically created when SIP phones register in Cisco Unified CME. The **dialplan-pattern** command builds a second dial peer for the expanded number because an extension number matches the pattern. Both numbers are recognized by Cisco Unified CME as being associated with a SIP phone.

For example, the following POTS dial peer is automatically created for extension number 1001 when the associated SIP phone registers in Cisco Unified CME:

```
dial-peer voice 20001 pots
 destination-pattern 1001
 voice-port 50/0/2
```

If the extension number (1001) also matches a dial-plan pattern that is configured using the **dialplan-pattern** command, such as 40855510.., a second dial peer is dynamically created so that calls to both the 1001 and 4085551001 numbers can be completed. Based on the dial-plan pattern to be matched, the following additional POTS dial peer is created:

```
dial-peer voice 20002 pots
 destination-pattern 4085551001
 voice-port 50/0/2
```

Using the **no** form of this command will remove the dial peer that was created for the expanded number.

All dial peers can be displayed by using the **show dial-peer voice summary** command. All dial peers for numbers associated to SIP phones only can be displayed by using the **show voice register dial-peers** command. Dial peers created by using the **dialplan-expansion** command cannot be seen in the running configuration.

The value of the extension-length argument must be equal to the length of extension number to be matched, otherwise, the extension number cannot be expanded. For example, the following command maps all 3-digit extension numbers to the telephone number 40855501.., so that extension 111 is expanded but 4-digit extension number 1111 is not.

```
dialplan-pattern 1 40855501.. extension-length 3
```

When the **extension-pattern** keyword and *extension-pattern* argument are configured, the leading digits of the extension pattern variable are stripped away and replaced with the corresponding leading digits of the dial-plan pattern to create the expanded number. For example, the following command maps all 3-digit extension numbers with the leading digit of “4” to the telephone number 40855501.., so that extension 434 corresponds to 4085550134.

```
dialplan-pattern 1 40855501.. extension-length 3 extension-pattern 4..
```

To apply dialplan-pattern expansion on a per-system basis to individual SIP *redirecting* numbers in a Cisco Unified CME system, including original called and last reroute numbers, use the **call-forward** command.

Examples

The following example shows how to create a dialplan-pattern for expanding extension numbers 60xxx to E.164 numbers 5105555xxx.

```
Router(config)# voice register global
Router(config-register-global)# dialplan-pattern 1 5105550... extension-length 5
```

The following example is output from the **show dial-peer summary** command displaying information for four dial peers, one each for extensions 60001 and 60002 and because the dialplan-expansion command was configured to expand 6.... to 4085555...., one each for 4085550001 and 4085550002. The latter two dial peers will not appear in the running configuration.

```
Router# show dial-peer summary
             AD                                    PRE PASS                OUT
TAG    TYPE  MIN  OPER PREFIX    DEST-PATTERN      FER THRU SESS-TARGET    STATT
20010  pots  up   up             60002$             0                          0
20011  pots  up   up             60001$             0                          9
20012  pots  up   up             5105555001$        0                          9
20013  pots  up   up             5105555002$        0                          0
```

Related Commands

Command	Description
call-forward (voice register)	Applies dial-plan pattern expansion globally to redirecting number.
show dial-peer summary	Displays all dial peers created in Cisco Unified CME.
show voice register dial-peer	Displays dial-peer information for SIP extensions in Cisco Unified CME.

digit collect kpml

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

digit collect kpml

no digit collect kpml

Syntax Description

This command has no arguments or keywords.

Command Default

KPML digit collection is enabled.

Command Modes

Voice register pool configuration
Voice register template configuration

Command History

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

Usage Guidelines

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

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

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

Examples

The following example shows KPML enabled on SIP phone 4:

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

Related Commands

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

directory

To define the order in which the names of Cisco IP phone users are displayed in the local directory, use the **directory** command in telephony-service configuration mode. To return to the default, use the **no** form of this command.

directory {**first-name-first** | **last-name-first**}

no directory {**first-name-first** | **last-name-first**}

Syntax Description

first-name-first	First name is entered first in the Cisco IP phone directory name field.
last-name-first	Last name is entered first in the Cisco IP phone directory name field.

Defaults

first-name-first

Command Modes

Telephone configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(2)XT	Cisco ITS 2.0	This command was introduced on the Cisco 1750, Cisco 1751, Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 2600-XM, Cisco 2691, Cisco 3725, and Cisco 3745.
12.2(11)T	Cisco ITS 2.01	This command was implemented on the Cisco 1760.

Usage Guidelines

This command defines name order in the local directory. The directory itself is generated from entries made using the **name** command and the **number** command in ephone-dn configuration mode.

Note The name information must be entered in the correct order in the **name** command.

The location for the file that is accessed when the Directories button is pressed is specified in the **url** (telephony-service) command.

Examples

The following example shows how to configure the local directory with the last name first:

```
Router(config)# telephony-service
Router(config-telephony)# directory last-name-first
```

Related Commands

Command	Description
name	Specifies a name to be associated with an extension (ephone-dn).
number	Specifies a telephone number to be associated with an extension (ephone-dn).
telephony-service	Enters telephony-service configuration mode.
url	Provisions URLs for the displays associated with buttons on Cisco IP phones.

directory entry

To add a systemwide phone directory and speed-dial definition, use the **directory entry** command in telephony-service configuration mode. To remove a definition, use the **no** form of this command.

directory entry {*directory-tag number* **name** *name* | **clear**}

no directory entry {*directory-tag* | **clear**}

Syntax Description

directory-tag	Digit string that provides a unique identifier for this entry. Range is from 1 to 99.
number	String of up to 32 digits that provides the full telephone number for this entry.
name *name*	String of up to 24 characters that provides a name for this entry.
clear	Removes all directory entries that were made with this command.

Defaults

Entries do not exist.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.3(11)XL	Cisco CME 3.2.1	This feature was modified to enable systemwide speed-dialing of entries from 34 to 99.
12.3(14)T	Cisco CME 3.3	This command was integrated into Cisco IOS Release 12.3(14)T.

Usage Guidelines

The Cisco CallManager Express system automatically creates a local phone directory consisting of the telephone numbers and names that are entered during ephone-dn configuration. Additional directory entries can be made by administrators using the **directory entry** command. Phone number directory listings are displayed in the order in which they are entered.

A single entry can be removed using the **no directory entry** ***directory-tag*** command.

Directory entries that have directory-tag numbers from 34 to 99 also can be used as systemwide speed-dial numbers. That is, if you have the following definition for the headquarters office, any phone user can speed-dial the number:

```
Router(config)# telephony-service
Router(config-telephony)# directory entry 51 4085550123 name Headquarters
```

Analog phone users press the asterisk (*) key and the speed-dial identifier (tag number) to dial a speed-dial number.

IP phone users follow this procedure to dial a speed-dial number:

1. With the phone on-hook, an IP phone user presses a two-digit speed-dial code (that is, 05 for the entry with tag 5). A new soft key, Abbr, appears in the phone display.
2. The phone user picks up the phone handset and presses the Abbr soft key. The full telephone number associated with the speed-dial tag is dialed.

Examples

The following example adds six telephone listings to the local directory. The last two entries, with the identifiers 50 and 51, can be speed-dialed by anyone on the system because their identifiers (directory-tags) are between 34 and 99.

```
Router(config)# telephony-service
Router(config-telephony)# directory entry 1 4045550110 name Atlanta
Router(config-telephony)# directory entry 2 3125550120 name Chicago
Router(config-telephony)# directory entry 4 2125550140 name New York City
Router(config-telephony)# directory entry 5 2065550150 name Seattle
Router(config-telephony)# directory entry 50 4085550123 name Corp Headquarters
Router(config-telephony)# directory entry 51 4085550145 name Division Headquarters
```

Related Commands

Command	Description
telephony-service	Enters telephony-service configuration mode.

display-logout

To specify a message to display on phones in an ephone hunt group when all phones in the hunt group are logged out, use the **display-logout** command in ephone-hunt configuration mode. To return this value to the default, use the **no** form of this command.

display-logout *string*

no display-logout

Syntax Description

string	Character string to be displayed on hunt group member IP phones when all members are logged out.

Command Default

No logout message exists.

Command Modes

Ephone-hunt configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command defines a plain-text message that displays on phones with ephone-dns that are members of a hunt group when all the members of the group are logged out. The message can be used to notify agents that no agents are available to take hunt group calls. It can also be used to tell agents about the disposition of any incoming calls to the hunt group when no agents are available to answer calls. For example, you could set the display to read “All Agents Unavailable,” or “Hunt Group Voice Mail” or “Hunt Group Night Service.”

Examples

The following example specifies a message to display when all agents are logged out of hunt group 3.

```
ephone-hunt 3 peer
 pilot 4200
 list 1001, 1002, 1003
 display-logout All Agents Logged Out
 hops 3
 timeout 7, 10, 15
 max-timeout 25
 final 4500
```

Related Commands

Command	Description
ephone-hunt	Defines an ephone hunt group and enters ephone-hunt configuration mode.

dnd (voice register pool)

To enable the Do-Not-Disturb (DND) feature, use the **dnd-control** command in voice register pool configuration mode. To disable the DND, use the **no** form of this command.

dnd

no dnd

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled

Command Modes

Voice register pool configuration

Command History

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

Examples

The following example shows how to enable DND:

```
Router(config)# voice register pool 1
Router(config-register-pool)# dnd
```

Related Commands

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

dnd feature-ring

To allow phone buttons configured with the feature-ring option to not ring when their phones are in do-not-disturb (DND) mode, use the **dnd feature-ring** command in ephone configuration mode. To allow lines configured for feature ring to ring when the phone is in DND mode, use the **no** form of this command.

dnd feature-ring

no dnd feature-ring

Syntax Description

This command has no arguments or keywords.

Defaults

When incoming calls occur, all of the buttons configured on IP phones in DND mode will not ring.

Command Modes

Ephone configuration mode

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)XL	Cisco CME 3.2.1	This command was introduced.
12.3(14)T	Cisco CME 3.3	This command was integrated into Cisco IOS Release 12.3(14)T.

Usage Guidelines

This command applies only to ephones that are configured with the feature-ring option provided by the **button** command.

Note that the affirmative form of the command is enabled by default and feature-ring lines will not ring when the phone is in DND mode. To enable feature-ring lines to ring when the phone is in DND mode, use the **no dnd feature-ring** command.

Examples

For the following example, when DND is active on ephone 1 and ephone 2, button 1 will ring, but button 2 will not.

```
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 1001

Router(config)# ephone-dn 2
Router(config-ephone-dn)# number 1002

Router(config)# ephone-dn 10
Router(config-ephone)# number 1110
Router(config-ephone)# preference 0
Router(config-ephone)# no huntstop

Router(config)# ephone-dn 11
Router(config-ephone)# number 1111
Router(config-ephone)# preference 1
Router(config-ephone)# no huntstop
```

```
Router(config)# ephone 1
Router(config-ephone)# button 1f1
Router(config-ephone)# button 2o10,11
Router(config-ephone)# no dnd feature-ring

Router(config-ephone-dn)# ephone 2
Router(config-ephone)# button 1f2
Router(config-ephone)# button 2o10,11
Router(config-ephone)# no dnd feature-ring
```

Related Commands

Command	Description
button	Associates ephone-dns with individual buttons on a Cisco IP phone and specifies ring behavior.
ephone	Enters ephone configuration mode for an IP phone for the purposes of creating and configuring an ephone.

dnd-control (voice register template)

To enable the Do-Not-Disturb (DND) soft key on SIP phones, use the **dnd-control** command in voice register template configuration mode. To disable the DND soft key on a SIP phone, use the **no** form of this command.

dnd-control

no dnd-control

Syntax Description

This command has no arguments or keywords.

Defaults

Enabled

Command Modes

Voice register template configuration

Command History

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

Usage Guidelines

This command enables a soft key for Do-Not-Disturb (DND) in the specified template which can then be applied to SIP phones. The DND soft key is enabled by default. To disable the DND soft key, use the **dnd** command. To apply a template to a SIP phone, use the template command in voice register pool configuration mode.

Examples

The following example shows how to disable the DND soft key:

```
Router(config)# voice register template 1
Router(config-register-template)# dnd-control
```

Related Commands\

Command	Description
voice register template	Enter voice register template configuration mode.

dn-webedit

To enable the adding of extensions (ephone-dns) through the Cisco CallManager Express (Cisco CME) graphical user interface (GUI), use the **dn-webedit** command in telephony-service configuration mode. To disable this feature, use the **no** form of this command.

dn-webedit

no dn-webedit

Syntax Description

This command has no arguments or keywords.

Defaults

Extensions cannot be added through the Cisco CME GUI.

Command Modes

Telephony-service configuration

Command History

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

Usage Guidelines

The **dn-webedit** command enables the adding of extensions through the web-based GUI. If the **dn-webedit** command is enabled, a customer administrator or a system administrator can modify and assign extensions associated with the Cisco CME router. If this ability is disabled, extensions must be added using the router command-line interface (CLI).

If the set of extension numbers used by the router is part of a larger telephone network, limitations on modification might be needed to ensure network integrity. Disabling the **dn-webedit** command prevents an administrator from allocating phone numbers and prevents assignment of numbers that may already be used elsewhere in the network.

Examples

The following example enables editing of directory numbers through the web-based GUI interface:

```
Router(config)# telephony-service
Router(config-telephony)# dn-webedit
```

Related Commands

Command	Description
telephony-service	Enters telephony-service configuration mode.
time-webedit	Enables time setting through the web interface.

dst (voice register global)

To set the time period for daylight saving time on SIP phones, use the **dst** command in voice register global configuration mode. To disable daylight saving time, use the **no** form of this command.

dst {**start** | **stop**} *month* [**day** *day-of-month* | **week** *week-number* **day** *day-of-week*] **time** *hour:minutes*}

no dst {**start** | **stop**}

Syntax Description

start	Sets beginning time for daylight saving time.
stop	Sets ending time for daylight saving time.
month	Abbreviated month. The following abbreviations are valid: **jan**, **feb**, **mar**, **apr**, **may**, **jun**, **jul**, **aug**, **sep**, **oct**, **nov**, **dec**.
day *day-of-month*	Date of the month. Range is 1 to 31.
week *week-number*	Number identifying the week of the month. Range is 1 to 4, or 8, where 8 represents the last week of the month.
day *day-of-week*	Abbreviated day of the week. The following abbreviations are valid: **sun**, **mon**, **tue**, **wed**, **thu**, **fri**, **sat**.
time *hour:minutes*	Beginning and ending time for daylight saving time, in HH:MM format using a 24-hour clock. The stop time must be greater than the start time. The value 24:00 is not valid. If you enter 00:00for both start time and stop time, daylight saving time is enabled for the entire 24-hour period on the specified date.

Defaults

Start: First week of April, Sunday, 2:00 a.m.
Stop: Last week of October, Sunday 2:00 a.m.

Command Modes

Voice register global configuration

Command History

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

Usage Guidelines

This command sets the stop and start times for daylight saving time if the **dst auto-adjust** command is configured.

Examples

The following example shows how to set automatic adjustment of daylight saving time:

```
Router(config)# voice register global
Router(config-register-global)# dst start Jan day 1 time 00:00
Router(config-register-global)# dst stop Mar day 31 time 23:99
```

Related Commands

Command	Description
date-format (voice register global)	Sets the date display format on SIP phones in a Cisco CME system.
dst auto-adjust (voice register global)	Enables automatic adjustment of daylight saving time on SIP phones.
time-format (voice register global)	Selects a 12-hour clock or a 24-hour clock for the time display format on Cisco IP phones in a SIP CME system.
timezone (voice register global)	Sets the time zone used for SIP phones in a Cisco CME system.
voice register global	Enters voice register global configuration mode in order to set global parameters for all supported Cisco SIP phones in a Cisco CME or Cisco SIP SRST environment.

dst auto-adjust (voice register global)

To enable automatic adjustment of daylight saving time on SIP phones, use the **dst auto-adjust** command in voice register global configuration mode. To disable daylight saving time auto adjustment, use the **no** form of this command.

dst auto-adjust

no dst auto-adjust

Syntax Description

This command has no arguments or keywords.

Defaults

Enabled

Command Modes

Voice register global configuration

Command History

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

Usage Guidelines

Automatic adjustment for daylight saving time is enabled by default. To disable auto adjusting for DST, use the **no dst auto-adjust** command. To set the start and stop times for DST, use the **dst** command.

Examples

The following example shows how to disable the automatic adjustment for daylight saving time:

```
Router(config)# voice register global
Router(config-register-global)# no dst auto-adjust
```

Related Commands

Command	Description
date-format (voice register global)	Sets the date display format on SIP phones in a Cisco CME system.
dst (voice register global)	Sets the start and stop time if using daylight saving time on SIP phones.
time-format (voice register global)	Selects a 12-hour clock or a 24-hour clock for the time display format on Cisco IP phones in a SIP CME system.
timezone (voice register global)	Sets the time zone used for SIP phones in a Cisco CME system.
voice register global	Enters voice register global configuration mode in order to set global parameters for all supported Cisco SIP phones in a Cisco CME or Cisco SIP SRST environment.

dtmf-relay (voice register pool)

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

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

no dtmf-relay

Syntax Description

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

Command Default

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

Command Modes

Voice register pool configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(4)T	Cisco SIP SRST 3.0	This command was introduced.
12.4(4)T	Cisco CME 3.4 Cisco SIP SRST 3.4	This command was added to Cisco Unified CME.

Usage Guidelines

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

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

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

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

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

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

Note

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

Examples

Cisco Unified CME

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

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

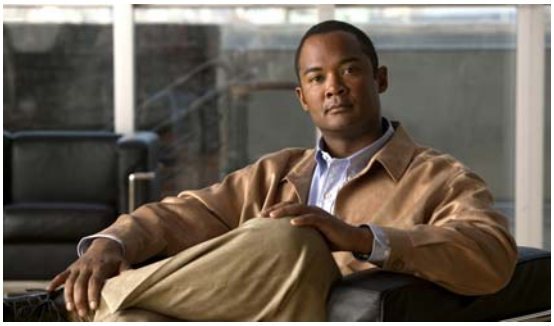
Cisco Unified SIP SRST

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

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

Related Commands

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



CiscoUnified CME Commands: Debug

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This chapter contains commands to configure and maintain Cisco Unified Communications Manager Express (formally known as Cisco Unified CallManager Express). The commands are presented in alphabetical order. Some commands required for configuring Cisco Unified Communications Manager Express (Cisco Unified CME) may be found in other Cisco IOS command references. Use the command reference master index or search online to find these commands.

debug callmonitor

To collect and display debugging traces for call monitor, use the **debug callmonitor** command in privileged EXEC mode. To disable debugging, use the **no** form of this command.

debug callmonitor {**all** | **core** | **detail** | **errors** | **events** | **hwconf** | **info** | **xml**}

no debug command {**all** | **core** | **detail** | **errors** | **events** | **hwconf** | **info** | **xml**}

Syntax Description

all	All call-monitor debugging traces.
core	Core information debugging traces.
detail	Detailed debugging traces.
errors	Call-monitor error debugging traces.
events	Call-monitor event debugging traces.
hwconf	Debugging traces related to hardware configuration.
info	Call-monitor information debugging traces.
xml	Call-monitor XML encoding debugging traces.

Command Default

There is no default for this command.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.4(11)XW1	This command was introduced.

Examples

The following example is partial output from this command:

```
Syslog logging: enabled (11 messages dropped, 2 messages rate-limited,
                0 flushes, 0 overruns, xml disabled, filtering disabled)

No Active Message Discriminator.



No Inactive Message Discriminator.


    Console logging: disabled
    Monitor logging: level debugging, 0 messages logged, xml disabled,
                     filtering disabled
    Buffer logging:  level debugging, 444378 messages logged, xml disabled,
                     filtering disabled
    Logging Exception size (4096 bytes)
    Count and timestamp logging messages: disabled
    Persistent logging: disabled
    Trap logging: level informational, 461 message lines logged
```

```
Log Buffer (1000000 bytes):

Jun  4 22:30:24.222: //CMM/INFO:
Jun  4 22:30:24.222: //CMM/INFO:
Jun  4 22:30:24.222: //CMM/INFO:cmm_notify_trigger() 15, callID 99685, 5114016,
1884814040, 1632257208
Jun  4 22:30:24.222: //CMM/INFO:    target_node 0
Jun  4 22:30:24.222: //CMM/INFO:Lineinfo node Search FAILED
Jun  4 22:30:24.222: //CMM/INFO:create_lineinfo_node
Jun  4 22:30:24.222: //CMM/INFO:    target_node 66AF3714
Jun  4 22:30:24.222: //CMM/INFO:   - dn 4016

Jun  4 22:30:24.222: //CMM/INFO:  CallEntry 709C3FB8
Jun  4 22:30:24.222: //CMM/INFO:  dstCallID -1
Jun  4 22:30:24.222: //CMM/INFO:  line_info 66AF3720, dn 4016
Jun  4 22:30:24.222: //CMM/INFO:    * cmm_crs_proc_tr_rpt_orig
Jun  4 22:30:24.222: //CMM/INFO:    callID = 99685, CG 5114016, GCID
=05591A85-122211DC-8645A1CA-4B604A7A
Jun  4 22:30:24.222: //CMM/INFO:increase_gcid_ref_count 99685 0
Jun  4 22:30:24.222: //CMM/INFO:find_gcidinfo_node
Jun  4 22:30:24.222: //CMM/INFO:      target_node 0
Jun  4 22:30:24.222: //CMM/INFO:      Gcidinfo node Search FAILED
Jun  4 22:30:24.222: //CMM/INFO:create_gcidinfo_node
Jun  4 22:30:24.222: //CMM/INFO:      target_node 6544A9CC
Jun  4 22:30:24.222: //CMM/INFO:   - gcid 05591A85-122211DC-8645A1CA-4B604A7A
Jun  4 22:30:24.222: //CMM/INFO:      count = 1
Jun  4 22:30:24.222: //CMM/INFO:insert_ssptrs_to_gcid for line_info 66AF3720 (dn 4016),
GCID 05591A85-122211DC-8645A1CA-4B604A7A
Jun  4 22:30:24.222:     ss_ptr list :-
Jun  4 22:30:24.222:     ss_ptr list :-
Jun  4 22:30:24.222: //CMM/INFO:
Jun  4 22:30:24.222: //CMM/INFO:
Jun  4 22:30:24.222: //CMM/INFO:cmm_notify_trigger() 1, callID 99685, 5114016, 16,
1695547392
Jun  4 22:30:24.222: //CMM/INFO:    target_node 66AF3714
Jun  4 22:30:24.222: //CMM/INFO:   - dn 4016

Jun  4 22:30:24.222: //CMM/INFO:  CallEntry 709C3FB8
Jun  4 22:30:24.222: //CMM/INFO:  dstCallID -1
Jun  4 22:30:24.222: //CMM/INFO:  line_info 66AF3720, dn 4016
Jun  4 22:30:24.222: //CMM/INFO:    * cmm_crs_proc_tr_call_orig
Jun  4 22:30:24.222: //CMM/INFO:        orig --> callID 99685, line_info 66AF3720,
call_inst 655AF384, gcid 05591A85-122211DC-8645A1CA-4B604A7A
Jun  4 22:30:24.222: //CMM/INFO:is_sccp_endpoint DN 4016
Jun  4 22:30:24.222: //CMM/INFO:
Jun  4 22:30:24.222:     sccp endpoint TRUE
Jun  4 22:30:24.222: //CMM/INFO:find_gcidinfo_node
Jun  4 22:30:24.222: //CMM/INFO:      target_node 6544A9CC
Jun  4 22:30:24.222: //CMM/INFO:   - gcid 05591A85-122211DC-8645A1CA-4B604A7A
Jun  4 22:30:24.222: //CMM/INFO:cmm_send_dialog_notify  sub_info 0
Jun  4 22:30:24.222:     ss_ptr list :-
Jun  4 22:30:24.222: //CMM/INFO:        <== DIALOG MGR ==>
Jun  4 22:30:24.222: //CMM/INFO:            :: CMM_EV_CALL_CONN_ORIGINATED
Jun  4 22:30:24.222: //CMM/INFO:                 - Gcid
05591A85-122211DC-8645A1CA-4B604A7A
Jun  4 22:30:24.222: //CMM/INFO:                 - Calling 4016
Jun  4 22:30:24.222: //CMM/INFO:                 - Called
Jun  4 22:30:24.222: //CMM/INFO:                 - ConnAddr 4016
Jun  4 22:30:24.222: //CMM/INFO:                 - Type 0
Jun  4 22:30:24.222: //CMM/INFO:                 - parentGcid
00000000-00000000-00000000-00000000
Jun  4 22:30:24.222: //CMM/INFO:find_gcidinfo_node
Jun  4 22:30:24.222: //CMM/INFO:      target_node 6544A9CC
Jun  4 22:30:24.222: //CMM/INFO:   - gcid 05591A85-122211DC-8645A1CA-4B604A7A
```

```
Jun  4 22:30:24.222: //CMM/DETAIL: type: CMM_EV_CALL_CONN_ORIGINATED, filter analyzing....
[4016, , 4016]
Jun  4 22:30:24.222: //CMM/INFO:find_gcidinfo_node
Jun  4 22:30:24.222: //CMM/INFO:      target_node 6544A9CC
Jun  4 22:30:24.222: //CMM/INFO:   - gcid 05591A85-122211DC-8645A1CA-4B604A7A
Jun  4 22:30:24.222: //CMM/DETAIL:gcid is not part of conference. [4016, , 4016] checking
originateFilter...
Jun  4 22:30:24.222: //CMM/DETAIL:originateFilter[callid=99685, pdn=16, pchan=1] is not
set. [4016, , 4016] is  not filtered
Jun  4 22:30:24.222: //CMM/INFO:find_gcidinfo_node
Jun  4 22:30:24.222: //CMM/INFO:      target_node 6544A9CC
Jun  4 22:30:24.222: //CMM/INFO:   - gcid 05591A85-122211DC-8645A1CA-4B604A7A
Jun  4 22:30:24.222: //CMM/INFO:cmm_send_dialog_notify  sub_info 0
Jun  4 22:30:24.222:     ss_ptr list :-
Jun  4 22:30:24.222: //CMM/INFO:        <== DIALOG MGR ==>
Jun  4 22:30:24.222: //CMM/INFO:            :: CMM_EV_CALL_CONN_ACTIVE
Jun  4 22:30:24.222: //CMM/INFO:                 - Gcid
05591A85-122211DC-8645A1CA-4B604A7A
Jun  4 22:30:24.222: //CMM/INFO:                 - Calling 4016
Jun  4 22:30:24.222: //CMM/INFO:                 - Called
Jun  4 22:30:24.222: //CMM/INFO:                 - ConnAddr 4016
Jun  4 22:30:24.222: //CMM/INFO:                 - LastRedirectAddr
Jun  4 22:30:24.222: //CMM/INFO:                 - Type 0
Jun  4 22:30:24.222: //CMM/INFO:                 - parentGcid
00000000-00000000-00000000-00000000
Jun  4 22:30:24.222: //CMM/INFO:find_gcidinfo_node
Jun  4 22:30:24.222: //CMM/INFO:      target_node 6544A9CC
Jun  4 22:30:24.222: //CMM/INFO:   - gcid 05591A85-122211DC-8645A1CA-4B604A7A
Jun  4 22:30:24.222: //CMM/DETAIL: type: CMM_EV_CALL_CONN_ACTIVE, filter analyzing....
[4016, , 4016]
Jun  4 22:30:24.222: //CMM/DETAIL:called number is not specified. [4016, , 4016]
Jun  4 22:30:24.222: //CMM/DETAIL:originateFilter[callid=99685, pdn=16, pchan=1] is not
set, [4016, , 4016] is not filtered
Jun  4 22:30:25.670: //CMM/INFO:
Jun  4 22:30:25.670: //CMM/INFO:
Jun  4 22:30:25.670: //CMM/INFO:cmm_notify_trigger() 14, callID 99686, 8101, 1902058375, 0
Jun  4 22:30:25.670: //CMM/INFO:    target_node 65DB15E4
Jun  4 22:30:25.670: //CMM/INFO:   - dn 8101

Jun  4 22:30:25.670: //CMM/INFO:  CallEntry 709C2988
Jun  4 22:30:25.670: //CMM/INFO:  dstCallID 99685
Jun  4 22:30:25.670: //CMM/INFO:  line_info 65DB15F0, dn 8101
Jun  4 22:30:25.670: //CMM/INFO:    * cmm_crs_proc_tr_call_active
Jun  4 22:30:25.670: //CMM/INFO:    callID = 99686,src_callid  = 99685, CG 4016, CD =
8101, GCID =05591A85-122211DC-8645A1CA-4B604A7A
Jun  4 22:30:25.670: //CMM/INFO:increase_gcid_ref_count 99686 0
Jun  4 22:30:25.670: //CMM/INFO:find_gcidinfo_node
Jun  4 22:30:25.670: //CMM/INFO:      target_node 6544A9CC
Jun  4 22:30:25.670: //CMM/INFO:   - gcid 05591A85-122211DC-8645A1CA-4B604A7A
Jun  4 22:30:25.670: //CMM/INFO:insert_ssptrs_to_gcid for line_info 65DB15F0 (dn 8101),
GCID 05591A85-122211DC-8645A1CA-4B604A7A
Jun  4 22:30:25.670:     ss_ptr list :-
Jun  4 22:30:25.670: //CMM/INFO:    adding server tag 1 refID A623DA05
Jun  4 22:30:25.670:     ss_ptr list :-  1,
Jun  4 22:30:25.670: //CMM/INFO:      count = 2
Jun  4 22:30:25.670: //CMM/INFO:    set originalCalled = 8101
Jun  4 22:30:25.670: //CMM/INFO:
Jun  4 22:30:25.670: //CMM/INFO:
```

debug capf-server

To collect debug information about the CAPF server, use the **debug capf-server** command in privileged EXEC mode. To disable collection of debug information, use the **no** form of this command.

debug capf-server {**all** | **error** | **events** | **messages**}

no debug capf-server

Syntax Description

all	Collect all CAPF information available.
error	Collect only information about CAPF errors.
events	Collect only information about CAPF status events.
messages	Collect only CAPF system messages.

Command Default

Collection of CAPF debug information is disabled.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Modification
12.4(4)XC	This command was introduced.
12.4(9)T	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used with Cisco Unified CallManager Express phone authentication.

Examples

The following example shows debug messages for the CAPF server.

```
Router# debug capf-server all

001891: .Jul 21 18:17:07.014: %IPPHONE-6-UNREGISTER_NORMAL: ephone-1:SEP000E325C9A43
IP:10.10.10.194 So
cket:3 DeviceType:Phone has unregistered normally.
001892: .Jul 21 18:17:20.495: New Connection from phone, socket 1
001893: .Jul 21 18:17:20.495: Created New Handshake Process
001894: .Jul 21 18:17:20.499: SSL Handshake Error -6983
001895: .Jul 21 18:17:21.499: SSL Handshake Error -6983
001896: .Jul 21 18:17:22.555: SSL Handshake Successful
001897: .Jul 21 18:17:22.555: ephone_capf_send_auth_req:
001898: .Jul 21 18:17:22.555: ephone_capf_ssl_write: 12 bytes
001899: .Jul 21 18:17:22.711: ephone_capf_ssl_read: Read 35 bytes
001900: .Jul 21 18:17:22.711: ephone_capf_handle_phone_msg: msgtype 2
001901: .Jul 21 18:17:22.711: ephone_capf_process_auth_res_msg: SEP000E325C9A43 AuthMode 2
001902: .Jul 21 18:17:22.711: ephone_capf_send_delete_cert_req_msg: SEP000E325C9A43
001903: .Jul 21 18:17:22.711: ephone_capf_ssl_write: 8 bytes
001904: .Jul 21 18:17:23.891: ephone_capf_ssl_read: Read 12 bytes
001905: .Jul 21 18:17:23.891: ephone_capf_handle_phone_msg: msgtype 14
001906: .Jul 21 18:17:23.891: certificate delete successful for SEP000E325C9A43
```

```
001907: .Jul 21 18:17:24.695: ephone_capf_release_session: SEP000E325C9A43
001908: .Jul 21 18:17:24.695: ephone_capf_send_end_session_msg: SEP000E325C9A43
001909: .Jul 21 18:17:24.695: ephone_capf_ssl_write: 12 bytes
001910: .Jul 21 18:17:25.095: %IPPHONE-6-REG_ALARM: 22: Name=SEP000E325C9A43 Load=7.2(2.0)
Last=Rese
t-Reset
001911: .Jul 21 18:17:25.099: %IPPHONE-6-REGISTER: ephone-1:SEP000E325C9A43
IP:10.10.10.194 Socket:2 De
viceType:Phone has registered.

001912: .Jul 21 18:18:05.171: %IPPHONE-6-UNREGISTER_NORMAL: ephone-1:SEP000E325C9A43
IP:1.1.1.127 So
cket:2 DeviceType:Phone has unregistered normally.
001913: .Jul 21 18:18:18.288: New Connection from phone, socket 1
001914: .Jul 21 18:18:18.288: Created New Handshake Process
001915: .Jul 21 18:18:18.292: SSL Handshake Error -6983
001916: .Jul 21 18:18:19.292: SSL Handshake Error -6983
001917: .Jul 21 18:18:20.348: SSL Handshake Successful
001918: .Jul 21 18:18:20.348: ephone_capf_send_auth_req:
001919: .Jul 21 18:18:20.348: ephone_capf_ssl_write: 12 bytes^Z

001920: .Jul 21 18:18:20.492: ephone_capf_ssl_read: Read 35 bytes
001921: .Jul 21 18:18:20.492: ephone_capf_handle_phone_msg: msgtype 2
001922: .Jul 21 18:18:20.492: ephone_capf_process_auth_res_msg: SEP000E325C9A43 AuthMode 2
001923: .Jul 21 18:18:20.492: ephone_capf_send_PhKeyGenReq_msg: SEP000E325C9A43 KeySize
1024
001924: .Jul 21 18:18:20.492: ephone_capf_ssl_write: 13 bytes
001925: .Jul 21 18:18:20.540: ephone_capf_ssl_read: Read 8 bytes
001926: .Jul 21 18:18:20.540: ephone_capf_handle_phone_msg: msgtype 17
001927: .Jul 21 18:18:20.540: ephone_capf_process_req_in_progress: SEP000E325C9A43 delay
0sh
001928: .Jul 21 18:18:21.924: %SYS-5-CONFIG_I: Configured from console by user1 on console
```

debug cch323 video

To provide debugging output for video components within the H.323 subsystem, use the **debug cch323 video** command in privileged EXEC mode. To disable debugging output, use the **no** form of this command.

debug cch323 video

no debug cch323 video

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Modification
12.4(4)XC	This command was introduced.
12.4(9)T	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Use this command to enable a debugging trace for the video component in an H.323 network.

Examples

Originating Gateway Example

The following is sample output of the debugging log for an originating Cisco Unified CallManager Express (Cisco Unified CME) gateway after the **debug cch323 video** command was enabled:

```
Router# show log

Syslog logging: enabled (11 messages dropped, 487 messages rate-limited,
                0 flushes, 0 overruns, xml disabled, filtering disabled)
    Console logging: disabled
    Monitor logging: level debugging, 0 messages logged, xml disabled,
                     filtering disabled
    Buffer logging: level debugging, 1144 messages logged, xml disabled,
                    filtering disabled
    Logging Exception size (4096 bytes)
    Count and timestamp logging messages: disabled
    Trap logging: level informational, 1084 message lines logged

Log Buffer (6000000 bytes):

Jun 13 09:19:42.006: //103030/C7838B198002/H323/cch323_get_peer_info: Entry
Jun 13 09:19:42.006: //103030/C7838B198002/H323/cch323_get_peer_info: Have peer
Jun 13 09:19:42.006: //103030/C7838B198002/H323/cch323_set_pref_codec_list: First
preferred codec(bytes)=16(20)
Jun 13 09:19:42.006: //103030/C7838B198002/H323/cch323_get_peer_info: Flow Mode set to
FLOW_THROUGH
Jun 13 09:19:42.006: //103030/C7838B198002/H323/cch323_get_caps_chn_info: No peer leg
setup params
Jun 13 09:19:42.006: //103030/C7838B198002/H323/cch323_get_caps_chn_info: Setting
CCH323_SS_NTFY_VIDEO_INFO
```

```
Jun 13 09:19:42.006: //103030/C7838B198002/H323/cch323_set_h323_control_options_outgoing:
h245 sm mode = 8463
Jun 13 09:19:42.006: //103030/C7838B198002/H323/cch323_set_h323_control_options_outgoing:
h323_ctl=0x20
Jun 13 09:19:42.010: //103030/C7838B198002/H323/cch323_rotary_validate: No peer_ccb
available
```

Terminating Gateway Example

The following is sample output of the debugging log for a terminating Cisco Unified Survivable Remote Site Telephony (Cisco Unified SRST) gateway after the **debug cch323 video** command was enabled:

```
Router# show log

Syslog logging: enabled (11 messages dropped, 466 messages rate-limited,
                0 flushes, 0 overruns, xml disabled, filtering disabled)
    Console logging: disabled
    Monitor logging: level debugging, 0 messages logged, xml disabled,
                     filtering disabled
    Buffer logging: level debugging, 829 messages logged, xml disabled,
                    filtering disabled
    Logging Exception size (4096 bytes)
    Count and timestamp logging messages: disabled
    Trap logging: level informational, 771 message lines logged

Log Buffer (200000 bytes):

Jun 13 09:19:42.011: //103034/C7838B198002/H323/setup_ind: Receive bearer cap infoXRate
24, rateMult 12
Jun 13 09:19:42.011: //103034/C7838B198002/H323/cch323_set_h245_state_mc_mode_incoming:
h245 state m/c mode=0x10F, h323_ctl=0x2F
Jun 13 09:19:42.015: //-1/xxxxxxxxxxxx/H323/cch245_event_handler: callID=103034
Jun 13 09:19:42.019: //-1/xxxxxxxxxxxx/H323/cch245_event_handler: Event
CC_EV_H245_SET_MODE: data ptr=0x465D5760
Jun 13 09:19:42.019: //-1/xxxxxxxxxxxx/H323/cch323_set_mode: callID=103034, flow Mode=1
spi_mode=0x6
Jun 13 09:19:42.019: //103034/C7838B198002/H323/cch323_do_call_proceeding: set_mode NOT
called yet...saved deferred CALL_PROC
Jun 13 09:19:42.019: //103034/C7838B198002/H323/cch323_h245_connection_sm: state=0,
event=0, ccb=4461B518, listen state=0
Jun 13 09:19:42.019: //103034/C7838B198002/H323/cch323_process_set_mode: Setting inbound
leg mode flags to 0x10F, flow-mode to FLOW_THROUGH
Jun 13 09:19:42.019: //103034/C7838B198002/H323/cch323_process_set_mode: Sending deferred
CALL_PROC
Jun 13 09:19:42.019: //103034/C7838B198002/H323/cch323_do_call_proceeding: set_mode called
so we can proceed with CALLPROC
Jun 13 09:19:42.027: //103034/C7838B198002/H323/cch323_h245_connection_sm: state=1,
event=2, ccb=4461B518, listen state=1
Jun 13 09:19:42.027: //103034/C7838B198002/H323/cch323_send_cap_request: Setting mode to
VIDEO MODE
Jun 13 09:19:42.031: //103034/C7838B198002/H323/cch323_h245_cap_ind: Masks au=0xC data=0x2
uinp=0x32
```

Related Commands

Command	Description
debug ephone video	Sets video debugging for the Cisco Unified IP phone.
show call active video	Displays call information for SCCP video calls in progress.

Command (continued)	Description
show call history video	Displays call history information for SCCP video calls.
show debugging	Displays information about the types of debugging that are enabled for your router.

debug cme-xml

To generate debug messages for the Cisco Unified CallManager Express XML application, use the **debug cme-xml** command in privileged EXEC mode. To disable debugging, use the **no** form of the command.

debug cme-xml

no debug cme-xml

Syntax Description

This command has no keywords or arguments.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Modification
12.4(4)XC	This command was introduced.
12.4(9)T	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

The **show fb-its-log** command displays the contents of the XML event table.

Examples

The following example shows the progress of an XML request that has been sent to Cisco Unified CallManager Express:

```
Router# debug cme-xml

*Aug 5 06:27:25.727: CME got a raw XML message.
*Aug 5 06:27:25.727: doc 0x63DB85E8, doc->doc_type 3, req 0x655FDCD0
*Aug 5 06:27:25.727: CME extracted a XML document
*Aug 5 06:27:25.727: Response buffer 0x63DCFD58, len = 4096
*Aug 5 06:27:25.727: First Tag ID SOAP_HEADER_TAG_ID 58720257
*Aug 5 06:27:25.727: First Attribute ID SOAP_ENV_ATTR 50331649
*Aug 5 06:27:25.727: cme_xml_process_soap_header
*Aug 5 06:27:25.727: cme_xml_process_soap_body
*Aug 5 06:27:25.731: cme_xml_process_axl
*Aug 5 06:27:25.731: cme_xml_process_request
*Aug 5 06:27:25.731: cme_xml_process_ISgetGlobal
*Aug 5 06:27:25.731: CME XML sent 811 bytes response.
```

Related Commands

Command	Description
show fb-its-log	Displays Cisco Unified CallManager Express XML API information.

debug credentials

To set debugging on the credentials service that runs between the Cisco Unified CME CTL provider and CTL client or between the Cisco Unified SRST router and Cisco Unified CallManager, use the **debug credentials** command in privileged EXEC mode. To disable debugging output, use the **no** form of this command.

debug credentials

no debug credentials

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Modification
12.3(14)T	This command was introduced for Cisco Unified SRST.
12.4(4)XC	This command was introduced for Cisco Unified CME.
12.4(9)T	This command was integrated into Cisco IOS Release 12.4(9)T for Cisco Unified CME.

Usage Guidelines

Cisco Unified CME

Use this command with Cisco Unified CME phone authentication to monitor a CTL provider as it provides credentials to the CTL client.

Cisco Unified SRST

Use this command to monitor Cisco Unified CallManager while it requests certificates from the Cisco Unified SRST router. It sets debugging on the credentials service that runs between the SRST router and Cisco Unified CallManager

Examples

Cisco Unified CME

The following sample output displays the CTL provider establishing a TLS session with the CTL client and providing all the relevant credentials for the services that are running on this router to the CTL client.

```
Router# debug credentials

Credentials server debugging is enabled

May 25 12:08:17.944: Credentials service: Start TLS Handshake 1 10.5.43.174 4374
May 25 12:08:17.948: Credentials service: TLS Handshake returns OPSSLReadWouldBlockErr
May 25 12:08:18.948: Credentials service: TLS Handshake returns OPSSLReadWouldBlockErr
May 25 12:08:19.948: Credentials service: TLS Handshake returns OPSSLReadWouldBlockErr
May 25 12:08:20.964: Credentials service: TLS Handshake completes
```

Cisco Unified SRST

The following is sample output showing the credentials service that runs between the Cisco Unified SRST router and Cisco Unified CallManager. The credentials service provides Cisco Unified CallManager with the certificate from the SRST router.

```
Router# debug credentials

Credentials server debugging is enabled
Router#
May 25 12:08:17.944: Credentials service: Start TLS Handshake 1 10.5.43.174 4374
May 25 12:08:17.948: Credentials service: TLS Handshake returns OPSSLReadWouldBlockErr
May 25 12:08:18.948: Credentials service: TLS Handshake returns OPSSLReadWouldBlockErr
May 25 12:08:19.948: Credentials service: TLS Handshake returns OPSSLReadWouldBlockErr
May 25 12:08:20.964: Credentials service: TLS Handshake completes
```

Table 2 describes the significant fields shown in the display.

Table 2 debug credentials Field Descriptions

Field	Description
Start TLS Handshake 1 10.5.43.174 4374	Indicates the beginning of the TLS handshake between the secure Cisco Unified SRST router and Cisco Unified CallManager. In this example, 1 indicates the socket, 10.5.43.174 is the IP address, and 4374 is the port of Cisco Unified CallManager.
TLS Handshake returns OPSSLReadWouldBlockErr	Indicates that the handshake is in process.
TLS Handshake completes	Indicates that the TLS handshake has finished and that the Cisco Unified CallManager has received the secure Cisco Unified SRST device certificate.

Related Commands

Command	Description
credentials	Enters credentials configuration mode to configure a Cisco Unified CME CTL provider certificate or a Cisco Unified SRST router certificate.
ctl-service admin	Specifies a user name and password to authenticate the CTL client during the CTL protocol.
ip source-address (credentials)	Enables the Cisco Unified CME or SRST router to receive messages through the specified IP address and port.
show credentials	Displays the credentials settings on a Cisco Unified CME or SRST router.
show debugging	Displays information about the types of debugging that are enabled for your router.
trustpoint (credentials)	Specifies the name of the trustpoint to be associated with a Cisco Unified CME CTL provider certificate or with a Cisco Unified SRST router certificate.

debug ctl-client

To collect debug information about the CTL client, use the **debug ctl-client** command in privileged EXEC configuration mode. To disable collection of debug information, use the **no** form of this command.

debug ctl-client

no debug ctl-client

Syntax Description

This command has no arguments or keywords.

Command Default

Collection of CTL client debug information is disabled.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Modification
12.4(4)XC	This command was introduced.
12.4(9)T	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used with Cisco Unified CME phone authentication.

Examples

The following example shows debug messages for the CTL client:

```
Router# debug ctl-client

001954: .Jul 21 18:23:02.136: ctl_client_create_ctlfile:
001955: .Jul 21 18:23:02.272: create_ctl_record: Function 0 Trustpoint cisco1
001956: .Jul 21 18:23:02.276: create_ctl_record: record added for function 0
001957: .Jul 21 18:23:02.276: create_ctl_record: Function 0 Trustpoint sast2
001958: .Jul 21 18:23:02.280: create_ctl_record: record added for function 0
001959: .Jul 21 18:23:02.280: create_ctl_record: Function 1 Trustpoint cisco1
001960: .Jul 21 18:23:02.284: create_ctl_record: record added for function 1
001961: .Jul 21 18:23:02.284: create_ctl_record: Function 3 Trustpoint cisco1
001962: .Jul 21 18:23:02.288: create_ctl_record: record added for function 3
001963: .Jul 21 18:23:02.288: create_ctl_record: Function 4 Trustpoint cisco1
001964: .Jul 21 18:23:02.292: create_ctl_record: record added for function 4
001965: .Jul 21 18:23:02.424: ctl_client_create_ctlfile: Signature length 128
001966: .Jul 21 18:23:02.640: CTL File Created Successfully
```

debug ephone alarm

To set SkinnyStation alarm messages debugging for the Cisco IP phone, use the **debug ephone alarm** command in privileged EXEC mode. To disable debugging output, use the **no** form of this command.

debug ephone alarm [**mac-address** *mac-address*]

no debug ephone alarm [**mac-address** *mac-address*]

Syntax Description

mac-address	(Optional) Defines the MAC address of the Cisco IP phone.
mac-address	(Optional) Specifies the MAC address of the Cisco IP phone.

Defaults

No default behavior or values

Command Modes

Privileged EXEC

Command History

Release	Modification
12.2(2)XT	This command was introduced on the following platforms: Cisco 1750, Cisco 1751, Cisco 2600 series and Cisco 3600 series multiservice routers; and Cisco IAD2420 series Integrated Access Devices (IADs).
12.2(8)T	This command was implemented on the Cisco 3725 and Cisco 3745 routers.
12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
12.2(11)T	This command was implemented on the Cisco 1760 routers.

Usage Guidelines

The **debug ephone alarm** command shows all the SkinnyStation alarm messages sent by the Cisco IP phone. Under normal circumstances, this message is sent by the Cisco IP phone just before it registers, and the message has the severity level for the alarm set to “Informational” and contains the reason for the phone reboot or re-register. This type of message is entirely benign and does not indicate an error condition.

If the **mac-address** keyword is not used, the **debug ephone alarm** command debugs all Cisco IP phones that are registered to the router. You can remove debugging for the Cisco IP phones that you do not want to debug by using the **mac-address** keyword with the **no** form of this command.

You can enable or disable debugging on any number of Cisco IP phones. To see the Cisco IP phones that have debugging enabled, enter the **show ephone** command and look at the debug field in the output. When debugging is enabled for a Cisco IP phone, the debug output is displayed for the directory numbers associated with the Cisco IP phone.

Examples

The following example shows a SkinnyStation alarm message that is sent before the Cisco IP phone registers:

```
Router# debug ephone alarm

phone keypad reset
CM-closed-TCP
CM-bad-state
```

Related Commands

Command	Description
debug ephone detail	Sets detail debugging for the Cisco IP phone.
debug ephone error	Sets error debugging for the Cisco IP phone.
debug ephone keepalive	Sets keepalive debugging for the Cisco IP phone.
debug ephone loopback	Sets MWI debugging for the Cisco IP phone.
debug ephone pak	Provides voice packet level debugging and prints the contents of one voice packet in every 1024 voice packets.
debug ephone raw	Provides raw low-level protocol debugging display for all SCCP messages.
debug ephone register	Sets registration debugging for the Cisco IP phone.
debug ephone state	Sets state debugging for the Cisco IP phone.
debug ephone statistics	Sets statistics debugging for the Cisco IP phone.
show debugging	Displays information about the types of debugging that are enabled for your router.

debug ephone blf

To display debugging information for Busy Lamp Field (BLF) presence features, use the **debug ephone blf** command in privileged EXEC mode. To disable debugging, use the **no** form of this command.

debug ephone blf [**mac-address** *mac-address*]

no debug ephone blf [**mac-address** *mac-address*]

Syntax Description

mac-address *mac-address*	(Optional) Specifies the MAC address of a specific IP phone.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.4(11)XJ	This command was introduced.
12.4(15)T	This command was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

Use this command for troubleshooting BLF speed-dial and BLF call-list features for phones in a presence service.

Examples

The following is sample output from the **debug ephone blf** command.

```
Router# debug ephone blf

EPHONE BLF debugging is enabled

*Sep  4 07:18:26.307: skinny_asnl_callback: subID 16 type 4
*Sep  4 07:18:26.307: ASNL_RESP_NOTIFY_INDICATION
*Sep  4 07:18:26.307: ephone-1[1]:ASNL notify indication message, feature index 4, subID
[16]
*Sep  4 07:18:26.307: ephone-1[1]:line status 6, subID [16]
*Sep  4 07:18:26.307: ephone-1[1]:StationFeatureStatV2Message sent, status 2
*Sep  4 07:18:26.307: skinny_asnl_callback: subID 23 type 4
*Sep  4 07:18:26.307: ASNL_RESP_NOTIFY_INDICATION
*Sep  4 07:18:26.307: ephone-2[2]:ASNL notify indication message, feature index 2, subID
[23]
*Sep  4 07:18:26.311: ephone-2[2]:line status 6, subID [23]
*Sep  4 07:18:26.311: ephone-2[2]:StationFeatureStatV2Message sent, status 2
*Sep  4 07:18:28.951: skinny_asnl_callback: subID 16 type 4
*Sep  4 07:18:28.951: ASNL_RESP_NOTIFY_INDICATION
*Sep  4 07:18:28.951: ephone-1[1]:ASNL notify indication message, feature index 4, subID
[16]
*Sep  4 07:18:28.951: ephone-1[1]:line status 1, subID [16]
*Sep  4 07:18:28.951: ephone-1[1]:StationFeatureStatV2Message sent, status 1
*Sep  4 07:18:28.951: skinny_asnl_callback: subID 23 type 4
*Sep  4 07:18:28.951: ASNL_RESP_NOTIFY_INDICATION
*Sep  4 07:18:28.951: ephone-2[2]:ASNL notify indication message, feature index 2, subID
[23]
*Sep  4 07:18:28.951: ephone-2[2]:line status 1, subID [23]
```

```
*Sep  4 07:18:28.951: ephone-2[2]:StationFeatureStatV2Message sent, status 1
```

Related Commands

Command	Description
blf-speed-dial	Enables BLF monitoring for a speed-dial number on a phone registered to Cisco Unified CME.
presence call-list	Enables BLF monitoring for call lists and directories on phones registered to a Cisco Unified CME router.
show presence global	Displays configuration information about the presence service.
show presence subscription	Displays information about active presence subscriptions.

debug ephone ccm-compatible

To display Cisco CallManager notification updates for calls between Cisco CallManager and Cisco CallManager Express, use the **debug ephone ccm-compatible** command in privileged EXEC mode. To disable debugging output, use the **no** form of this command.

debug ephone ccm-compatible [**mac-address** *mac-address*]

no debug ephone ccm-compatible [**mac-address** *mac-address*]

Syntax Description

mac-address *mac-address*	(Optional) Specifies the MAC address of a Cisco IP phone for debugging.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.3(7)T	This command was introduced.

Usage Guidelines

This command displays call flow notification information for all calls between Cisco CallManager and Cisco CallManager Express, but it is most useful for filtering out specific information for transfer and forward cases. For basic call information, use the **debug ephone state** command.

If you do not specify the **mac-address** keyword, the **debug ephone ccm-compatible** command debugs all Cisco IP phones that are registered to the router. You can remove debugging for the Cisco IP phones that you do not want to debug by using the **no** form of this command with the **mac-address** keyword.

Debugging can be enabled or disabled on any number of Cisco IP phones. Cisco IP phones that have debugging enabled are listed in the debug field of the **show ephone** command output. When debugging is enabled for a Cisco IP phone, debug output is displayed for all phone extensions (virtual voice ports) associated with that phone.

Examples

The following sample output displays call flow notifications between Cisco CallManager and Cisco CallManager Express:

```
Router# debug ephone ccm-compatible

*May  1 04:30:02.650:ephone-2[2]:DtAlertingTone/DtHoldTone - mediaActive reset during
CONNECT
*May  1 04:30:02.654:ephone-2[2]:DtHoldTone - force media STOP state
*May  1 04:30:02.654://93/xxxxxxxxxxxx/CCAPI/ccCallNotify:(callID=0x5D,nData->
bitmask=0x00000007)
*May  1 04:30:02.654://93/xxxxxxxxxxxx/VTSP:(50/0/3):-1:0:5/vtsp_process_event:
 vtsp:[50/0/3 (93), S_CONNECT, E_CC_SERVICE_MSG]
*May  1 04:30:02.654://93/xxxxxxxxxxxx/VTSP:(50/0/3):-1:0:5/act_service_msg_dow
n:.
*May  1 04:30:02.658:dn_callerid_update DN 3 number= 12009 name= CCM7960 in state
CONNECTED
*May  1 04:30:02.658:dn_callerid_update (incoming) DN 3 info updated to
*May  1 04:30:02.658:calling= 12009 called= 13003 origCalled=
```

```
*May  1 04:30:02.658:callingName= CCM7960, calledName= , redirectedTo =
*May  1 04:30:02.658:ephone-2[2][SEP003094C2999A]:refreshDisplayLine for line 1
 DN 3 chan 1
*May  1 04:30:03.318:ephone-2[2]:DisplayCallInfo incoming call
*May  1 04:30:03.318:ephone-2[2]:Call Info DN 3 line 1 ref 24 called 13003 calling 12009
origcalled 13003 calltype 1
*May  1 04:30:03.318:ephone-2[2]:Original Called Name UUT4PH3
*May  1 04:30:03.318:ephone-2[2]:CCM7960 calling
*May  1 04:30:03.318:ephone-2[2]:UUT4PH3
```

Related Commands

Command	Description
debug ephone state	Displays call state information.
show debugging	Displays information about the types of debugging that are enabled for your router.
show ephone	Displays information about registered Cisco IP phones.

debug ephone detail

To set detail debugging for the Cisco IP phone, use the **debug ephone detail** command in privileged EXEC mode. To disable debugging output, use the **no** form of this command.

debug ephone detail [**mac-address** *mac-address*]

no debug ephone detail [**mac-address** *mac-address*]

Syntax Description

mac-address	(Optional) Defines the MAC address of the Cisco IP phone.
mac-address	(Optional) Specifies the MAC address of the Cisco IP phone.

Defaults

No default behavior or values

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

The **debug ephone detail** command includes the error and state levels.

If the **mac-address** keyword is not used, the **debug ephone detail** command debugs all Cisco IP phones that are registered to the router. You can remove debugging for the Cisco IP phones that you do not want to debug by using the **mac-address** keyword with the **no** form of this command.

You can enable or disable debugging on any number of Cisco IP phones. To see the Cisco IP phones that have debugging enabled, enter the **show ephone** command and look at the debug field in the output. When debugging is enabled for a Cisco IP phone, the debug output is displayed for the directory numbers associated with the Cisco IP phone.

Examples

The following is sample output of detail debugging of the Cisco IP phone with MAC address 0030.94c3.8724. The sample is an excerpt of some of the activities that takes place during call setup, connected state, active call, and the call being disconnected.

```
Router# debug ephone detail mac-address 0030.94c3.8724

Ephone detail debugging is enabled

1d04h: ephone-1[1]:OFFHOOK
.
.
1d04h: Skinny Call State change for DN 1 SIEZE
.
.
1d04h: ephone-1[1]:SetCallState line 1 DN 1 TsOffHook
.
.
1d04h: ephone-1[1]:SetLineLamp 1 to ON
.
.
1d04h: ephone-1[1]:KeypadButtonMessage 5
.
.
1d04h: ephone-1[1]:KeypadButtonMessage 0
.
.
1d04h: ephone-1[1]:KeypadButtonMessage 0
.
.
1d04h: ephone-1[1]:KeypadButtonMessage 2
.
.
1d04h: ephone-1[1]:Store ReDial digit: 5002
.
SkinnyTryCall to 5002 instance 1
.
.
1d04h: ephone-1[1]:Store ReDial digit: 5002
1d04h: ephone-1[1]:
SkinnyTryCall to 5002 instance 1
.
.
1d04h: Skinny Call State change for DN 1 ALERTING
.
.
1d04h: ephone-1[1]:SetCallState line 1 DN 1 TsRingOut
.
.
1d04h: ephone-1[1]:SetLineLamp 1 to ON
1d04h: SetCallInfo calling dn 1 dn 1
calling [5001] called [5002]
.
.
1d04h: ephone-1[1]: Jane calling
1d04h: ephone-1[1]: Jill
.
.
1d04h: SkinnyUpdateDnState by EFXS_RING_GENERATE
  for DN 2 to state RINGING
.
.
1d04h: SkinnyGetCallState for DN 2 CONNECTED
.
```

```
.
1d04h: ephone-1[1]:SetLineLamp 3 to ON
1d04h: ephone-1[1]:UpdateCallState DN 1 state 4 calleddn 2
.
.
1d04h: Skinny Call State change for DN 1 CONNECTED
.
.
1d04h: ephone-1[1]:OpenReceive DN 1 codec 4:G711Ulaw64k  duration 10 ms bytes 80
.
.
1d04h: ephone-1[1]:OpenReceiveChannelAck 1.2.172.21 port=20180
1d04h: ephone-1[1]:Outgoing calling DN 1 Far-ephone-2 called DN 2
1d04h: SkinnyGetCallState for DN 1 CONNECTED
.
.
1d04h: ephone-1[1]:SetCallState line 3 DN 2 TsOnHook
.
.
1d04h: ephone-1[1]:SetLineLamp 3 to OFF
.
.
1d04h: ephone-1[1]:SetCallState line 1 DN 1 TsOnHook
.
.
1d04h: ephone-1[1]:Clean Up Speakerphone state
1d04h: ephone-1[1]:SpeakerPhoneOnHook
1d04h: ephone-1[1]:Clean up activeline 1
1d04h: ephone-1[1]:StopTone sent to ephone
1d04h: ephone-1[1]:Clean Up phone offhook state
1d04h: SkinnyGetCallState for DN 1 IDLE
1d04h: called DN -1, calling DN -1 phone -1
1d04h: ephone-1[1]:SetLineLamp 1 to OFF
1d04h: UnBinding ephone-1 from DN 1
1d04h: UnBinding called DN 2 from DN 1
1d04h: ephone-1[1]:ONHOOK
1d04h: ephone-1[1]:SpeakerPhoneOnHook
1d04h: ephone-1[1]:ONHOOK NO activeline
.
```

Related Commands.

Command	Description
debug ephone alarm	Sets SkinnyStation alarm messages debugging for the Cisco IP phone.
debug ephone error	Sets error debugging for the Cisco IP phone.
debug ephone keepalive	Sets keepalive debugging for the Cisco IP phone.
debug ephone loopback	Sets MWI debugging for the Cisco IP phone.
debug ephone pak	Provides voice packet level debugging and prints the contents of one voice packet in every 1024 voice packets.
debug ephone raw	Provides raw low-level protocol debugging display for all SCCP messages.
debug ephone register	Sets registration debugging for the Cisco IP phone.
debug ephone state	Sets state debugging for the Cisco IP phone.
debug ephone statistics	Sets statistics debugging for the Cisco IP phone.
show debugging	Displays information about the types of debugging that are enabled for your router.

debug ephone error

To set error debugging for the Cisco IP phone, use the **debug ephone error** command in privileged EXEC mode. To disable debugging output, use the **no** form of this command.

debug ephone error [**mac-address** *mac-address*]

no debug ephone error [**mac-address** *mac-address*]

Syntax Description

mac-address	(Optional) Defines the MAC address of the Cisco IP phone.
mac-address	(Optional) Specifies the MAC address of the Cisco IP phone.

Defaults

No default behavior or values

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

The **debug ephone error** command cancels debugging at the detail and state level.

If the **mac-address** keyword is not used, the **debug ephone error** command debugs all Cisco IP phones that are registered to the router. You can remove debugging for the Cisco IP phones that you do not want to debug by using the **mac-address** keyword with the **no** form of this command.

You can enable or disable debugging on any number of Cisco IP phones. To see the Cisco IP phones that have debugging enabled, enter the **show ephone** command and look at the debug field in the output. When debugging is enabled for a Cisco IP phone, the debug output is displayed for the directory numbers associated with the Cisco IP phone.

Examples

The following is sample output of error debugging for the Cisco IP phone with MAC address 0030.94c3.8724:

```
Router# debug ephone error mac-address 0030.94c3.8724

EPHONE error debugging is enabled

socket [2] send ERROR 11
Skinny Socket [2] retry failure
```

Related Commands

Command	Description
debug ephone alarm	Sets SkinnyStation alarm messages debugging for the Cisco IP phone.
debug ephone detail	Sets detail debugging for the Cisco IP phone.
debug ephone keepalive	Sets keepalive debugging for the Cisco IP phone.
debug ephone loopback	Sets MWI debugging for the Cisco IP phone.
debug ephone pak	Provides voice packet level debugging and prints the contents of one voice packet in every 1024 voice packets.
debug ephone raw	Provides raw low-level protocol debugging display for all SCCP messages.
debug ephone register	Sets registration debugging for the Cisco IP phone.
debug ephone state	Sets state debugging for the Cisco IP phone.
debug ephone statistics	Sets statistics debugging for the Cisco IP phone.
show debugging	Displays information about the types of debugging that are enabled for your router.

debug ephone extension-assigner

To display status messages produced by the extension assigner application, use the **debug ephone extension-assigner** command in privileged EXEC mode. To disable debugging output, use the **no** form of this command.

debug ephone extension-assigner

no debug ephone extension-assigner

Syntax Description

This command has no arguments or keywords.

Command Default

Debug ephone extension-assigner is disabled.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC4	Cisco Unified CME 4.0(3)	This command was introduced.
12.4(11)XJ	Cisco Unified CME 4.1	This command was introduced.
12.4(15)T	Cisco Unified CME 4.1	This command was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

This command displays status messages produced by the extension assigner application, including messages related to the functions performed by the following Tcl commands:

- phone query**—**Verifies whether the ephone tag has been assigned a MAC address.
- phone assign**—**Binds the MAC address from the caller's phone to a preexisting ephone template.
- phone unassign**—**Removes the MAC address from the ephone tag.

Before using this command, you must load the Tcl script for the extension assigner application.

Examples

The following is sample output of extension assigner debugging as the extension assigner application queries phones for their status and issues commands to assign or unassign extension numbers.

```
*Jun 9 19:08:10.627: ephone_query: inCallID=47, tag=4, ephone_tag=4
*Jun 9 19:08:10.627: extAssigner_IsEphoneMacPreset: ephone_tag = 4,
ipKeyswitch.max_ephones = 96
*Jun 9 19:08:10.627: extAssigner_IsEphoneMacPreset: ephone_ptr->mac_addr_str =
000B46BDE075, MAC_EXT_RESERVED_VALUE = 02EAEAEA0000
*Jun 9 19:08:10.627: SkinnyGetActivePhoneIndexFromCallid: callID = 47
*Jun 9 19:08:10.627: SkinnyGetActivePhoneIndexFromCallid: vdbPtr->physical_interface_type
(26); CV_VOICE_EFXS (26)
*Jun 9 19:08:10.627: SkinnyGetActivePhoneIndexFromCallid: vdbPtr->type (6);
CC_IF_TELEPHONY (6)
*Jun 9 19:08:10.627: SkinnyGetActivePhoneIndexFromCallid: htsp->sig_type (26);
CV_VOICE_EFXS (26)
```

```
*Jun 9 19:08:10.627: SkinnyGetActivePhoneIndexFromCallid: dn = 4, chan = 1
*Jun 9 19:08:10.627: ephone_query: EXTASSIGNER_RC_SLOT_ASSIGNED_TO_CALLING_PHONE
*Jun 9 19:08:22.763: ephone_unassign: inCallID=47, tag=4, ephone_tag=4
*Jun 9 19:08:22.763: extAssigner_IsEphoneMacPreset: ephone_tag = 4,
ipKeyswitch.max_ephones = 96

*Jun 9 19:08:22.763: extAssigner_IsEphoneMacPreset: ephone_ptr->mac_addr_str =
000B46BDE075, MAC_EXT_RESERVED_VALUE = 02EAEAEA000
*Jun 9 19:08:22.763: is_ephone_auto_assigned: button-1 dn_tag=4
*Jun 9 19:08:22.763: is_ephone_auto_assigned: NO
*Jun 9 19:08:22.763: SkinnyGetActivePhoneIndexFromCallid: callID = 47
*Jun 9 19:08:22.763: SkinnyGetActivePhoneIndexFromCallid: vdbPtr->physical_interface_type
(26); CV_VOICE_EFXS (26)
*Jun 9 19:08:22.767: SkinnyGetActivePhoneIndexFromCallid: vdbPtr->type (6);
CC_IF_TELEPHONY (6)
*Jun 9 19:08:22.767: SkinnyGetActivePhoneIndexFromCallid: htsp->sig_type (26);
CV_VOICE_EFXS (26)
*Jun 9 19:08:22.767: SkinnyGetActivePhoneIndexFromCallid: dn = 4, chan = 1
*Jun 9 19:08:29.795: ephone-4[8]:fStationOnHookMessage: Extension Assigner request
restart, cmd=2, new mac=02EAEAEA0004, ephone_tag=4
*Jun 9 19:08:30.063: %IPPHONE-6-UNREGISTER_NORMAL: ephone-4:SEP000B46BDE075 IP:5.5.0.1
Socket:8 DeviceType:Phone has unregistered normally.
*Jun 9 19:08:30.063: ephone-4[8][SEP000B46BDE075]:extAssigner_assign: new
mac=02EAEAEA0004, ephone-tag=4
*Jun 9 19:08:30.063: extAssigner_simple_assign: mac=02EAEAEA0004, tag=4
*Jun 9 19:08:30.063: ephone_updateCNF: update cnf_file ephone_tag=4
*Jun 9 19:08:30.063: extAssigner_assign: restart again (mac=02EAEAEA0004) ephone_tag=4
*Jun 9 19:08:30.131: %IPPHONE-6-REG_ALARM: 23: Name=SEP000B46BDE075 Load=8.0(2.0)
Last=Reset-Restart
*Jun 9 19:08:30.135: %IPPHONE-6-REGISTER_NEW: ephone-7:SEP000B46BDE075 IP:5.5.0.1
Socket:10 DeviceType:Phone has registered.
*Jun 9 19:08:30.503: %IPPHONE-6-UNREGISTER_NORMAL: ephone-7:SEP000B46BDE075 IP:5.5.0.1
Socket:10 DeviceType:Phone has unregistered normally.
*Jun 9 19:08:43.127: %IPPHONE-6-REG_ALARM: 22: Name=SEP000B46BDE075 Load=8.0(2.0)
Last=Reset-Reset
*Jun 9 19:08:43.131: %IPPHONE-6-REGISTER: ephone-7:SEP000B46BDE075 IP:5.5.0.1 Socket:13
DeviceType:Phone has registered.
```

Related Commands

Command	Description
debug ephone state	Sets state debugging for Cisco IP phones.
debug voip application script	Displays status messages produced by voice over IP application scripts.

debug ephone keepalive

To set keepalive debugging for the Cisco IP phone, use the **debug ephone keepalive** command in privileged EXEC mode. To disable debugging output, use the **no** form of this command.

debug ephone keepalive [**mac-address** *mac-address*]

no debug ephone keepalive [**mac-address** *mac-address*]

Syntax Description

mac-address	(Optional) Defines the MAC address of the Cisco IP phone.
mac-address	(Optional) Specifies the MAC address of the Cisco IP phone.

Defaults

No default behavior or values

Command Modes

Privileged EXEC

Command History

Release	Modification
12.1(5)YD	This command was introduced on the following platforms: Cisco 2600 series and Cisco 3600 series multiservice routers, and Cisco IAD2420 series Integrated Access Devices (IADs).
12.2(2)XT	This command was implemented on the Cisco 1750 and Cisco 1751 multiservice routers.
12.2(8)T	This command was implemented on the Cisco 3725 and Cisco 3745 routers.
12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
12.2(11)T	This command was implemented on the Cisco 1760 routers.

Usage Guidelines

The **debug ephone keepalive** command sets keepalive debugging.

If the **mac-address** keyword is not used, the **debug ephone keepalive** command debugs all Cisco IP phones that are registered to the router. You can remove debugging for the Cisco IP phones that you do not want to debug by using the **mac-address** keyword with the **no** form of this command.

You can enable or disable debugging on any number of Cisco IP phones. To see the Cisco IP phones that have debugging enabled, enter the **show ephone** command and look at the debug field in the output. When debugging is enabled for a Cisco IP phone, the debug output is displayed for the directory numbers associated with the Cisco IP phone.

Examples

The following is sample output of the keepalive status for the Cisco IP phone with MAC address 0030.94C3.E1A8:

```
Router# debug ephone keepalive mac-address 0030.94c3.E1A8

EPHONE keepalive debugging is enabled for phone 0030.94C3.E1A8

1d05h: ephone-1 Set interface FastEthernet0/0  ETHERNET
1d05h: ephone-1[1]:Keepalive socket[1] SEP003094C3E1A8
1d05h: ephone-1 Set interface FastEthernet0/0  ETHERNET
1d05h: ephone-1[1]:Keepalive socket[1] SEP003094C3E1A8
1d05h: Skinny Checking for stale sockets
1d05h: ephone-1 Set interface FastEthernet0/0  ETHERNET
1d05h: ephone-1[1]:Keepalive socket[1] SEP003094C3E1A8
1d05h: ephone-1 Set interface FastEthernet0/0  ETHERNET
1d05h: ephone-1[1]:Keepalive socket[1] SEP003094C3E1A8
1d05h: Skinny active socket list (3/96):  1 2 4
```

Related Commands

Command	Description
debug ephone alarm	Sets SkinnyStation alarm messages debugging for the Cisco IP phone.
debug ephone detail	Sets detail debugging for the Cisco IP phone.
debug ephone error	Sets error debugging for the Cisco IP phone.
debug ephone loopback	Sets MWI debugging for the Cisco IP phone.
debug ephone pak	Provides voice packet level debugging and prints the contents of one voice packet in every 1024 voice packets.
debug ephone raw	Provides raw low-level protocol debugging display for all SCCP messages.
debug ephone register	Sets registration debugging for the Cisco IP phone.
debug ephone state	Sets state debugging for the Cisco IP phone.
debug ephone statistics	Sets statistics debugging for the Cisco IP phone.
show debugging	Displays information about the types of debugging that are enabled for your router.

debug ephone loopback

To set debugging for loopback calls, use the **debug ephone loopback** command in privileged EXEC mode. To disable debugging, use the **no** form of this command.

debug ephone loopback [**mac-address** *mac-address*]

no debug ephone loopback [**mac-address** *mac-address*]

Syntax Description

mac-address *mac-address*	(Optional) Specifies the MAC address of a Cisco IP phone for debugging.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.2(2)XT	This command was introduced for Cisco IOS Telephony Services (now known as Cisco CallManager Express) Version 2.0 on the Cisco 1750, Cisco 1751, Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691.
12.3(4)T	This command was integrated into Cisco IOS Release 12.3(4)T.

Usage Guidelines

The **debug ephone loopback** command sets debugging for incoming and outgoing calls on all loopback-dn pairs or on the single loopback-dn pair that is associated with the IP phone that has the MAC address specified in this command.

If you enable the **debug ephone loopback** command and the **debug ephone pak** command at the same time, the output displays packet debug output for the voice packets that are passing through the loopback-dn pair.

You can enable or disable debugging on any number of Cisco IP phones. To see the Cisco IP phones that have debugging enabled, enter the **show ephone** command and look at the debug field in the output. When debugging is enabled for a Cisco IP phone, the debug output is displayed for the directory numbers associated with that Cisco IP phone.

Examples

The following example contains two excerpts of output for a call that is routed through a loopback. The first excerpt is output from the **show running-config** command and displays the loopback configuration used for this example. The second excerpt is output from the **debug ephone loopback** command.

```
Router# show running-config

.
.
.
```

```
ephone-dn 14
 number 1514
!
!
ephone-dn  42
 number 17181..
 loopback-dn 43 forward 4
 no huntstop
!
!
ephone-dn  43
 number 19115..
 loopback-dn 42 forward 4
!
.
.
.
```

A loopback call is started. An incoming call to 1911514 (ephone-dn 43) uses the loopback pair of ephone-dns to become an outgoing call to extension 1514. The number in the outgoing call has only four digits because the **loopback-dn** command specifies forwarding of four digits. The outgoing call uses ephone-dn 42, which is also specified in the **loopback-dn** command under ephone-dn 43. When the extension at 1514 rings, the following debug output is displayed:

```
Router# debug ephone loopback

Mar  7 00:57:25.376:Pass processed call info to special DN 43 chan 1
Mar  7 00:57:25.376:SkinnySetCallInfoLoopback DN 43 state IDLE to DN 42 state IDLE
Mar  7 00:57:25.376:Called Number = 1911514 Called Name =
Mar  7 00:57:25.376:Calling Number = 8101 Calling Name =
 orig Called Number =
Copy Caller-ID info from Loopback DN 43 to DN 42
Mar  7 00:57:25.376:DN 43  Forward 1514
Mar  7 00:57:25.376:PredictTarget match 1514 DN 14 is idle
Mar  7 00:57:25.380:SkinnyUpdateLoopbackState DN 43 state RINGING calledDn -1
Mar  7 00:57:25.380:Loopback DN 42 state IDLE
Mar  7 00:57:25.380:Loopback DN 43 calledDN -1 callingDn -1 G711Ulaw64k
Mar  7 00:57:25.380:SkinnyUpdateLoopbackState DN 43 to DN 42 signal OFFHOOK
Mar  7 00:57:25.380:SetDnCodec Loopback DN 43 codec 4:G711Ulaw64k  vad 0 size 160
Mar  7 00:57:25.380:SkinnyDnToneLoopback DN 42 state SIEZE to DN 43 state RINGING
Mar  7 00:57:25.380:TONE ON DtInsideDialTone
Mar  7 00:57:25.380:SkinnyDnToneLoopback called number = 1911514
Mar  7 00:57:25.380:DN 43  Forward 1514
Mar  7 00:57:25.380:DN 42 from 43  Dial 1514
Mar  7 00:57:25.384:SkinnyDnToneLoopback DN 42 state ALERTING to DN 43 state RINGING
Mar  7 00:57:25.384:TONE OFF
Mar  7 00:57:25.384:SkinnyDnToneLoopback DN 42 state ALERTING to DN 43 state RINGING
Mar  7 00:57:25.384:TONE OFF
Mar  7 00:57:25.384:SkinnyUpdateLoopbackState DN 42 state ALERTING calledDn -1
Mar  7 00:57:25.384:Loopback DN 43 state RINGING
Mar  7 00:57:25.384:Loopback Alerting DN 42 calledDN -1 callingDn -1 G711Ulaw64k
Mar  7 00:57:25.388:ephone-5[7]:DisplayCallInfo incoming call
Mar  7 00:57:25.388:SkinnyDnToneLoopback DN 42 state ALERTING to DN 43 state RINGING
Mar  7 00:57:25.388:TONE ON DtAlertingTone
Mar  7 00:57:25.388:SkinnyDnToneLoopback DN 42 to DN 43 deferred alerting by
DtAlertingTone
Mar  7 00:57:25.388:EFXS_STATE_ONHOOK_RINGING already done for DN 43 chan 1
Mar  7 00:57:25.388:Set prog_ind 0 for DN 42 chan 1
.
.
.
```

When extension 1514 answers the call, the following debug output is displayed:

```
.
.
.
Mar  7 00:57:32.158:SkinnyDnToneLoopback DN 42 state ALERTING to DN 43 state RINGING
Mar  7 00:57:32.158:TONE OFF
Mar  7 00:57:32.158:dn_support_g729 true DN 42 chan 1 (loopback)
Mar  7 00:57:32.158:SetDnCodec Loopback DN 43 codec 4:G711Ulaw64k  vad 0 size 160
Mar  7 00:57:32.158:SkinnyUpdateLoopbackState DN 42 state CALL_START calledDn 14
Mar  7 00:57:32.158:Loopback DN 43 state RINGING
Mar  7 00:57:32.158:SkinnyUpdateLoopbackState DN 42 to DN 43 deferred alerting by
CALL_START already sent
Mar  7 00:57:32.158:SetDnCodec reassert defer_start for DN 14 chan 1
Mar  7 00:57:32.158:Delay media until loopback DN 43 is ready
Mar  7 00:57:32.158:SkinnyUpdateLoopbackCodec check for DN 14 chan 1 from DN 42 loopback
DN 43
Mar  7 00:57:32.158:SkinnyUpdateLoopbackCodec DN chain is 14 1, other=42, lb=43, far=-1 1,
final=43 1
Mar  7 00:57:32.158:SkinnyUpdateLoopbackCodec DN 14 chan 1 DN 43 chan 1 codec 4 match
Mar  7 00:57:32.162:SkinnyUpdateLoopbackState DN 42 state CONNECTED calledDn 14
Mar  7 00:57:32.162:Loopback DN 43 state RINGING
Mar  7 00:57:32.162:SkinnyUpdateLoopbackState DN 42 to DN 43 signal ANSWER
Mar  7 00:57:32.162:Loopback DN 42 calledDN 14 callingDn -1 G711Ulaw64k
Mar  7 00:57:32.162:Loopback DN 43 calledDN -1 callingDn -1 incoming G711Ulaw64k
Mar  7 00:57:32.162:ephone-5[7][SEP000DBDBEF37D]:refreshDisplayLine for line 1 DN 14 chan
1
Mar  7 00:57:32.162:dn_support_g729 true DN 43 chan 1 (loopback)
Mar  7 00:57:32.162:SetDnCodec Loopback DN 42 codec 4:G711Ulaw64k  vad 0 size 160
Mar  7 00:57:32.162:SkinnyUpdateLoopbackState DN 43 state CALL_START calledDn -1
Mar  7 00:57:32.162:Loopback DN 42 state CONNECTED
Mar  7 00:57:32.162:SkinnyUpdateLoopbackState DN 43 has defer_dn 14 chan 1 set
Mar  7 00:57:32.162:SkinnyUpdateLoopbackState DN 43 has defer_dn 14 chan 1:
 -invoke SkinnyOpenReceive
Mar  7 00:57:32.162:SkinnyUpdateLoopbackCodec check for DN 14 chan 1 from DN 42 loopback
DN 43
Mar  7 00:57:32.162:SkinnyUpdateLoopbackCodec DN chain is 14 1, other=42, lb=43, far=-1 1,
final=43 1
Mar  7 00:57:32.162:SkinnyUpdateLoopbackCodec DN 14 chan 1 DN 43 chan 1 codec 4 match
Mar  7 00:57:32.162:SkinnyUpdateLoopbackState DN 43 state CALL_START calledDn -1
Mar  7 00:57:32.162:Loopback DN 42 state CONNECTED
Mar  7 00:57:32.454:SkinnyGetDnAddrInfo DN 43 LOOPBACK
update media address to 10.0.0.6 25390 from DN 14
Mar  7 00:57:33.166:ephone-5[7]:DisplayCallInfo incoming call
.
.
.
```

When the called extension, 1514, goes back on-hook, the following debug output is displayed:

```
.
.
.
Mar  7 00:57:39.224:Loopback DN 42 disc reason 16 normal state CONNECTED
Mar  7 00:57:39.224:SkinnyUpdateLoopbackState DN 42 state CALL_END calledDn -1
Mar  7 00:57:39.224:Loopback DN 43 state CONNECTED
Mar  7 00:57:39.224:SkinnyUpdateLoopbackState DN 42 to DN 43 signal ONHOOK
Mar  7 00:57:39.236:SkinnyDnToneLoopback DN 42 state IDLE to DN 43 state IDLE
Mar  7 00:57:39.236:TONE OFF
Mar  7 00:57:39.236:SkinnyDnToneLoopback DN 43 state IDLE to DN 42 state IDLE
Mar  7 00:57:39.236:TONE OFF
```

Table 3 describes the significant fields shown in the display.

Table 3 debug ephone loopback Field Descriptions

Field	Description
Called Number	Original called number as presented to the incoming side of the loopback-dn.
Forward	Outgoing number that is expected to be dialed by the outgoing side of the loopback-dn pair.
PredictTarget Match	Extension (ephone-dn) that is anticipated by the loopback-dn to be the far-end termination for the call.
signal OFFHOOK	Indicates that the outgoing side of the loopback-dn pair is going off-hook prior to placing the outbound call leg.
Dial	Outbound side of the loopback-dn that is actually dialing the outbound call leg.
deferred alerting	Indicates that the alerting, or ringing, tone is returning to the original inbound call leg in response to the far-end ephone-dn state.
DN chain	Chain of ephone-dns that has been detected, starting from the far-end that terminates the call. Each entry in the chain indicates an ephone-dn tag and channel number. Entries appear in the following order, from left to right: • Ephone-dn tag and channel of the far-end call terminator (in this example, ephone-dn 14 is extension 1514). • other—Ephone-dn tag of the outgoing side of the loopback. • lb—Ephone-dn tag of the incoming side of the loopback. • far—Ephone-dn tag and channel of the far-end call originator, or -1 for a nonlocal number. • final—Ephone-dn tag for the originator of the call on the incoming side of the loopback. If the originator is not a local ephone-dn, this is set to -1. This number represents the final ephone-dn tag in the chain, looking toward the originator.
codec match	Indicates that there is no codec conflict between the two calls on either side of the loopback-dn.
GetDnAddrInfo	IP address of the IP phone at the final destination extension (ephone-dn), after resolving the chain of ephone-dns involved.
disc_reason	Disconnect cause code, in decimal. These are normal CC_CAUSE code values that are also used in call control API debugging. Common cause codes include the following: • 16—Normal disconnect. • 17—User busy. • 19—No answer. • 28—Invalid number.

Related Commands

Command	Description
debug ephone pak	Provides voice packet level debugging.
loopback-dn	Configures loopback-dn virtual loopback voice ports used to establish demarcation points for VoIP voice calls and supplementary services.
show ephone	Displays information about registered Cisco IP phones.
show ephone-dn loopback	Displays information for ephone-dns that have been set up for loopback calls.

debug ephone message

To enable message tracing between ephones, use the **debug ephone message** command in privileged EXEC mode. To disable debugging output, use the **no** form of this command.

debug ephone message [**detail**]

no debug ephone message

Syntax Description

detail	(Optional) Displays signaling connection control protocol (SCCP) messages sent and received between ephones in the Cisco Unified CallManager Express (Cisco Unified CME) system.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Modification
12.4(4)XC	This command was introduced.
12.4(9)T	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

The **debug ephone message** command enables message tracing between ephones.

The **debug ephone** command debugs all ephones associated with a Cisco Unified CME router.

You can enable or disable debugging on any number of ephones. To see the ephones that have debugging enabled, enter the **show ephone** command and look at the debug field in the output. When debugging is enabled for a ephone, the debug output is displayed for the directory numbers associated with the ephone.

Examples

The following is sample output for the **debug ephone message** command for ephones:

```
Router# debug ephone message

EPHONE skinny message debugging is enabled
*Jul 17 12:12:54.883: Received message from phone 7, SkinnyMessageID = StationKe
epAliveMessageID
*Jul 17 12:12:54.883: Sending  message to   phone 7, SkinnyMessageID = StationKe
epAliveAckMessageID
```

The following command disables ephone message debugging:

```
Router# no debug ephone message

EPHONE skinny message debugging is disabled
```

Related Commands

Command	Description
debug ephone alarm	Sets SkinnyStation alarm messages debugging for the ephone.
debug ephone detail	Sets detail debugging for the ephone.
debug ephone error	Sets error debugging for the ephone.
debug ephone mwi	Sets MWI debugging for the ephone.
debug ephone pak	Provides voice packet level debugging and displays the contents of one voice packet in every 1024 voice packets.
debug ephone raw	Provides raw low-level protocol debugging display for all SCCP messages.
debug ephone register	Sets registration debugging for the ephone.
debug ephone state	Sets state debugging for the ephone.
debug ephone statistics	Sets statistics debugging for the ephone.
debug ephone video	Sets video debugging for the ephone.
show debugging	Displays information about the types of debugging that are enabled for your router.
show ephone	Displays information about ephones.

debug ephone moh

To set debugging for music on hold (MOH), use the **debug ephone moh** command in privileged EXEC mode. To disable debugging, use the **no** form of this command.

debug ephone moh [**mac-address** *mac-address*]

no debug ephone moh [**mac-address** *mac-address*]

Syntax Description

mac-address *mac-address*	(Optional) Specifies the MAC address of a Cisco IP phone for debugging.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.2(2)XT	This command was introduced for Cisco IOS Telephony Services (now known as Cisco CallManager Express) Version 2.0 and Cisco Survivable Remote Site Telephony (SRST) Version 2.0 on the Cisco 1750, Cisco 1751, Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691.
12.3(4)T	This command was integrated into Cisco IOS Release 12.3(4)T.

Usage Guidelines

Always use the **no moh** command before modifying or replacing the MOH file in Flash memory.

When a configuration using the **multicast moh** command is used and the **debug ephone moh** command is enabled, if you delete or modify the MOH file in the router's Flash memory, the debug output can be excessive and can flood the console. The multicast MOH configuration should be removed before using the **no moh** command when the **debug ephone moh** command is enabled.

Examples

The following sample output shows MOH activity prior to the first MOH session. Note that if you enable multicast MOH, that counts as the first session.

```
Router# debug ephone moh

Mar  7 00:52:33.817:MOH AU file
Mar  7 00:52:33.817:skinny_open_moh_play set type to 3
Mar  7 00:52:33.825: 2E73 6E64 0000 0018 0007 3CCA 0000 0001
Mar  7 00:52:33.825: 0000 1F40 0000 0001 FFFF FFFF FFFF FFFF
Mar  7 00:52:33.825: FFFF FFFF FFFF FFFF FFFF FFFF FFFF FFFF
Mar  7 00:52:33.825: FFFF FFFF FFFF FFFF FFFF FFFF FFFF FFFF
Mar  7 00:52:33.825: FFFF FFFF FFFF FFFF FFFF FFFF FFFF FFFF
```

```
Mar  7 00:52:33.825: FFFF FFFF FFFF FFFF FFFF FFFF FFFF FFFF
Mar  7 00:52:33.825: FFFF FFFF FFFF FFFF FFFF FFFF FFFF FFFF
Mar  7 00:52:33.825: FFFF FFFF FFFF FFFF FFFF FFFF FFFF FFFF
Mar  7 00:52:33.825: FFFF FFFF FFFF FFFF FFFF FFFF FFFF FFFF
Mar  7 00:52:33.825: FFFF FFFF FFFF FFFF FFFF FFFF FFFF FFFF
Mar  7 00:52:33.825: FFFF FFFF FFFF FFFF FFFF FFFF FFFF FFFF
Mar  7 00:52:33.825: FFFF FFFF FFFF FFFF FFFF FFFF FFFF FFFF
Mar  7 00:52:33.825: FFFF FFFF FFFF FFFF FFFF FFFF FFFF FFFF
Mar  7 00:52:33.825: FFFF FFFF FFFF FFFF FFFF FFFF FFFF FFFF
Mar  7 00:52:33.825: FFFF FFFF FFFF FFFF FFFF FFFF FFFF FFFF
Mar  7 00:52:33.825: FFFF FFFF FFFF FFFF FFFF FFFF FFFF FFFF
Mar  7 00:52:33.825:
Mar  7 00:52:33.825:AU file processing Found .snd
Mar  7 00:52:33.825:AU file data start at 24 end at 474338
Mar  7 00:52:33.825:AU file codec Media_Payload_G711Ulaw64k
Mar  7 00:52:33.825:MOH read file header type AU start 24 end 474338
Mar  7 00:52:33.825:MOH pre-read block 0 at write-offset 0 from 24
Mar  7 00:52:33.833:MOH pre-read block 1 at write-offset 8000 from 8024
Mar  7 00:52:33.845:Starting read server with play-offset 0 write-offset 16000
```

Table 4 describes the significant fields shown in the display.

Table 4 debug ephone moh Field Descriptions

Field	Description
type	0—invalid 1—raw file 2—wave format file (.wav) 3—AU format (.au) 4—live feed
AU file processing Found .snd	A .snd header was located in the AU file.
AU file data start at, end at	Data start and end file offset within the MOH file, as indicated by the file header.
read file header type	File format found (AU, WAVE, or RAW).
pre-read block, write-offset	Location in the internal MOH buffer to which data is being written, and location from which that data was read in the file.
play-offset, write-offset	Indicates the relative positioning of MOH file read-ahead buffering. Data is normally written from a Flash file into the internal circular buffer, ahead of the location from which data is being played or output.

Related Commands

Command	Description
moh (telephony-service)	Generates an audio stream from a file for MOH in a Cisco CME system.
multicast moh	Uses the MOH audio stream as a multicast source in a Cisco CME system.

debug ephone mwi

To set message waiting indication (MWI) debugging for the Cisco IOS Telephony Service router, use the **debug ephone mwi** command in privileged EXEC mode. To disable debugging output, use the **no** form of this command.

debug ephone mwi

no debug ephone mwi

Syntax Description

This command has no arguments or keywords.

Defaults

No default behavior or values

Command Modes

Privileged EXEC

Command History

Release	Modification
12.2(2)XT	This command was introduced on the following platforms: Cisco 1750, Cisco 1751, Cisco 2600 series and Cisco 3600 series multiservice routers; and Cisco IAD2420 series Integrated Access Devices (IADs).
12.2(8)T	This command was implemented on the Cisco 3725 and Cisco 3745 routers.
12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
12.2(11)T	This command was implemented on the Cisco 1760 routers.

Usage Guidelines

The **debug ephone mwi** command sets message waiting indication debugging for the Cisco IOS Telephony Service router. Because the MWI protocol activity is not specific to any individual Cisco IP phone, setting the MAC address keyword qualifier for this command is not useful.

Note Unlike the other related **debug ephone** commands, the **mac-address** keyword does not help debug a particular Cisco IP phone.

Examples

The following is sample output of the message waiting indication status for the Cisco IOS Telephony Service router:

```
Router# debug ephone mwi
```

Related Commands

Command	Description
debug ephone alarm	Sets SkinnyStation alarm messages debugging for the Cisco IP phone.
debug ephone detail	Sets detail debugging for the Cisco IP phone.
debug ephone error	Sets error debugging for the Cisco IP phone.
debug ephone keepalive	Sets keepalive debugging for the Cisco IP phone.
debug ephone pak	Provides voice packet level debugging and prints the contents of one voice packet in every 1024 voice packets.
debug ephone raw	Provides raw low-level protocol debugging display for all SCCP messages.
debug ephone register	Sets registration debugging for the Cisco IP phone.
debug ephone state	Sets state debugging for the Cisco IP phone.
debug ephone statistics	Sets statistics debugging for the Cisco IP phone.
show debugging	Displays information about the types of debugging that are enabled for your router.

debug ephone pak

To provide voice packet level debugging and to print the contents of one voice packet in every 1024 voice packets, use the **debug ephone pak** command in privileged EXEC mode. To disable debugging output, use the **no** form of this command.

debug ephone pak [**mac-address** *mac-address*]

no debug ephone pak [**mac-address** *mac-address*]

Syntax Description

mac-address	(Optional) Defines the MAC address of the Cisco IP phone.
mac-address	(Optional) Specifies the MAC address of the Cisco IP phone.

Defaults

No default behavior or values

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

The **debug ephone pak** command provides voice packet level debugging and prints the contents of one voice packet in every 1024 voice packets.

If the **mac-address** keyword is not used, the **debug ephone pak** command debugs all Cisco IP phones that are registered to the router. You can remove debugging for the Cisco IP phones that you do not want to debug by using the **mac-address** keyword with the **no** form of this command.

You can enable or disable debugging on any number of Cisco IP phones. To see the Cisco IP phones that have debugging enabled, enter the **show ephone** command and look at the debug field in the output. When debugging is enabled for a Cisco IP phone, the debug output is displayed for the directory numbers associated with the Cisco IP phone.

Examples

The following is sample output of packet debugging for the Cisco IP phone with MAC address 0030.94c3.8724:

```
Router# debug ephone pak mac-address 0030.94c3.8724

EPHONE packet debugging is enabled for phone 0030.94c3.8724

01:29:14: ***ph_xmit_ephone DN 3 tx_pkts 5770 dest=10.2.1.1 orig len=32
 pakcopy=0 discards 27 ip_enctype 0 0 last discard: unsupported payload type
01:29:14: to_skinny_duration 130210 offset -30 last -40 seq 0 adj 0
01:29:14: IP:   45B8 003C 0866 0000 3F11 3F90 2800 0001 0A02 0101
01:29:14: TTL 63 TOS B8 prec 5
01:29:14: UDP:  07D0 6266 0028 0000
01:29:14: sport 2000 dport 25190 length 40 checksum 0
01:29:14: RTP:  8012 16AF 9170 6409 0E9F 0001
01:29:14: is_rtp:1 is_frf11:0 vlen:0 delta_t:160 vofr1:0 vofr2:0
scodec:11 rtp_bits:8012 rtp_codec:18 last_bad_payload 19
01:29:14: vencap FAILED
01:29:14: PROCESS SWITCH
01:29:15: %SYS-5-CONFIG_I: Configured from console by console
01:29:34: ***SkinnyPktIp DN 3 10.2.1.1 to 40.0.0.1 pkts 4880 FAST sw
01:29:34: from_skinny_duration 150910
01:29:34: nw 3BBC2A8 addr 3BBC2A4 mac 3BBC2A4 dg 3BBC2C4 dgs 2A
01:29:34: MAC:  1841 0800
01:29:34: IP:   45B8 0046 682E 0000 3E11 E0BD 0A02 0101 2800 0001
01:29:34: TTL 62 TOS B8 prec 5
01:29:34: UDP:  6266 07D0 0032 0000
01:29:34: sport 25190 dport 2000 length 50 checksum 0
01:29:34: RTP:  8012 55FF 0057 8870 3AF4 C394
01:29:34: RTP: rtp_bits 8012 seq 55FF ts 578870 ssrc 3AF4C394
01:29:34: PAYLOAD:
01:29:34:       1409 37C9 54DE 449C 3B42 0446 3AAB 182E
01:29:34:       56BC 5184 58E5 56D3 13BE 44A7 B8C4
01:29:34:
01:29:37: ***ph_xmit_ephone DN 3 tx_pkts 6790 dest=10.2.1.1 orig len=32
 pakcopy=0 discards 31 ip_enctype 0 0 last discard: unsupported payload type
01:29:37: to_skinny_duration 153870 offset -150 last -40 seq 0 adj 0
01:29:37: IP:   45B8 003C 0875 0000 3F11 3F81 2800 0001 0A02 0101
01:29:37: TTL 63 TOS B8 prec 5
01:29:37: UDP:  07D0 6266 0028 0000
01:29:37: sport 2000 dport 25190 length 40 checksum 0
01:29:37: RTP:  8012 1AAF 9173 4769 0E9F 0001
01:29:37: is_rtp:1 is_frf11:0 vlen:0 delta_t:160 vofr1:0 vofr2:0
```

Related Commands

Command	Description
debug ephone alarm	Sets SkinnyStation alarm messages debugging for the Cisco IP phone.
debug ephone detail	Sets detail debugging for the Cisco IP phone.
debug ephone error	Sets error debugging for the Cisco IP phone.
debug ephone keepalive	Sets keepalive debugging for the Cisco IP phone.
debug ephone loopback	Sets MWI debugging for the Cisco IP phone.
debug ephone raw	Provides raw low-level protocol debugging display for all SCCP messages.
debug ephone register	Sets registration debugging for the Cisco IP phone.
debug ephone state	Sets state debugging for the Cisco IP phone.

Command	Description
debug ephone statistics	Sets statistics debugging for the Cisco IP phone.
show debugging	Displays information about the types of debugging that are enabled for your router.

debug ephone qov

To display quality of voice (QOV) statistics for calls when preset limits are exceeded, use the **debug ephone qov** command in privileged EXEC mode. To disable debugging, use the **no** form of this command.

debug ephone qov [**mac-address** *mac-address*]

no debug ephone qov [**mac-address** *mac-address*]

Syntax Description

mac-address *mac-address*	(Optional) Specifies the MAC address of a Cisco IP phone for debugging.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.2(15)ZJ2	This command was introduced for Cisco CallManager Express 3.0 and Cisco Survivable Remote Site Telephony (SRST) Version 3.0.
12.3(4)T	This command was integrated into Cisco IOS Release 12.3(4)T.

Usage Guidelines

Once enabled, the **debug ephone qov** command produces output only when the QOV statistics reported by phones exceed preset limits. Phones are polled every few seconds for QOV statistics on VoIP calls only, not on local PSTN calls. An output report is produced when limits are surpassed for either or both of the following:

- Lost packets—A report is triggered when two adjacent QOV samples show an increase of four or more lost packets between samples. The report is triggered by an increase of lost packets in a short period of time, not by the total number of lost packets.
- Jitter and latency—A report is triggered when either jitter or latency exceeds 100 milliseconds.

To receive a QOV report at the end of each call regardless of whether the QOV limits have been exceeded, enable the **debug ephone alarm** command in addition to the **debug ephone qov** command.

The **debug ephone statistics** command displays the raw statistics that are polled from phones and used to generate QOV reports.

Examples

The following sample output describes QOV statistics for a call on ephone 5:

```
Router# debug ephone qov

Mar  7 00:54:57.329:ephone-5[7]:QOV DN 14 chan 1 (1514) ref 4 called=1514 calling=8101
Mar  7 00:54:57.329:ephone-5[7][SEP000DBDBEF37D]:Lost 91 Jitter 0 Latency 0
Mar  7 00:54:57.329:ephone-5[7][SEP000DBDBEF37D]:previous Lost 0 Jitter 0 Latency 0
Mar  7 00:54:57.329:ephone-5[7][SEP000DBDBEF37D]:Router sent 1153 pkts, current phone got
1141
received by all (shared) phones 0
Mar  7 00:54:57.329:ephone-5[7]:worst jitter 0 worst latency 0
Mar  7 00:54:57.329:ephone-5[7]:Current phone sent 1233 packets

Mar  7 00:54:57.329:ephone-5[7]:Signal Level to phone 3408 (-15 dB) peak 3516 (-15 dB)
```

Table 5 describes the significant fields shown in the display.

Table 5 debug ephone qov Field Descriptions

Field	Description
Lost	Number of lost packets reported by the IP phone.
Jitter, Latency	The most recent jitter and latency parameters reported by the IP phone.
previous Lost, Jitter, Latency	Values from the previous QOV statistics report that were used as the comparison points against which the current statistics triggered generation of the current report.
Router sent pkts	Number of packets sent by the router to the IP phone. This number is the total for the entire call, even if the call is moved from one phone to another during a call, which can happen with shared lines.
current phone got	Number of packets received by the phone currently terminating the call. This number is the total for the entire call, even if the call is moved from one phone to another during a call, which can happen with shared lines.
worst jitter, worst latency	Highest value reported by the phone during the call.
Current phone sent packets	Number of packets that the current phone claims it sent during the call.
Signal Level to phone	Signal level seen in G.711 voice packet data prior to the sending of the most recent voice packet to the phone. The first number is the raw sample value, converted from G.711 to 16-bit linear format and left-justified. The number in parentheses is the value in decibels (dB), assuming that 32,767 is about +3 dB. **Note** This value is meaningful only if the call uses a G.711 codec.

Related Commands

Command	Description
debug ephone alarm	Displays alarm messages for IP phones.
debug ephone statistics	Displays call statistics for IP phones.

debug ephone raw

To provide raw low-level protocol debugging display for all Skinny Client Control Protocol (SCCP) messages, use the **debug ephone raw** command in privileged EXEC mode. To disable debugging output, use the **no** form of this command.

debug ephone raw [**mac-address** *mac-address*]

no debug ephone raw [**mac-address** *mac-address*]

Syntax Description

mac-address	(Optional) Defines the MAC address of the Cisco IP phone.
mac-address	(Optional) Specifies the MAC address of the Cisco IP phone.

Defaults

No default behavior or values

Command Modes

Privileged EXEC

Command History

Release	Modification
12.1(5)YD	This command was introduced on the following platforms: Cisco 2600 series and Cisco 3600 series multiservice routers, and Cisco IAD2420 series Integrated Access Devices (IADs).
12.2(2)XT	This command was implemented on the Cisco 1750 and Cisco 1751 multiservice routers.
12.2(8)T	This command was implemented on the Cisco 3725 and Cisco 3745 routers.
12.2(8)T1	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
12.2(11)T	This command was implemented on the Cisco 1760 routers.

Usage Guidelines

The **debug ephone raw** command provides raw low-level protocol debug display for all SCCP messages. The debug display provides byte level display of Skinny TCP socket messages.

If the **mac-address** keyword is not used, the **debug ephone raw** command debugs all Cisco IP phones that are registered to the router. You can remove debugging for the Cisco IP phones that you do not want to debug by using the **mac-address** keyword with the **no** form of this command.

You can enable or disable debugging on any number of Cisco IP phones. To see the Cisco IP phones that have debugging enabled, enter the **show ephone** command and look at the debug field in the output. When debugging is enabled for a Cisco IP phone, the debug output is displayed for the directory numbers associated with the Cisco IP phone.

Examples

The following is sample output of raw protocol debugging for the Cisco IP phone with MAC address 0030.94c3.E1A8:

```
Router# debug ephone raw mac-address 0030.94c3.E1A8

EPHONE raw protocol debugging is enabled for phone 0030.94C3.E1A8

1d05h: skinny socket received 4 bytes on socket [1]
0  0  0  0
1d05h:
1d05h: SkinnyMessageID = 0
1d05h: skinny send 4 bytes
4  0  0  0  0  0  0  0  0  1  0  0
1d05h: socket [1] sent 12 bytes OK (incl hdr) for ephone-(1)

1d06h: skinny socket received 4 bytes on socket [1]
0  0  0  0
1d06h:
1d06h: SkinnyMessageID = 0
1d06h: skinny send 4 bytes
4  0  0  0  0  0  0  0  0  1  0  0
1d06h: socket [1] sent 12 bytes OK (incl hdr) for ephone-(1)
```

Related Commands

Command	Description
debug ephone alarm	Sets SkinnyStation alarm messages debugging for the Cisco IP phone.
debug ephone detail	Sets detail debugging for the Cisco IP phone.
debug ephone error	Sets error debugging for the Cisco IP phone.
debug ephone keepalive	Sets keepalive debugging for the Cisco IP phone.
debug ephone loopback	Sets MWI debugging for the Cisco IP phone.
debug ephone pak	Provides voice packet level debugging and prints the contents of one voice packet in every 1024 voice packets.
debug ephone register	Sets registration debugging for the Cisco IP phone.
debug ephone state	Sets state debugging for the Cisco IP phone.
debug ephone statistics	Sets statistics debugging for the Cisco IP phone.
show debugging	Displays information about the types of debugging that are enabled for your router.

debug ephone register

To set registration debugging for the Cisco IP phone, use the **debug ephone register** command in privileged EXEC mode. To disable debugging output, use the **no** form of this command.

debug ephone register [**mac-address** *mac-address*]

no debug ephone register [**mac-address** *mac-address*]

Syntax Description

mac-address	(Optional) Defines the MAC address of the Cisco IP phone.
mac-address	(Optional) Specifies the MAC address of the Cisco IP phone.

Defaults

No default behavior or values

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

The **debug ephone register** command sets registration debugging for the Cisco IP phones.

If the **mac-address** keyword is not used, the **debug ephone register** command debugs all Cisco IP phones that are registered to the router. You can remove debugging for the Cisco IP phones that you do not want to debug by using the **mac-address** keyword with the **no** form of this command.

You can enable or disable debugging on any number of Cisco IP phones. To see the Cisco IP phones that have debugging enabled, enter the **show ephone** command and look at the debug field in the output. When debugging is enabled for a Cisco IP phone, the debug output is displayed for the directory numbers associated with the Cisco IP phone.

Examples

The following is sample output of registration debugging for the Cisco IP phone with MAC address 0030.94c3.8724:

```
Router# debug ephone register mac-address 0030.94c3.8724

Ephone registration debugging is enabled

1d06h: New Skinny socket accepted [1] (2 active)
1d06h: sin_family 2, sin_port 50778, in_addr 10.1.0.21
1d06h: skinny_add_socket 1 10.1.0.21 50778
1d06h: ephone-(1)[1] StationRegisterMessage (2/3/12) from 10.1.0.21
1d06h: ephone-(1)[1] Register StationIdentifier DeviceName SEP003094C3E1A8
1d06h: ephone-(1)[1] StationIdentifier Instance 1    deviceType 7
1d06h: ephone-1[-1]:stationIpAddr 10.1.0.21
1d06h: ephone-1[-1]:maxStreams 0
1d06h: ephone-(1) Allow any Skinny Server IP address 10.1.0.6
.
.
.
1d06h: ephone-1[1]:RegisterAck sent to ephone 1: keepalive period 30
.
```

Related Commands

Command	Description
debug ephone alarm	Sets SkinnyStation alarm messages debugging for the Cisco IP phone.
debug ephone detail	Sets detail debugging for the Cisco IP phone.
debug ephone error	Sets error debugging for the Cisco IP phone.
debug ephone keepalive	Sets keepalive debugging for the Cisco IP phone.
debug ephone loopback	Sets MWI debugging for the Cisco IP phone.
debug ephone pak	Provides voice packet level debugging and prints the contents of one voice packet in every 1024 voice packets.
debug ephone raw	Provides raw low-level protocol debugging display for all SCCP messages.
debug ephone state	Sets state debugging for the Cisco IP phone.
debug ephone statistics	Sets statistics debugging for the Cisco IP phone.
show debugging	Displays information about the types of debugging that are enabled for your router.

debug ephone sccp-state

To set debugging for the SCCP call state, use the **debug ephone sccp-state** command in privileged EXEC mode. To disable debugging output, use the **no** form of this command.

debug ephone sccp-state [**mac-address** *mac-address*]

no debug ephone sccp-state [**mac-address** *mac-address*]

Syntax Description

mac-address *mac-address*	(Optional) Specifies the MAC address of a phone.

Command Default

Debugging is not enabled for SCCP state.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Modification
12.4(4)XC	This command was introduced.
12.4(9)T	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used with Cisco Unified CallManager Express (Cisco Unified CME).

This command outputs only the debug messages that correspond to SCCP messages sent to IP phones to indicate the SCCP phone call state, such as RingIn, OffHook, Connected, and OnHook. These debug messages are also included in the output for the **debug ephone detail** command among other information.

Examples

The following example sets SCCP state debugging for one Cisco Unified CME phone with the MAC address of 678B.AEF9.DAB5.

```
Router# debug ephone sccp-state mac-address 678B.AEF9.DAB5

EPHONE SCCP state message debugging is enabled
  for ephones 000B.BEF9.DFB5

*Mar  8 06:38:45.863: %ISDN-6-CONNECT: Interface Serial2/0/0:22 is now connected to 
4085254871 unknown
*Mar  8 06:38:50.487: ephone-2[13]:SetCallState line 4 DN 60(60) chan 1 ref 100 TsRingIn
*Mar  8 06:38:52.399: ephone-2[13]:SetCallState line 4 DN 60(-1) chan 1 ref 100 TsOffHook
*Mar  8 06:38:52.399: ephone-2[13]:SetCallState line 4 DN 60(-1) chan 1 ref 100 
TsConnected
*Mar  8 06:38:58.415: %ISDN-6-CONNECT: Interface Serial2/0/0:22 is now connected to 
4085254871 unknown
*Mar  8 06:38:59.963: ephone-2[13]:SetCallState line 4 DN 60(-1) chan 1 ref 100 TsOnHook
*Mar  8 06:38:59.975: %ISDN-6-DISCONNECT: Interface Serial2/0/0:22 disconnected from 
4085254871 , call lasted 7 seconds
```

Related Commands

Command	Description
debug ephone detail	Sets detail debugging for one or all Cisco Unified IP phones.

debug ephone state

To set state debugging for the Cisco IP phone, use the **debug ephone state** command in privileged EXEC mode. To disable debugging output, use the **no** form of this command.

debug ephone state [**mac-address** *mac-address*]

no debug ephone state [**mac-address** *mac-address*]

Syntax Description

mac-address	(Optional) Defines the MAC address of the Cisco IP phone.
mac-address	(Optional) Specifies the MAC address of the Cisco IP phone.

Defaults

No default behavior or values

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

The **debug ephone state** command sets state debugging for the Cisco IP phones.

If the **mac-address** keyword is not used, the **debug ephone state** command debugs all Cisco IP phones that are registered to the router. You can remove debugging for the Cisco IP phones that you do not want to debug by using the **mac-address** keyword with the **no** form of this command.

You can enable or disable debugging on any number of Cisco IP phones. To see the Cisco IP phones that have debugging enabled, enter the **show ephone** command and look at the debug field in the output. When debugging is enabled for a Cisco IP phone, the debug output is displayed for the directory numbers associated with the Cisco IP phone.

Examples

The following is sample output of state debugging for the Cisco IP phone with MAC address 0030.94c3.E1A8:

```
Router# debug ephone state mac-address 0030.94c3.E1A8

EPHONE state debugging is enabled for phone 0030.94C3.E1A8

1d06h: ephone-1[1]:OFFHOOK
1d06h: ephone-1[1]:SIEZE on activeline 0
1d06h: ephone-1[1]:SetCallState line 1 DN 1 TsOffHook
1d06h: ephone-1[1]:Skinny-to-Skinny call DN 1 to DN 2 instance 1
1d06h: ephone-1[1]:SetCallState line 1 DN 1 TsRingOut
1d06h: ephone-1[1]:Call Info DN 1 line 1 ref 158 called 5002 calling 5001
1d06h: ephone-1[1]: Jane calling
1d06h: ephone-1[1]: Jill
1d06h: ephone-1[1]:SetCallState line 3 DN 2 TsRingIn
1d06h: ephone-1[1]:Call Info DN 2 line 3 ref 159 called 5002 calling 5001
1d06h: ephone-1[1]: Jane calling
1d06h: ephone-1[1]: Jill
1d06h: ephone-1[1]:SetCallState line 3 DN 2 TsCallRemoteMultiline
1d06h: ephone-1[1]:SetCallState line 1 DN 1 TsConnected
1d06h: ephone-1[1]:OpenReceive DN 1 codec 4:G711Ulaw64k  duration 10 ms bytes 80
1d06h: ephone-1[1]:OpenReceiveChannelAck 1.2.172.21 port=24010
1d06h: ephone-1[1]:StartMedia 1.2.172.22 port=24612
1d06h: DN 1 codec 4:G711Ulaw64k duration 10 ms bytes 80
1d06h: ephone-1[1]:CloseReceive
1d06h: ephone-1[1]:StopMedia
1d06h: ephone-1[1]:SetCallState line 3 DN 2 TsOnHook
1d06h: ephone-1[1]:SetCallState line 1 DN 1 TsOnHook
1d06h: ephone-1[1]:SpeakerPhoneOnHook
1d06h: ephone-1[1]:ONHOOK
1d06h: ephone-1[1]:SpeakerPhoneOnHook
1d06h: SkinnyReportDnState DN 1 ONHOOK
```

Related Commands

Command	Description
debug ephone alarm	Sets SkinnyStation alarm messages debugging for the Cisco IP phone.
debug ephone detail	Sets detail debugging for the Cisco IP phone.
debug ephone error	Sets error debugging for the Cisco IP phone.
debug ephone keepalive	Sets keepalive debugging for the Cisco IP phone.
debug ephone loopback	Sets MWI debugging for the Cisco IP phone.
debug ephone pak	Provides voice packet level debugging and prints the contents of one voice packet in every 1024 voice packets.
debug ephone raw	Provides raw low-level protocol debugging display for all SCCP messages.
debug ephone register	Sets registration debugging for the Cisco IP phone.
debug ephone	Sets statistics debugging for the Cisco IP phone.
show debugging	Displays information about the types of debugging that are enabled for your router.

debug ephone statistics

To set call statistics debugging for the Cisco IP phone, use the **debug ephone statistics** command in privileged EXEC mode. To disable debugging output, use the **no** form of this command.

debug ephone statistics [**mac-address** *mac-address*]

no debug ephone statistics [**mac-address** *mac-address*]

Syntax Description

mac-address	(Optional) Defines the MAC address of the Cisco IP phone.
mac-address	(Optional) Specifies the MAC address of the Cisco IP phone.

Defaults

No default behavior or values

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

The **debug ephone statistics** command provides a debug monitor display of the periodic messages from the Cisco IP phone to the router. These include transmit-and-receive packet counts and an estimate of drop packets. The call statistics can also be displayed for live calls using the **show ephone** command.

If the **mac-address** keyword is not used, the **debug ephone statistics** command debugs all Cisco IP phones that are registered to the router. You can remove debugging for the Cisco IP phones that you do not want to debug by using the **mac-address** keyword with the **no** form of this command.

You can enable or disable debugging on any number of Cisco IP phones. To see the Cisco IP phones that have debugging enabled, enter the **show ephone** command and look at the debug field in the output. When debugging is enabled for a Cisco IP phone, the debug output is displayed for the directory numbers associated with the Cisco IP phone.

Examples

The following is sample output of statistics debugging for the Cisco IP phone with MAC address 0030.94C3.E1A8:

```
Router# debug ephone statistics mac-address 0030.94C3.E1A8

EPHONE statistics debugging is enabled for phone 0030.94C3.E1A8

1d06h: Clear Call Stats for DN 1 call ref 162
1d06h: Clear Call Stats for DN 1 call ref 162
1d06h: Clear Call Stats for DN 1 call ref 162
1d06h: Clear Call Stats for DN 2 call ref 163
1d06h: ephone-1[1]:GetCallStats line 1 ref 162 DN 1: 5001
1d06h: ephone-1[1]:Call Stats for line 1 DN 1 5001 ref 162
1d06h: ephone-1[1]:TX Pkts 0 bytes 0 RX Pkts 0 bytes 0
1d06h: ephone-1[1]:Pkts lost 4504384 jitter 0 latency 0
1d06h: ephone-1[1]:Src 0.0.0.0 0 Dst 0.0.0.0 0 bytes 80 vad 0 G711Ulaw64k
1d06h: ephone-1[1]:GetCallStats line 1 ref 162 DN 1: 5001
1d06h: STATS: DN 1 Packets Sent 0
1d06h: STATS: DN 2 Packets Sent 0
1d06h: ephone-1[1]:Call Stats found DN -1 from Call Ref 162
1d06h: ephone-1[1]:Call Stats for line 0 DN -1 5001 ref 162
1d06h: ephone-1[1]:TX Pkts 275 bytes 25300 RX Pkts 275 bytes 25300
1d06h: ephone-1[1]:Pkts lost 0 jitter 0 latency 0
```

Related Commands

Command	Description
debug ephone alarm	Sets SkinnyStation alarm messages debugging for the Cisco IP phone.
debug ephone detail	Sets detail debugging for the Cisco IP phone.
debug ephone error	Sets error debugging for the Cisco IP phone.
debug ephone keepalive	Sets keepalive debugging for the Cisco IP phone.
debug ephone loopback	Sets MWI debugging for the Cisco IP phone.
debug ephone pak	Provides voice packet level debugging and prints the contents of one voice packet in every 1024 voice packets.
debug ephone raw	Provides raw low-level protocol debugging display for all SCCP messages.
debug ephone register	Sets registration debugging for the Cisco IP phone.
debug ephone state	Sets state debugging for the Cisco IP phone.
show debugging	Displays information about the types of debugging that are enabled for your router.

debug ephone video

To set video debugging for ephones, use the **debug ephone video** command in privileged EXEC mode. To disable debugging output, use the **no** form of this command.

debug ephone video

no debug ephone video

Syntax Description

This command has no arguments or keywords.

Command Default

Debugging is disabled for ephone video.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Modification
12.4(4)XC	This command was introduced.
12.4(9)T	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

The **debug ephone video** command sets ephone video traces, which provide information about different video states for the call, including video capabilities selection, start, and stop.

The **debug ephone** command debugs all ephones that are registered to the Cisco Unified CallManager Express (Cisco Unified CME) system.

You can enable or disable debugging on any number of ephones. To see the ephones that have debugging enabled, enter the **show ephone** command and look at the debug field in the output. When debugging is enabled for a ephone, the debug output is displayed for the directory numbers associated with the ephone.

Examples

The following is sample output for the **debug ephone video** command for ephones:

```
Router# debug ephone video

*Mar 13 16:10:02.703: SkinnyVideoCodecMatch_Caps2Caps: match capability: tx_idxcap = 4,
tx_idxpref = 3,
*Mar 13 16:10:02.703:                rx_idxcap = 0, rx_idxpref = 0, videoBitRate = 7040
tx_mpi = 1
*Mar 13 16:10:04.711: ephone-19[1][SEPFFFA00000019]:checkToOpenMultiMedia: dn=19, chan=1
*Mar 13 16:10:04.711: ephone-19[1]:skinnyDP[19].s2s = 0
*Mar 13 16:10:04.711: ephone-19[1]:s2s is not set - hence not video capable
*Mar 13 16:10:04.719: ephone-19[1][SEPFFFA00000019]:SkinnyStartMultiMediaTransmission:
chan 1 dn 19
*Mar 13 16:10:04.723: ephone-19[1]:Accept OLC and open multimedia channel
*Mar 13 16:10:04.723: ephone-19[1][SEPFFFA00000019]:SkinnyOpenMultiMediaReceiveChannel: dn
19 chan 1
```

```
*Mar 13 16:10:04.967: ephone-19[1][SEPFFFA00000019]:fStationOpenReceiveChannelAckMessage:
MEDIA_DN 19 MEDIA_CHAN 1
*Mar 13 16:10:04.967: ephone-19[1]:fStationOpenMultiMediaReceiveChannelAckMessage:
*Mar 13 16:10:04.967: ephone-19[1]:Other_dn == -1
sk3745-2#
*Mar 13 16:10:14.787: ephone-19[1]:SkinnyStopMedia: Stop Multimedia
*Mar 13 16:10:14.787: ephone-19[1][SEPFFFA00000019]:SkinnyCloseMultiMediaReceiveChannel:
passThruPartyID = 0, callReference = 23
*Mar 13 16:10:14.787: ephone-19[1]:SkinnyStopMultiMediaTransmission: line 1 chan 1 dn 19
```

Related Commands

Command	Description
debug ephone alarm	Sets SkinnyStation alarm messages debugging for the ephone.
debug ephone detail	Sets detail debugging for the ephone.
debug ephone error	Sets error debugging for the ephone.
debug ephone message	Sets message debugging for the ephone.
debug ephone mwi	Sets MWI debugging for the ephone.
debug ephone pak	Provides voice packet level debugging and displays the contents of one voice packet in every 1024 voice packets.
debug ephone raw	Provides raw low-level protocol debugging display for all SCCP messages.
debug ephone register	Sets registration debugging for the ephone.
debug ephone state	Sets state debugging for the ephone.
debug ephone statistics	Sets statistics debugging for the ephone.
show debugging	Displays information about the types of debugging that are enabled for your router.
show ephone	Displays information about registered ephones.

debug ephone vm-integration

To display pattern manipulation information used for integration with voice-mail applications, use the **debug ephone vm-integration** command in privileged EXEC mode. To disable debugging output, use the **no** form of this command.

debug ephone vm-integration [**mac-address** *mac-address*]

no debug ephone vm-integration [**mac-address** *mac-address*]

Syntax Description

mac-address *mac-address*	(Optional) Specifies the MAC address of a Cisco IP phone for debugging.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.3(7)T	This command was introduced.

Usage Guidelines

This command displays the voice-mail integration patterns that were created using the **pattern** commands in vm-integration configuration mode. The patterns are used to forward calls to a voice-mail number that is set with the **voicemail** command.

If you do not specify the **mac-address** keyword, the **debug ephone vm-integration** command debugs all Cisco IP phones that are registered to the router. To remove debugging for Cisco IP phones, enter the **no** form of this command with the **mac-address** keyword.

Examples

The following sample output shows information for the vm-integration tokens that have been defined:

```
Router# debug ephone vm-integration
```

```
*Jul 23 15:38:03.294:ephone-3[3]:StimulusMessage 15 (1) From ephone 2
*Jul 23 15:38:03.294:ephone-3[3]:Voicemail access number pattern check
*Jul 23 15:38:03.294:SkinnyGetCallState for DN 3 chan 1 IDLE
*Jul 23 15:38:03.294:called DN -1 chan 1, calling DN -1 chan 1 phone -1 s2s:0
*Jul 23 15:38:03.294:dn number for dn 3 is 19003
*Jul 23 15:38:03.294:Updated number for token 1 is 19003
*Jul 23 15:38:03.294:CDN number for dn 3 is
*Jul 23 15:38:03.294:Updated number for token 2 is
*Jul 23 15:38:03.294:Updated number for token 0 is
*Jul 23 15:38:03.294:Update is 219003*
*Jul 23 15:38:03.294:New Voicemail number is 19101219003*
```

Table 6 describes the significant fields shown in the display.

Table 6 debug ephone vm-integration Field Descriptions

Field	Description
token 0	First token that was defined in the pattern.
token 1	Second token that was defined in the pattern.
token 2	Third token that was defined in the pattern.

Related Commands

Command	Description
pattern direct	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when a user presses the Messages button on a phone.
pattern ext-to-ext busy	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an internal extension reaches a busy extension and the call is forwarded to voice mail.
pattern ext-to-ext no-answer	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an internal extension fails to connect to an extension and the call is forwarded to voice mail.
pattern trunk-to-ext busy	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an external trunk call reaches a busy extension and the call is forwarded to voice mail.
pattern trunk-to-ext no-answer	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when an external trunk call reaches an unanswered extension and the call is forwarded to voice mail.
vm-integration	Enters voice-mail integration configuration mode and enables voice-mail integration with DTMF and analog voice-mail systems.
voicemail	Defines the telephone number that is speed-dialed when the Messages button on a Cisco IP phone is pressed.

debug mwi relay errors

To debug message waiting indication (MWI) relay errors, use the **debug mwi relay errors** command in privileged EXEC mode. To disable debugging output, use the **no** form of this command.

debug mwi relay errors

no debug mwi relay errors

Syntax Description

This command has no arguments or keywords.

Defaults

No default behavior or values

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

The **debug mwi relay errors** command provides a debug monitor display of any error messages, when MWI Relay Server (Cisco IOS Telephony Server) is trying to do MWI Relay to extensions on remote Cisco IOS Telephony Service (ITS).

Examples

The following examples show errors when MWI Relay Server tries to do an MWI Relay to extension 7004, but location of 7004 is not known to the MWI Relay Server:

```
Router# debug mwi relay errors

mwi-relay error info debugging is on
01:46:48: MWI-APP: mwi_notify_status: No ClientID (7004) registered
```

Related Commands

Command	Description
debug ephone mwi	Sets MWI debugging for the Cisco IOS Telephony Service router.
debug mwi relay events	Sets MWI relay events debugging for the Cisco IOS Telephony Service router.

debug mwi relay events

To set message waiting indication (MWI) relay events debugging, use the **debug mwi relay events** command in privileged EXEC mode. To disable debugging output, use the **no** form of this command.

debug mwi relay events

no debug mwi relay events

Syntax Description

This command has no arguments or keywords.

Defaults

No default behavior or values

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

The **debug mwi relay events** command provides a debug monitor display of events, when MWI Relay Server (Cisco IOS Telephony Server) is trying to do MWI Relay to extensions on remote Cisco IOS Telephony Services (ITS).

Examples

The following debugging messages are shown when the MWI Relay server tries to send MWI Information to remote client 7001 and the location of 7001 is known by the MWI Relay Server:

```
Router# debug mwi relay events

mwi-relay events info debugging is on

01:45:34: mwi_notify_status: Queued event for mwi_app_queue
01:45:34: MWI-APP: mwi_app_process_event:
01:45:34: MWI-APP: mwi_app_process_event: MWI Event for ClientID(7001)@(1.8.17.22)
```

Related Commands

Command	Description
debug ephone mwi	Sets MWI debugging for the Cisco IOS Telephony Service router.
debug mwi relay errors	Sets MWI relay errors debugging for the Cisco IOS Telephony Service router.

debug voice register errors

To display debug information on voice register module errors during registration in a Cisco Unified CallManager Express (Cisco Unified CME) or Cisco Unified Session Initiation Protocol (SIP) Survivable Remote Site Telephony (SRST) environment, use the **debug voice register errors** command in privileged EXEC mode. To disable debugging, use the **no** form of the command.

debug voice register errors

no debug voice register errors

Syntax Description

This command has no arguments or keywords

Command Default

Disabled

Command Modes

Privileged EXEC mode

Command History

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

Usage Guidelines

Registration errors include failure to match pools or any internal errors that happen during registration.

Examples

Cisco Unified CME

The following is sample output for this command for a registration request with authentication enabled:

```
…
*May 6 18:07:26.971: VOICE_REG_POOL: Register request for (4901) from (10.5.49.83)
*May 6 18:07:26.971: VOICE_REG_POOL: key(9499C07A000036A3) added to nonce table
*May 6 18:07:26.975: VOICE_REG_POOL: Contact doesn't match any pools
*May 6 18:07:26.975: //4/89D7750A8005/SIP/Error/ccsip_spi_register_incoming_registration:
Registration Authorization failed with authorization header=
…
```

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

When authentication is enabled and if the phone requesting registration cannot be authenticated, the message “Registration Authorization failed with authorization header” is displayed.

Cisco Unified SIP SRST

The following is sample output from this command:

```
Router# debug voice register errors

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

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

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

Table 7 describes the significant fields shown in the display.

Table 7 debug voice register errors Field Descriptions

Field	Description
Contact (doesn’t match any pools)	Contact refers to the location of the SIP devices and the IP address.
key (*MAC address*)	Unique MAC address of a locally available individual SIP phone used to support a degree of authentication in Cisco Unified CME.
Register request for (*telephone number*) from (*IP address*).	The unique key for each registration is the telephone number.
Registration Authorization (failed with authorization header)	Registration Authorization message is displayed when **authenticate** command is configured in Cisco Unified CME.

Related Commands

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

debug voice register events

To display debug information on voice register module events during Session Initiation Protocol (SIP) phone registrations in a Cisco Unified CallManager Express (Cisco Unified CME) or Cisco Unified SIP Survivable Remote Site Telephony (SRST) environment, use the **debug voice register events** command in privileged EXEC mode. To disable debugging, use the **no** form of this command.

debug voice register events

no debug voice register events

Syntax Description

This command has no arguments or keywords

Command Default

Disabled

Command Modes

Privileged EXEC mode

Command History

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

Usage Guidelines

Using the **debug voice register events** command should suffice to view registration activity.

Registration activity includes matching of pools, registration creation, and automatic creation of dial peers. For more details and error conditions, you can use the **debug voice register errors** command.

Cisco Unified CME

The following example shows output from this command:

```
*May 6 18:07:27.223: VOICE_REG_POOL: Register request for (4901) from (1.5.49.83)
*May 6 18:07:27.223: VOICE_REG_POOL: Contact matches pool 1 number list 1
*May 6 18:07:27.223: VOICE_REG_POOL: key(4901) contact(10.5.49.83) add to contact table
*May 6 18:07:27.223: VOICE_REG_POOL: No entry for (4901) found in contact table
*May 6 18:07:27.223: VOICE_REG_POOL: key(4901) contact(10.5.49.83) added to contact
tableVOICE_REG_POOL pool->tag(1), dn->tag(1), submask(1)
*May 6 18:07:27.223: VOICE_REG_POOL: Creating param container for dial-peer 40001.
*May 6 18:07:27.223: VOICE_REG_POOL: Created dial-peer entry of type 0
*May 6 18:07:27.223: VOICE_REG_POOL: Registration successful for 4901, registration id is
2
...
```

The phone number 4901 associated with voice register pool 1, voice register dn 1, registered successfully. A dynamic normal (type 0) VoIP dial peer has been created for entry 4901. The dial peer can be verified using the **show voice register dial-peers** and **show sip-ua status registrar** commands.

Cisco Unified SIP SRST

The following is sample output from this command:

```
Router# debug voice register events

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

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

```
Apr 22 10:50:38.119 PDT: VOICE_REG_POOL: Register request for (91021) from (192.168.0.3)
Apr 22 10:50:38.119 PDT: VOICE_REG_POOL: Contact matches pool 2
Apr 22 10:50:38.123 PDT: VOICE_REG_POOL: key(91021) contact(192.168.0.3) add to contact
table
Apr 22 10:50:38.123 PDT: VOICE_REG_POOL: key(91021) exists in contact table
Apr 22 10:50:38.123 PDT: VOICE_REG_POOL: contact(192.168.0.3) exists in contact table, ref
updated
Apr 22 10:50:38.123 PDT: VOICE_REG_POOL: Created dial-peer entry of type 1
Apr 22 10:50:38.123 PDT: VOICE_REG_POOL: Registration successful for 91021, registration
id is 258
```

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

```
Apr 22 10:51:08.971 PDT: VOICE_REG_POOL: Register request for (95021) from (10.2.161.50)
Apr 22 10:51:08.971 PDT: VOICE_REG_POOL: Contact matches pool 3
Apr 22 10:51:08.971 PDT: VOICE_REG_POOL: key(95021) contact(10.2.161.50) add to contact
table
Apr 22 10:51:08.971 PDT: VOICE_REG_POOL: No entry for (95021) found in contact table
Apr 22 10:51:08.975 PDT: VOICE_REG_POOL: key(95021) contact(10.2.161.50) added to contact
table
Apr 22 10:51:08.979 PDT: VOICE_REG_POOL: Created dial-peer entry of type 0
Apr 22 10:51:08.979 PDT: VOICE_REG_POOL: Registration successful for 95021, registration
id is 259
Apr 22 10:51:09.019 PDT: VOICE_REG_POOL: Register request for (95012) from (10.2.161.50)
Apr 22 10:51:09.019 PDT: VOICE_REG_POOL: Contact matches pool 3
Apr 22 10:51:09.019 PDT: VOICE_REG_POOL: key(95012) contact(10.2.161.50) add to contact
table
Apr 22 10:51:09.019 PDT: VOICE_REG_POOL: No entry for (95012) found in contact table
Apr 22 10:51:09.023 PDT: VOICE_REG_POOL: key(95012) contact(10.2.161.50) added to contact
table
Apr 22 10:51:09.027 PDT: VOICE_REG_POOL: Created dial-peer entry of type 0
Apr 22 10:51:09.027 PDT: VOICE_REG_POOL: Registration successful for 95012, registration
id is 260
Apr 22 10:51:09.071 PDT: VOICE_REG_POOL: Register request for (95011) from (10.2.161.50)
Apr 22 10:51:09.071 PDT: VOICE_REG_POOL: Contact matches pool 3
Apr 22 10:51:09.071 PDT: VOICE_REG_POOL: key(95011) contact(10.2.161.50) add to contact
table
Apr 22 10:51:09.071 PDT: VOICE_REG_POOL: No entry for (95011) found in contact table
Apr 22 10:51:09.075 PDT: VOICE_REG_POOL: key(95011) contact(10.2.161.50) added to contact
table
Apr 22 10:51:09.079 PDT: VOICE_REG_POOL: Created dial-peer entry of type 0
Apr 22 10:51:09.079 PDT: VOICE_REG_POOL: Registration successful for 95011, registration
id is 261
Apr 22 10:51:09.123 PDT: VOICE_REG_POOL: Register request for (95500) from (10.2.161.50)
Apr 22 10:51:09.123 PDT: VOICE_REG_POOL: Contact matches pool 3
```

```
Apr 22 10:51:09.123 PDT: VOICE_REG_POOL: key(95500) contact(10.2.161.50) add to contact
table
Apr 22 10:51:09.123 PDT: VOICE_REG_POOL: No entry for (95500) found in contact table
Apr 22 10:51:09.127 PDT: VOICE_REG_POOL: key(95500) contact(10.2.161.50) added to contact
table
Apr 22 10:51:09.131 PDT: VOICE_REG_POOL: Created dial-peer entry of type 0
Apr 22 10:51:09.131 PDT: VOICE_REG_POOL: Registration successful for 95500, registration
id is 262
*Apr 22 11:52:54.523 PDT: VOICE_REG_POOL: Contact doesn't match any pools
*Apr 22 11:52:54.539 PDT: VOICE_REG_POOL: Register request for (33015) from (10.2.152.39)
*Apr 22 11:52:54.539 PDT: VOICE_REG_POOL: Contact doesn't match any pools
*Apr 22 11:52:54.559 PDT: VOICE_REG_POOL: Register request for (33017) from (10.2.152.39)
```

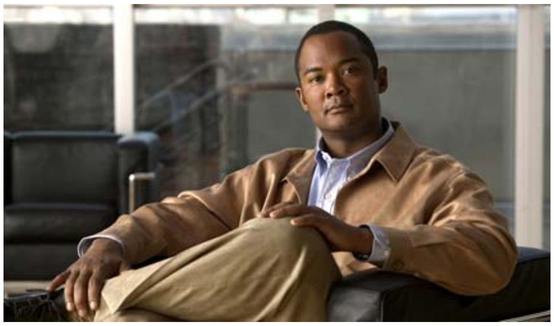
Table 8 describes the significant fields shown in the display.

Table 8 debug voice register events Field Descriptions

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

Related Commands

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



Cisco Unified CME Commands: E

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This chapter contains commands to configure and maintain Cisco Unified Communications Manager Express (formally known as Cisco Unified CallManager Express). The commands are presented in alphabetical order. Some commands required for configuring Cisco Unified Communications Manager Express (Cisco Unified CME) may be found in other Cisco IOS command references. Use the command reference master index or search online to find these commands.

ephone

To enter Ethernet phone (ephone) configuration mode for an IP phone for the purposes of creating and configuring an ephone, use the **ephone** command in global configuration mode. To disable the ephone and remove the IP phone configuration, use the **no** form of this command.

ephone *phone-tag*

no ephone *phone-tag*

Syntax Description

phone-tag	Unique sequence number that identifies an ephone during configuration tasks. The maximum number is platform-dependent; refer to Cisco IOS command-line interface (CLI) help.

Defaults

No Cisco IP phone is configured.

Command Modes

Global configuration

Command History

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

Usage Guidelines

Use the **ephone** command to enter ephone configuration mode. Use ephone configuration mode to create and configure Cisco Unified IP phones in Cisco Unified CME.

Before this command can be used for the first time, you must set the maximum number of ephones using the **max-ephones** command in telephony-service configuration mode. The maximum number of ephones varies by router platform and software version. For more information, refer to Cisco IOS command-line interface (CLI) help by entering the command name and a question mark (?) at the telephony-service configuration mode prompt:

```
Router(config-telephony)# max-ephones ?
```

When you are in ephone configuration mode, extensions (ephone-dns) that have already been defined using the **ephone-dn** command can be assigned to buttons on phones using the **button** command. You can also specify the MAC address of the phone instrument using the **mac-address** command. Other commands that are used in ephone configuration mode are described in the appropriate version of the Cisco CallManager Express documentation. Note that many of the commands in ephone configuration mode must be followed by a restart of the phone using the **restart (ephone)** or **restart all (telephony-service)** command.

Examples

The following example enters ephone configuration mode for a phone with the identifier 4 and assigns ephone-dn 1 to button 1:

```
Router(config)# ephone 4
Router(config-ephone)# button 1:1
```

Related Commands

Command	Description
button	Assigns a button number to the Cisco IP phone directory number.
ephone-dn	Enters ephone-dn configuration mode.
mac-address	Configures the MAC address of a Cisco IP phone.
max-ephones	Configures the maximum number of Cisco IP phones that can be supported by a router.
restart (ephone)	Performs a fast reboot of a single phone associated with a Cisco CME router.
restart all (telephony-service)	Performs a fast reboot of all phones associated with a Cisco CME router.

ephone-dn

To enter ephone-dn configuration mode for the purposes of creating and configuring an extension (ephone-dn) for a Cisco Unified IP phone line, an intercom line, a paging line, a voice-mail port, or a message-waiting indicator (MWI), use the **ephone-dn** command in global configuration mode. To delete an ephone-dn, use the **no** form of this command.

ephone-dn *dn-tag* [**dual-line**]

no ephone-dn *dn-tag*

Syntax Description

dn-tag	Unique sequence number that identifies a particular ephone-dn during configuration tasks. Range is from 1 to the number set by the **max-dn** command.
dual-line	(Optional) Enables dual-line mode for the ephone-dn.

Defaults

No ephone-dn is configured.

Command Modes

Global configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco ITS 1.0	This command was introduced on the following platforms: Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(2)XT	Cisco ITS 2.0	This command was implemented on the Cisco 1750 and Cisco 1751.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	Cisco ITS 2.0	This command was implemented on the Cisco 2600-XM and Cisco 2691.
12.2(11)T	Cisco ITS 2.01	This command was implemented on the Cisco 1760.
12.2(15)ZJ	Cisco CME 3.0	The **dual-line** keyword was added.
12.3(4)T	Cisco CME 3.4	The **dual-line** keyword was integrated into Cisco IOS Release 12.3(4)T.

Usage Guidelines

Use the **ephone-dn** command to enter ephone-dn configuration mode. Use ephone-dn configuration mode to create extensions (ephone-dns) in a Cisco Unified CME system. In ephone-dn configuration mode, you assign to the extension a number using the **number** command, a name to appear in the local directory using the **name** command, and other parameters using various commands.

Before using the **ephone-dn** command, you must set the maximum number of ephone-dns to appear in your system by using the **max-dn** command. The maximum number of ephone-dns that you can create depends on the software version, router platform, and amount of memory that you have installed. For the maximum number of ephone-dns and recommended memory for each platform, see the *Cisco Unified CallManager Express Supported Firmware, Platforms, Memory, and Voice Products* for your Cisco Unified CME version.

A dual-line ephone-dn has one virtual voice port and two channels to handle two independent calls. This capacity allows call waiting, call transfer, and conference functions within a single ephone-dn. Dual-line mode is supported on all phone types, but is not appropriate for voice-mail numbers, intercoms, or ephone-dns used for message-waiting indicators, paging, loopback, or hunt groups. Overlays of single-line hunt groups onto dual-line buttons are supported.

Ephone-dns are created in single-line mode if the **dual-line** keyword is not used. Changing an ephone-dn from dual-line mode to single-line mode (and vice versa) requires that you delete the ephone-dn and then recreate it.

Examples

The following example enters ephone-dn configuration mode to create the ephone-dn 5576:

```
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 5576
Router(config-ephone-dn)# exit
```

The following example creates an ephone-dn with the number 1001 in dual-line mode. The **no huntstop** command allows calls to continue to hunt to other ephone-dns if this one is busy or does not answer. The **huntstop channel** command disables call hunting to the second channel of this ephone-dn if the first channel is busy or does not answer.

```
Router(config)# ephone-dn 10 dual-line
Router(config-ephone-dn)# number 1001
Router(config-ephone-dn)# no huntstop
Router(config-ephone-dn)# huntstop channel
Router(config-ephone-dn)# exit
```

Related Commands

Command	Description
huntstop	Sets the ephone-dn huntstop attribute or the ephone-dn dual-line channel huntstop attribute.
max-dn	Sets the maximum number of ephone-dns that can be supported by a router.
name	Associates a name with an extension (ephone-dn).
number	Associates a telephone or extension number with an extension (ephone-dn).

ephone-dn-template

To enter ephone-dn-template configuration mode and create an ephone-dn template containing a standard set of ephone-dn features, use the **ephone-dn-template** command in global configuration mode. To delete an ephone-dn template, use the **no** form of this command.

ephone-dn-template *template-tag*

no ephone-dn-template *template-tag*

Syntax Description

template-tag	Identifier for this ephone-dn template. Range is from 1 to 15.

Command Default

No ephone-dn template is created.

Command Modes

Global configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Use this command to create an ephone-dn template. An ephone-dn template contains a set of ephone-dn attributes that you can easily apply to one or more ephone-dns.

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

Type **?** in ephone-dn-template configuration mode to see the commands that are available in this mode. The following example shows CLI help for ephone-dn-template configuration mode:

```
Router(config-ephone-dn-template)# ?

Ephone Dn template configuration commands:
  call-forward         Define E.164 telephone number for call forwarding
  call-waiting         Config call-waiting option
  caller-id            Configure port caller id parameters
  corlist              Class of Restriction on dial-peer for this dn
  default              Set a command to its defaults
  description          dn desc, for DN Qualified Display Name
  exit                 Exit from ephone-dn-template configuration mode
  hold-alert           Set Call On-Hold timeout alert parameters
  huntstop             Stop hunting on Dial-Peers
  mwi                  set message waiting indicator options (mwi)
  no                   Negate a command or set its defaults
  pickup-group         set the call pickup group number for the DN
  translate            Translation rule
  translation-profile  Translation profile
```

After creating an ephone-dn template, apply the template to one or more ephone-dns using the **ephone-dn-template** command in ephone-dn configuration mode. Even though you can define up to 15 different ephone templates, you cannot apply more than one template to a particular ephone-dn.

If you try to apply a second ephone-dn template to an ephone-dn that already has a template applied to it, the second template will overwrite the first ephone-dn template configuration after you use the **restart** command to reboot the phone.

To view your ephone-dn-template configurations, use the **show telephony-service ephone-dn-template** command. To see which ephone-dns have templates applied to them, use the **show running-config** command.

Examples

The following example creates ephone-dn template 3, which sets call forwarding on busy and no answer to forward calls to extension 4000 and sets the pickup group to 4. Ephone-dn template 3 is then applied to ephone-dn 23 and ephone-dn 33, which appear on ephones 13 and 14, respectively.

```
ephone-dn-template 3
 call-forwarding busy 4000
 call-forwarding noan 4000 timeout 30
 pickup group 4

ephone-dn 23
 number 2323
 ephone-dn-template 3

ephone-dn 33
 number 3333
 ephone-dn-template 3

ephone 13
 button 1:23

ephone 14
 button 1:33
```

Related Commands

Command	Description
ephone-dn-template (ephone-dn)	Applies an ephone-dn template to an ephone-dn.
restart (ephone)	Performs a fast reboot of a single phone associated with a Cisco Unified CME router.
restart (telephony-service)	Performs a fast reboot of one or all phones associated with a Cisco Unified CME router.
show telephony-service ephone-dn-template	Displays ephone-dn-template configurations.

ephone-dn-template (ephone-dn)

To apply an ephone-dn template to an ephone-dn, use the **ephone-dn-template** command in ephone-dn configuration mode. To remove the ephone-dn template, use the **no** form of this command.

ephone-dn-template *template-tag*

no ephone-dn-template *template-tag*

Syntax Description

template-tag	The template tag for a template created with the **ephone-dn-template** command in global configuration mode. Range is from 1 to 15.

Command Default

No ephone-dn template is applied to the ephone-dn.

Command Modes

Ephone-dn configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Use the **ephone-dn-template** command in ephone-dn configuration mode to apply an ephone-dn template to an ephone. You cannot apply more than one ephone-dn template to an ephone-dn.

If you try to apply a second ephone-dn template to an ephone-dn that already has an ephone-dn template applied to it, the second template will overwrite the first ephone-dn template configuration.

To view your ephone-dn-template configurations, use the **show telephony-service ephone-dn-template** command.

Examples

The following example shows how to create ephone-dn template 3, which sets call forwarding on busy and no answer to forward calls to extension 4000 and sets the pickup group to 4, and apply the template to ephone-dn 23 and ephone-dn 33, which appear on ephones 13 and 14, respectively.

```
ephone-dn-template 3
 call-forwarding busy 4000
 call-forwarding noan 4000 timeout 30
 pickup group 4

ephone-dn 23
 number 2323
 ephone-dn-template 3

ephone-dn 33
 number 3333
 ephone-dn-template 3
```

```
ephone 13
 button 1:23

ephone 14
 button 1:33
```

Related Commands

Command	Description
ephone-dn	Enters ephone-dn configuration mode.
ephone-dn-template	Creates an ephone-dn template and enters ephone-dn-template configuration mode.
show telephony-service ephone-dn-template	Displays ephone-dn template configurations.

ephone-hunt

To enter ephone-hunt configuration mode for the purposes of creating and configuring a hunt group for use in a Cisco Unified CME system, use the **ephone-hunt** command in global configuration mode. To delete a hunt group, use the **no** form of this command.

ephone-hunt *hunt-tag* {**longest-idle** | **peer** | **sequential**}

no ephone-hunt *hunt-tag*

Syntax Description

hunt-tag	Unique sequence number that identifies the ephone hunt group during configuration tasks. Range is from 1 to 100.
longest-idle	Hunt group in which calls go to the ephone-dn that has been idle the longest.
peer	Hunt group in which the first extension to ring is the number to the right (in the list) of the extension that was the last one to ring when the hunt group was last called. Ringing proceeds in a circular manner, left to right, for the number of hops specified when the ephone hunt group is defined.
sequential	Hunt group in which extensions ring in the order in which they are listed, left to right, when the hunt group is defined.

Command Default

No hunt group is defined.

Command Modes

Global configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.3(11)T	Cisco CME 3.2	The **longest-idle** keyword was added.
12.3(11)XL	Cisco CME 3.2.1	The maximum number of hunt groups was increased to 20.
12.3(14)T	Cisco CME 3.3	This command was integrated into Cisco IOS Release 12.3(14)T.
12.4(4)XC	Cisco Unified CME 4.0	The maximum number of hunt groups was increased to 100.
12.4(9)T	Cisco Unified CME 4.0	This command with the maximum number of hunt groups increased to 100 was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Use the **ephone-hunt** command to enter ephone-hunt configuration mode. Use ephone-hunt configuration mode to create ephone hunt groups in a Cisco Unified CME system.

A hunt group is a list of phone numbers that are assigned to take turns receiving incoming calls for one number, a pilot number that is defined with the **pilot** command. The list of numbers in the hunt group is defined using the **list** command. If a number in the list is busy or does not answer, the call is redirected to the next number in the list. The last number tried is the final number, which is defined using the **final** command.

The order in which the numbers are chosen can be longest-idle, peer, or sequential.

- If the order is longest-idle, each hop is directed to the ephone-dn that has been idle the longest. Idle time is determined from the last time that a phone registered, reregistered, or went on-hook.
- If the order is peer, the first number to which calls are directed is the number to the right of the number in the list that was the last number to ring on the previous occasion that the hunt group was called. If that number is busy or does not answer, the call is directed to the next number in the list and, in the process, circles back to the beginning of the list. In peer hunt groups, the **hops** command specifies how many times a call can hop from number to number before going to the final number, after which the call is no longer forwarded.
- If the order is sequential, the first number to which calls are directed is always the first number in the list. If that number is busy or does not answer, the call is redirected to the next available number in the list, from left to right.

Note If the number of times that a call is redirected to a new number exceeds 5, the **max-redirect** command must be used to increase the allowable number of redirects in the Cisco Unified CME system.

To configure a new hunt group, you must specify the **longest-idle**, **peer**, or **sequential** keyword. To change an existing ephone hunt group configuration, the keyword is not required. To change the type of hunt group from peer to sequential or sequential to peer, you must remove the existing hunt group first using the **no** form of the command and then recreate it.

Examples

The following example defines longest-idle ephone hunt group 1 with a pilot number 7501, a final number 8000, and 11 numbers in the list. After a call is redirected six times (makes six hops), it is redirected to the final number 8000.

```
ephone-hunt 1 longest-idle
 pilot 7501
 list 7001, 7002, 7023, 7028, 7045, 7062, 7067, 7072, 7079, 7085, 7099
 final 8000
 preference 1
 hops 6
 timeout 20
 no-reg
```

The following example defines peer hunt group number 2. Callers dial the pilot number 5610 to reach the hunt group. The first extension to ring the first time that this hunt group is called is 5601. If 5601 does not answer, the hunt proceeds from left to right, beginning with the extension directly to the right of 5601, for four hops. If none of those extensions answers before the hops limit is reached, the call is forwarded to extension 6000, which is the number for the voice-mail service.

If extension 5601 answers the first call, then the second time someone calls the hunt group, the first extension to ring is 5602. If this call hops until extension 5617 answers it, then the third time someone calls the hunt group, the first extension to ring is 5633. If extension 5633 does not answer, the call is redirected to extension 5601, and so forth.

```
Router(config)# ephone-hunt 2 peer
Router(config-ephone-hunt)# pilot 5610
Router(config-ephone-hunt)# list 5601, 5602, 5617, 5633
Router(config-ephone-hunt)# final 6000
Router(config-ephone-hunt)# hops 4
Router(config-ephone-hunt)# preference 1
Router(config-ephone-hunt)# timeout 30
Router(config-ephone-hunt)# exit
```

The following example defines sequential hunt group number 1. When callers dial extension 5601, the first phone to ring is 5001, then 5002, 5017, and 5028. If none of those extensions answers, the call is forwarded to extension 6000, which is the number for the voice-mail service.

```
Router(config)# ephone-hunt 1 sequential
Router(config-ephone-hunt)# pilot 5601
Router(config-ephone-hunt)# list 5001, 5002, 5017, 5028
Router(config-ephone-hunt)# final 6000
Router(config-ephone-hunt)# preference 1
Router(config-ephone-hunt)# timeout 30
Router(config-ephone-hunt)# exit
```

Related Commands

Command	Description
final	Defines the last ephone-dn in an ephone hunt group.
hops	Defines the number of times that a call is redirected to the next ephone-dn in a peer ephone-hunt-group list before proceeding to the final ephone-dn.
list	Defines the ephone-dns that participate in an ephone hunt group.
max-redirect	Changes the current number of allowable redirects in a Cisco Unified CME system.
no-reg (ephone-hunt)	Specifies that the pilot number of this ephone hunt group should not register with the H.323 gatekeeper.
pilot	Defines the ephone-dn that is dialed to reach an ephone hunt group.
preference (ephone-hunt)	Sets preference order for the ephone-dn associated with an ephone-hunt-group pilot number.
timeout (ephone-hunt)	Sets the number of seconds after which a call that is not answered is redirected to the next number in the hunt-group list.

ephone-hunt login

To authorize an ephone-dn to dynamically join and leave an ephone hunt group, use the **ephone-hunt login** command in ephone-dn configuration mode. To disable this capability, use the **no** form of this command.

ephone-hunt login

no ephone-hunt login

Syntax Description

This command has no arguments or keywords.

Command Default

An ephone-dn is not allowed to dynamically join and leave ephone hunt groups.

Command Modes

Ephone-dn configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Use the **show ephone-hunt** command to display current hunt group members, including those who joined the group dynamically.

Examples

The following example creates five ephone-dns and a hunt group that includes the first two ephone-dns and two wildcard slots. The last three ephone-dns are enabled for group hunt dynamic membership. Each of them can join and leave the hunt group whenever one of the slots is available.

```
ephone-dn 22
 number 4566

ephone-dn 23
 number 4567

ephone-dn 24
 number 4568
 ephone-hunt login

ephone-dn 25
 number 4569
 ephone-hunt login

ephone-dn 26
 number 4570
 ephone-hunt login

ephone-hunt 1 peer
```

```
list 4566,4567,*,*
final 7777
```

Related Commands

Command	Description
show ephone-hunt	Displays ephone-hunt group configuration, current status, and statistics.

ephone-hunt statistics write-all

To write ephone-hunt statistics information to a file, use the **ephone-hunt statistics write-all** command in privileged EXEC mode.

ephone-hunt statistics write-all *location*

Syntax Description

location	The URL or filename to which the statistics should be written.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Use this command to write out, in hourly increments, all the ephone hunt group statistics for the past seven days. This command is intended be used when normal hunt group statistics collection is interrupted, perhaps due to TFTP server failure. The commands that are normally used to provide hunt-group statistics are **hunt-group report delay hours**, **hunt-group report every hours**, **hunt-group report url**, and **statistics collect**. These commands allow you to specify shorter, more precise reporting periods and file-naming conventions.

Note Each year on the day that daylight saving time adjusts the time back by one hour at 2 a.m., the original 1 a.m. to 2 a.m. statistics for that day are lost because they are overwritten by the new 1 a.m. to 2 a.m. statistics.

Examples

The following example writes the ephone hunt group statistics to a file in flash called "huntstats." See the **hunt-group report url** command for explanations of the output fields.

```
Router# ephone-hunt statistics write-all flash:huntstats

Writing out all ephone hunt statistics to tftp now.
11:13:58 UTC Fri Apr 29 2005,
,
01, Fri 11:00 - 12:00, HuntGp, 01, 01, 00000, 00000, 00000, 0000, 0000, 000000, 000000,
0000, 00000, 000000, 000000,
01, Fri 12:00 - 13:00, HuntGp, 00, 00, 00000, 00000, 00000, 0000, 0000, 000000, 000000,
0000, 00000, 000000, 000000,
01, Fri 13:00 - 14:00, HuntGp, 00, 00, 00000, 00000, 00000, 0000, 0000, 000000, 000000,
0000, 00000, 000000, 000000,
01, Fri 14:00 - 15:00, HuntGp, 00, 00, 00000, 00000, 00000, 0000, 0000, 000000, 000000,
0000, 00000, 000000, 000000,
01, Fri 15:00 - 16:00, HuntGp, 00, 00, 00000, 00000, 00000, 0000, 0000, 000000, 000000,
0000, 00000, 000000, 000000,
.
```

.
.

Related Commands

Command	Description
show ephone-hunt	Displays ephone hunt group information.
show ephone-hunt statistics	Displays ephone hunt group statistics.
hunt-group report delay hours	Delays hunt-group statistics collection for a specified number of hours.s
hunt-group report every hours	Sets the hourly interval at which Cisco Unified CME B-ACD call statistics are automatically transferred to a file.
hunt-group report url	Sets filename parameters and the URL path where Cisco Unified CME B-ACD call statistics are to be sent using TFTP.
statistics collection	Enables the collection of call statistics for an ephone hunt group.

ephone-template

To create an ephone template to configure a set of phone features and to enter ephone-template configuration mode, use the **ephone-template** command in global configuration mode. To delete an ephone template, use the **no** form of this command.

ephone-template *template-tag*

no ephone-template *template-tag*

Syntax Description

template-tag	Identifier for this ephone template. Range is from 1 to 20.

Command Default

No ephone template is created.

Command Modes

Global configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)T	Cisco CME 3.2	This command was introduced.
12.4(4)XC	Cisco Unified CME 4.0	The maximum number of templates that can be created was increased from 5 to 20.
12.4(9)T	Cisco Unified CME 4.0	The modification to increase the maximum number of templates that can be created from 5 to 20 was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Use this command to create an ephone template containing a set of ephone commands. The template can then be easily applied to one or more ephones.

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

Type **?** in ephone-template configuration mode to see the commands that are available in this mode and that can be included in an ephone-template. The following example shows CLI help for ephone-template configuration mode at the time that this document was written:

```
Router(config-ephone-template)#?

Ephone template configuration commands:
  after-hour        ephone exempt from after-hour blocking
  codec             Set preferred codec for calls with other phones on this
                    router
  default           Set a command to its defaults
  exit              Exit from ephone-template configuration mode
  fastdial          Define ip-phone fastdial number
  features          define features blocked
  keep-conference   Do not disconnect conference when conference initiator
                    hangs-up.Connect remaining parties together directly using
```

```
                    call transfer.
  keepalive         Define keepalive timeout period to unregister IP phone
  keyphone          Identify an IP phone as keyphone
  mtp               Always send media packets to this router
  network-locale    Select the network locale for this template.
  night-service     Define night-service bell
  no                Negate a command or set its defaults
  paging-dn         set audio paging dn group for phone
  service           Service configuration in ephone template
  softkeys          define softkeys per state
  speed-dial        Define ip-phone speed-dial number
  transfer          transfer related configuration
  transfer-park     customized transfer to park configuration
  transfer-pattern  customized transfer-pattern configuration
  type              Define ip-phone type
  user-locale       Select the user locale for this template.
```

After creating an ephone template, apply the template to one or more ephones using the **ephone-template** command in ephone configuration mode. Even though you can define up to 20 different ephone templates, you cannot apply more than one template to a particular ephone.

After applying a template to an ephone or removing a template from an ephone, use the following commands:

- **restart**—Performs a fast reboot of the phone.
- **create cnf-files—**Rebuilds configuration files.

If you try to apply a second ephone template to an ephone that already has an ephone template applied to it, the second template will overwrite the first ephone template configuration after you use the **restart** command to reboot the phone.

To view your ephone-template configurations, use the **show telephony-service ephone-template** command. To see which ephones have templates applied to them, use the **show running-config** command.

Examples

The following example creates two ephone templates. The **softkeys** commands in ephone-template configuration mode define what soft keys are displayed and their order. Template 1 is applied to ephone 32, which has the extension 2555, and template 2 is applied to ephone 38, which has the extension 2666.

```
ephone-template 1
 softkeys idle Dnd Redial Newcall Pickup Login
 softkeys seized Redial Cfwdall Gpickup Pickup
 softkeys alerting Callback Endcall
 softkeys connected Confrn Hold Endcall

ephone-template 2
 softkeys idle Redial Pickup
 softkeys seized Redial Pickup
 softkeys connected Hold Endcall

ephone-dn 25
 number 2555

ephone-dn 26
 number 2666

ephone 32
 button 1:25
 ephone-template 1
```

```
ephone 38
 button 1:26
 ephone-template 2
```

The following example creates an ephone template to block the use of Park and Trnsfer soft keys. It is applied to extension 2333.

```
ephone-template 15
 features blocked Park Trnsfer

ephone-dn 2
 number 2333

ephone 3
 button 1:2
 ephone-template 15
```

Related Commands

Command	Description
create cnf-files	Builds phone configuration files.
ephone-template (ephone)	Applies an ephone template to an ephone.
restart (ephone)	Performs a fast reboot of a single phone associated with a Cisco Unified CME router.
restart (telephony-service)	Performs a fast reboot of one or all phones associated with a Cisco Unified CME router.
show telephony-service ephone-template	Displays ephone-template configurations.

ephone-template (ephone)

To apply an ephone template to an ephone, use the **ephone-template** command in ephone configuration mode. To remove the ephone template, use the **no** form of this command.

ephone-template *template-tag*

no ephone-template *template-tag*

Syntax Description

template-tag	The template tag for a template created with the **ephone-template** command in global configuration mode. Range is from 1 to 20.

Command Default

The default is that no ephone template is applied to an ephone.

Command Modes

Ephone configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)T	Cisco CME 3.2	This command was introduced.
12.4(4)XC	Cisco Unified 4.0	The maximum number of ephone templates that can be created was increased from 5 to 20.
12.4(9)T	Cisco Unified 4.0	The increased range for the *template-tag* argument was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Use the **ephone-template** command in ephone configuration mode to apply an ephone template to an ephone. You cannot apply more than one ephone template to an ephone.

After applying a template to an ephone, use the **restart** command to perform a fast reboot of the phone.

If you try to apply a second ephone template to an ephone that already has an ephone template applied to it, the second template will overwrite the first ephone template configuration after you use the **restart** command to reboot the phone.

To view your ephone-template configurations, use the **show telephony-service ephone-template** command.

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

Examples

The following example defines ephone templates 1 and 2 and applies ephone template 1 to ephones 1 through 3 and ephone template 2 to ephone 4.

```
ephone-template 1
 softkeys idle Dnd Redial Newcall Pickup Login
 softkeys seized Redial Cfwdall Gpickup Pickup
 softkeys alerting Callback Endcall
```

```
 softkeys connected Confrn Hold Endcall
 softkeys hold Newcall Resume

ephone-template 2
 softkeys idle Redial Pickup
 softkeys seized Redial Pickup
 softkeys alerting Endcall
 softkeys connected Hold Endcall
 softkeys hold Resume

 ephone 1
 ephone-template 1

ephone 2
 ephone-template 1

 ephone 3
 ephone-template 1

Rephone 4
 ephone-template 2

ephone 5
 ephone-template 2
```

The following example creates an ephone template to block the use of Park and Transfer soft keys on extension 2333.

```
ephone-template 15
 features blocked Park Trnsfer

ephone-dn 2
 number 2333

ephone 3
 button 1:2
 ephone-template 15
```

Related Commands

Command	Description
ephone	Enters ephone configuration mode for an IP phone.
ephone-template	Creates an ephone template and enters ephone-template configuration mode.
restart (ephone)	Performs a fast reboot of a single phone associated with a Cisco Unified CME router.
restart (telephony-service)	Performs a fast reboot of one or all phones associated with a Cisco Unified CME router.
show telephony-service ephone-template	Displays ephone-template configurations.

extension-assigner tag-type

To enable provision tags for identifying ephone configurations when using the extension assigner application, use the **extension-assigner tag-type** command in telephony-service configuration mode. To return to the default setting of using the ephone tag, use the **no** form of this command.

extension-assigner tag-type {ephone-tag | provision-tag}

no extension-assigner tag-type

Syntax Description

ephone-tag	Ephone tags must be used to identify ephone configurations.
provision-tag	Provision tags must be used to identify ephone configurations.

Command Default

Ephone tags are used to identify ephone configurations for the extension assigner application.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC4	Cisco Unified CME 4.0(3)	This command was introduced.
12.4(11)XJ	Cisco Unified CME 4.1	This command was introduced.
12.4(15)T	Cisco Unified CME 4.1	This command was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

This command enables you to use provision tags for identifying ephone configurations to be assigned by the extension application application.

A provision tag is an unique number other than an ephone tag, such as a jack number or an extension number, for identifying the ephone configuration to be assigned to a particular IP phone by the extension assigner application.

Use this command to specify which type of tag, ephone tag or provision tag, is to be used to identify ephone configurations for the extension assigner application. The default configuration is ephone tag.

If you use this command with the **provision-tag** keyword**,** use the **provision-tag** command to create provision tags.

Examples

The following example shows that this command is configured to enable provision tags to be used for identifying the ephone configurations to be assigned by the extension assigner application. Note that provision tag 1001 is configured for ephone 1. During phone installation, the installation technician can press 1001 on the telephone keypad to assign the ephone 1 configuration, with extension number 1001 on button 1, to the IP phone being installed.

```
Telephony-service
    extension-assigner tag-type provision-tag
    auto assign 101-102
```

```
    auto-reg-ephone

Ephone-dn 101
    number 1001

Ephone-dn 102
    number 1002

Ephone 1
    provision-tag 1001
    mac-address 02EA.EAEA.0001
    button 1:101

Ephone 2
    provision-tag 1002
    mac-address 02EA.EAEA.0002
    button 1:102
```

Related Commands

Command	Description
ephone	Enters ephone configuration mode for the purposes of creating and configuring an ephone.
provision-tag	Creates a provision tag for identifying an ephone configuration.

external-ring (voice register global)

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

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

no external-ring

Syntax Description

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

Command Default

The default ring sound is an internal ring pattern.

Command Modes

Voice register global configuration

Command History

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

Usage Guidelines

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

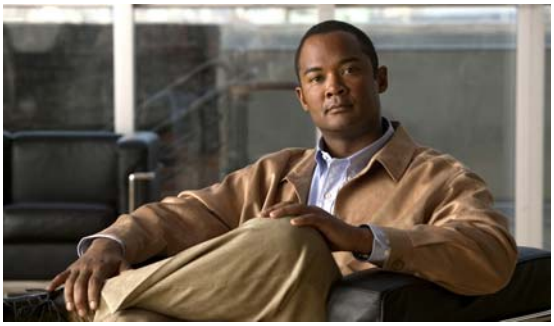
Examples

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

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

Related Commands

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



Cisco Unified CME Commands: F

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This chapter contains commands to configure and maintain Cisco Unified Communications Manager Express (formally known as Cisco Unified CallManager Express). The commands are presented in alphabetical order. Some commands required for configuring Cisco Unified Communications Manager Express (Cisco Unified CME) may be found in other Cisco IOS command references. Use the command reference master index or search online to find these commands.

fac

To enable all standard feature access codes (FACs) or to create and enable individual custom FACs, use the **fac** command in telephony-service configuration mode. To disable FACs, use the **no** form of this command.

fac {**standard** | **custom** {**alias** *alias-tag custom-fac* **to** *existing-fac* [*extra-digits*]} | *feature custom-fac*}}

no fac {**standard** | **custom** {**alias** *alias-tag* | *feature*}}

Syntax Description

standard	All predefined standard FACs are enabled.
custom	User-defined FAC for selecting a particular feature or function from the predefined set of features is enabled.
alias	Alternative FAC for dialing an existing FAC or existing FAC plus extra digits without removing the existing FAC is enabled.
alias-tag	Unique number that identifies this alias during configuration tasks. Range: 0 to 9.
custom-fac	User-defined code to dial using the keypad on an IP or analog phone. Code can be up to 256 characters long and contain numbers 0 to 9 and * and #.
to	Maps custom FAC being configured to specified target.
existing-fac	Already configured custom FAC that is automatically dialed when the phone user dials the custom FAC being configured.

extra-digits	(Optional) Additional digits that are automatically dialed when the phone user dials the custom FAC being configured. Valid entries are: • **target extension**—Telephone or extension number in Cisco Unified CME to which the incoming calls are to forwarded. Used with the Call Forward feature. • **group number**—Pickup group number, for a group other than the local group number. Used with the Pickup Group feature. • **pickup extension**—Telephone or extension number in Cisco Unified CME to be picked up when ringing. To be used with the Pickup Direct feature. • **park-slot number**—Number on which calls are to be temporarily parked. Use with the Call Park feature. Target park slot must be already configured in Cisco Unified CME. • **pilot number**—Telephone or extension number configured as a the pilot number for an ephone hunt group to be joined. Hunt group to be joined must allow dynamic membership.
feature	Predefined alphabetic string that identifies a particular feature or function. Valid options are: • **callfwd all**—Directs system to forward all incoming calls for this telephone or extension number. • **callfwd cancel**—Directs system to cancel the call-forward-all selection. • **dnd**—Enables Do Not Disturb (DND) feature. • **ephone-hunt cancel**—Leaves an ephone hunt group that is configured to allow dynamic membership. • **ephone-hunt hlog**—Activates or deactivates hunt group logout functionality, changing the status of the an ephone-dn for a hunt group agent from ready to not-ready or from not-ready to ready. • **ephone-hunt hlog-phone**—Activates or deactivates phone-level hunt group logout functionality, changing the status of all the extensions on a hunt group member phone from ready to not-ready or from not-ready to ready. • **ephone-hunt join**—Joins an ephone hunt group that is configured to allow dynamic membership. If multiple hunt groups have been created that allow dynamic membership, the hunt group to be joined is identified by its pilot number. • **park**—Enables Call Park feature. • **pickup direct**—Picks up a ringing call at any extension. • **pickup group**—Picks up a ringing call in a different pickup group than yours. • **pickup local**—Picks up a ringing call in your pickup group. • **redial**—Redials the last number called. • **voicemail**—Dials the voice-mail number.

Command Default

FACs are disabled on IP phones.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Use this command to enable all predefined standard feature access codes (FACs) or to create one or more custom FACs.

FACs enable phone users to use the keypad on an analog or IP phone registered in Cisco Unified CME to select or activate/deactivate a particular feature or function from a predefined set of features. For example, a phone user might press **1, then press 2345 to forward all incoming calls to extension 2345.

Standard FACs and custom FACs are mutually exclusive. You can enable all standard FACS or create and enable one or more custom FACs.

Most FACs are valid only immediately after a phone user goes off-hook. The only exception is the call-park FAC. The call-park FAC actually invokes a call transfer to a park slot. To use the call-park FAC, a phone user must have an active call and must press the Transfer soft key (IP phone) or hookflash (analog phone) before dialing the call-park FAC. Dialing the FAC for the Call Park feature does not use the Park soft key function.

Use the **fac standard** command to enable all predefined standard FACs for all SCCP phones registered in Cisco Unified CME.

Use the **fac custom** command to create an individual custom FAC for selecting a particular feature or function from the predefined feature set.

Use the **fac custom** command with the **alias** keyword to create an alternative (custom) FAC for dialing an existing FAC, or existing FAC plus extra digits without removing the existing FAC. For example, an alias can be created to allow the phone user press **1 to forward all incoming calls to a particular extension *without* requiring the phone user to dial the target extension number.

To disable *all* custom FACs, use the **fac standard** command, which enables all standard FACs. To disable all standard FACs or to disable an individual custom FAC, use the **no** form of the **fac** command.

Use the **show telephony-service fac** command to display a list of FACs that are configured on the Cisco Unified CME router.

Examples

The following example shows how to enable standard FACs for all phones:

```
Router# telephony-service
Router(config-telephony)# fac standard
fac standard is set!
Router(config-telephony)#
```

The following example shows the output from the show telephony-service fac command when standard FACs are enabled:

```
Router# show telephony-service fac
```

```
telephony-service fac standard
 callfwd all **1
 callfwd cancel **2
 pickup local **3
 pickup group **4
 pickup direct **5
 park **6
 dnd **7
 redial **8
 voicemail **9
 ephone-hunt join *3
 ephone-hunt cancel #3
 ephone-hunt hlog *4
 ephone-hunt hlog-phone *5
```

The following example shows how the standard FAC for the Call Forward All feature is changed to a custom FAC (#45). Then an alias is created to map a second custom fac to #45 plus an extension (1111). The second custom FAC (#44) allows the phone user to press #44 to forward all calls all calls to extension 1111, without requiring the phone user to dial the extra digits that are the extension number.

```
Router# telephony-service
Router(config-telephony)# fac custom callfwd all #45
fac callfwd all code has been configured to #45
Router(config-telephony)# fac custom alias 0 #44 to #451111
fac alias0 code has been configurated to #44!
alias0 map code has been configurated to #451111!
```

The following example shows how to create three aliases for the Group Pickup feature. The FAC for group pickup is **4. The three new custom FACs are #1, #2, and #4 to pickup groups 121, 122, and 124, respectively. This allows a phone user to press #1 to pick up calls in group 121, #2 to pick up calls in group 122, and #4 to pick up calls in group 124.

```
Router# telephony-service
Router(config-telephony)# fac custom pickup group **4
fac pickup group code has been configured to **4
Router(config-telephony)# fac custom alias 1 #1 to **4121
fac alias1 code has been configurated to #1!
alias1 map code has been configurated to **4121!
Router(config-telephony)# fac custom alias 2 #2 to **4122
fac alias2 code has been configurated to #2!
alias2 map code has been configurated to **4122!
Router(config-telephony)# fac custom alias 4 #4 to **4124
fac alias4 code has been configurated to #4!
alias4 map code has been configurated to **4124!
Router(config-telephony)#
```

The following example shows the output from the show telephony-service fac command when custom FACs are configured:

```
Router# show telephony-service fac
telephony-service fac custom
 callfwd all #45
 alias 0 #44 to #451111
 alias 1 #1 to **4121
 alias 2 #2 to **4122
 alias 4 #4 to **4124

Router#
```

Related Commands

Command	Description
show telephony-service fac	Displays list of feature access codes (FACs) that are configured on the Cisco Unified CME.

fastdial

To create an entry for a personal speed-dial number, use the **fastdial** command in ephone or ephone-template configuration mode. To delete a personal speed-dial number, use the **no** form of this command.

fastdial *dial-tag number* **name** *name-string*

no fastdial *dial-tag*

Syntax Description

dial-tag	Unique sequence number that is used to identify a particular personal speed-dial number during configuration tasks. Range is from 1 to 24.
number	Telephone number or extension to be dialed.
name *name-string*	Label to appear in the Personal Speed Dial menu, containing a string of up to 24 alphanumeric characters. Personal speed dial is handled through an XML request, so characters that have special meaning to HTTP, such as ampersand (&), percent sign (%), semicolon (;), angle brackets (< >), and vertical bars (||), are not allowed.

Command Default

No personal speed-dial numbers are present.

Command Modes

Ephone configuration
Ephone-template configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.4(4)XC	Cisco Unified CME 4.0	This command was made available in ephone-template configuration mode.
12.4(9)T	Cisco Unified CME 4.0	This command in ephone-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

The **fastdial** command is supported only on the Cisco Unified IP Phone 7940s and 7940Gs and the Cisco Unified IP Phone 7960s and 7960Gs.

Phone users access personal speed-dial numbers through the **Directories > Local Services > Personal Speed Dial** menu. Personal speed-dial numbers appear on this menu in the order in which they are entered during configuration.

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

Examples

The following example creates a directory of five personal speed-dial numbers for an IP phone:

```
Router(config)# ephone 1
Router(config-ephone)# fastdial 1 5001 name Front Register
Router(config-ephone)# fastdial 2 5002 name Security
Router(config-ephone)# fastdial 3 5003 name Rear Register
Router(config-ephone)# fastdial 4 5004 name Office
Router(config-ephone)# fastdial 5 912135550122 Accounting
```

fastdial (voice register pool)

To create personal speed-dial numbers, use the **fastdial** command in voice register pool configuration mode. To delete a personal speed-dial number, use the **no** form of this command.

fastdial *dial-tag number* [**name** *name-string*]

no fastdial *dial-tag*

Syntax Description

dial-tag	Unique number that identifies a particular personal speed-dial number during configuration tasks. Range is 1 to 24.
number	Telephone number or extension to be dialed.
name *name-string*	(Optional) Label to appear in the Personal Speed Dial menu, containing a string of up to 24 alphanumeric characters. Personal speed dial is handled through an XML request, so characters that have special meaning to HTTP, such as ampersand (&), percent sign (%), semicolon (;), angle brackets (< >), and vertical bars (||), are not allowed.

Command Default

No personal speed-dial numbers are defined.

Command Modes

Voice register pool configuration

Command History

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

Usage Guidelines

This command creates personal speed-dial numbers through Cisco Unified CME. Support for this command depends on the model of SIP phone:

- Cisco Unified IP Phone 7941G, 7941GE, 7961G, 7961GE, 7970G, and 7971GE—Personal speed-dial numbers can be created through Cisco Unified CME and not by the user on the phone.
- Cisco Unified IP Phone 7905, 7912, 7940, and 7960—Personal speed-dial numbers can only be created by the user directly on the phone and not through Cisco Unified CME.

Phone users access personal speed-dial numbers through the **Directories** > **Personal Speed Dial** menu. Personal speed-dial numbers display on this menu in alphabetical order.

Examples

The following example shows a directory of five personal speed-dial numbers defined for a SIP phone:

```
Router(config)# voice register pool 1
Router(config-register-pool)# fastdial 1 5001 name Front Register
Router(config-register-pool)# fastdial 2 5002 name Security
Router(config-register-pool)# fastdial 3 5003 name Rear Register
```

```
Router(config-register-pool)# fastdial 4 5004 name Office
Router(config-register-pool)# fastdial 5 912135550122 Accounting
```

Related Commands

Command	Description
show voice register pool	Displays all configuration information associated with a particular voice register pool.

features blocked

To prevent one or more features from being used on a Cisco Unified CME phone, use the **features blocked** command in ephone-template configuration mode. To allow all features to be used, use the **no** form of this command.

features blocked [**CFwdAll**] [**Confrn**] [**GpickUp**] [**Park**] [**PickUp**] [**Trnsfer**]

no features blocked

Syntax Description

CFwdAll	Call forward all calls.
Confrn	Conference.
GpickUp	Group call pickup.
Park	Call park.
PickUp	Directed or local call pickup. This includes pickup last-parked call and pickup from another extension or park slot.
Trnsfer	Call transfer.

Command Default

Features are not blocked.

Command Modes

Ephone-template configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Use this command to specify one or more features to be blocked in an ephone template, then apply the template in ephone configuration mode to one or more ephones to prevent the use of the specified features by those ephones. This feature can be used on IP phones and analog phones. After applying the template, any soft keys associated with the blocked features will still be visible but will not have any effect.

Use the **show telephony-service ephone-template** command to display the contents of ephone templates.

Examples

In the following example, call park and call transfer are blocked on ephone 3.

```
ephone-template 1
 features blocked Park Trnsfer

ephone-dn 2
 number 2333
```

```
ephone 3
 button 1:2
 ephone-template 1
```

The following example blocks the use of the conference feature on ephone 3, which is an analog phone.

```
ephone-template 1
 features blocked Confrn

ephone-dn 78
 number 2579

ephone  3
 ephone-template 1
 mac-address C910.8E47.1282
 type anl
 button  1:78
```

Related Commands

Command	Description
show telephony-service ephone-template	Displays the contents of ephone templates.

feed

To enable an audio stream for multicast from a external live audio feed connected directly to the router by a foreign exchange office (FXO) or an E&M analog voice port, use the **feed** command in ephone-dn configuration mode. To disable the multicast audio stream, use the **no** form of this command.

feed ip *ip-address* **port** *port-number* [**route** *ip-address*] [**out-call** *outcall-number*]

no feed ip

Syntax Description

ip *ip-address*	Indicates that a particular audio stream is to be used as a multicast source and specifies the destination IP address for multicast.
port *port-number*	Specifies the media port for multicast. Range is from 2000 to 65535. Port 2000 is recommended because this port is already used for normal Real-Time Transport Protocol (RTP) media transmissions between IP phones and the Cisco CallManager Express (Cisco CME) router.
route *ip-address*	(Optional) Indicates the specific router interface on which to transmit the IP multicast packets. The default is that the audio stream is automatically output on the interface that corresponds to the address that was configured with the **ip source-address** command.
out-call *outcall-number*	(Optional) Sets up a call to the outcall number in order to connect to a live audio feed. If this keyword is not used, the live feed is assumed to derive from an incoming call to the ephone-dn that is being configured.

Defaults

No multicast audio stream is enabled on an extension.

Command Modes

Ephone-dn configuration

Command History

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

Usage Guidelines

When this command is used, a connection for a live feed audio stream is established as an automatically connected voice call. If the **out-call** keyword is used, the Cisco CME system calls out to the specified number for the audio stream. If the **out-call** keyword is not used, it is assumed that the call is incoming to the ephone-dn. This includes VoIP calls if voice activity detection (VAD) is disabled. The typical operation is for the Cisco CME ephone-dn to establish a call to a local router E&M voice port.

Connection via E&M is the recommended mechanism because it requires minimal external components. The E&M port must be placed in 4-wire operation, using E&M immediate signaling and with the auto-cut-through option enabled. You directly connect a line-level audio feed (standard audio jack) to pins 3 and 6 of an E&M RJ-45 connector. The E&M WAN interface card (WIC) has a built-in audio transformer that provides appropriate electrical isolation for the external audio source. (The audio connection on the E&M port does not require loop current.) The **signal immediate** and **auto-cut-through** commands disable E&M signaling on this voice port. A G.711 audio packet stream is generated by the digital signal processor (DSP) on the E&M port.

If you are using an FXO voice port for live-feed audio stream instead of an E&M port, connect the source to the FXO voice port. This connection requires an external adapter to supply normal telephone company (telco) battery voltage with the correct polarity to the tip-and-ring leads of the FXO port. The adapter must also provide transformer-based isolation between the external audio source and the tip-and-ring leads of the FXO port.

If the **out-call** keyword is used, an outbound call to the live-feed source is attempted (or reattempted) every 30 seconds until the call is connected to the ephone-dn (extension) for which the **feed** command was configured. Note that this ephone-dn is not associated with a physical phone.

The related **moh** (ephone-dn) and **multicast moh** commands provide the ability to multicast an audio stream that is also being used as the source for Cisco CME system music on hold (MOH).

Note IP phones do not support multicast at 224.x.x.x addresses.

Examples

The following example sets up a call to extension 7777 for a live audio stream and sends it via multicast:

```
Router(config)# ephone-dn 55
Router(config-ephone-dn)# feed ip 239.1.1.1 port 2000 route 10.10.23.3 out-call 7777
```

Related Commands

Command	Description
auto-cut-through	Enables call completion when an M-lead response is not provided.
ephone-dn	Enters ephone-dn configuration mode to set extension numbers and parameters for individual Cisco IP phone lines.
ip source-address	Identifies the IP address and port through which IP phones communicate with a Cisco CME router.
moh (ephone-dn)	Enables music on hold from a live feed and multicast of the MOH audio stream.
moh (telephony-service)	Enables music on hold from an audio file.
multicast moh	Enables multicast of a music-on-hold audio stream.
signal	Specifies the type of signaling for a voice port.

filename

To specify a custom XML file that contains the dial patterns to use for a SIP dial plan, use the **filename** command in voice register dialplan configuration mode. To remove the file, use the **no** form of this command.

filename *filename*

no filename

Syntax Description

filename	Name of the XML file in flash memory.

Command Default

A custom file is not used for the dial plan.

Command Modes

Voice register dialplan configuration

Command History

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

Usage Guidelines

This command selects a custom XML file containing dial patterns for the SIP dial plan. The file specified with this command must be loaded into flash memory. You must use the **type** command to specify the type of phone for which the dial plan is being defined before you can use this command. After you define a dial plan, assign it to a SIP phone by using the **dialplan** command.

The **pattern** command and **filename** command are mutually exclusive. You can use either the **pattern** command to define dial patterns manually, or the **filename** command to select a custom dial pattern file that is loaded in system flash.

If the custom XML file contains any errors, the dial plan might not work properly on the phone.

To remove a dial plan that is created using a custom XML file, use the **reset** command after removing the dial plan from the phone and creating a new configuration profile. Removing a dial plan that uses a dial pattern XML file does not take effect if you restart the phone with the **restart** command.

Note This command is not supported for Cisco Unified IP Phone 7905 or 7912.

Examples

The following example shows that a custom file named sample.xml is specified for dial plan 2.

```
Router(config)# voice register dialplan 2
Router(config-register-dialplan)# type 7940-7960-others
Router(config-register-dialplan)# filename sample.xml
```

Related Commands

Command	Description
dialplan	Assigns a dial plan to a SIP phone.
pattern	Defines a dial pattern for a SIP dial plan.
show voice register dialplan	Displays all configuration information for a specific SIP dial plan.
type (voice register dialplan)	Defines a phone type for a SIP dial plan.

file text (voice register global)

To generate ASCII text files of the configuration profiles for SIP phones, use the **file text** command in voice register global configuration mode. To return to the default, use the **no** form of this command.

file text

no file text

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled

Command Modes

Voice register global configuration

Command History

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

Usage Guidelines

Use this command to generate an ASCII text fils of the configuration profile for Cisco Unified IP Phone 7905s and 7905Gs, Cisco Unified IP Phone 7912s and 7912Gs, Cisco ATA-186s, or Cisco ATA-188s. By default, the system directly generates binary files to save disk space. Use this command if you prefer to generate ASCII text files.

Examples

The following example shows how to generate an ASCII text file version of the configuration profiles for Cisco Unified IP Phone 7905s and 7905Gs, Cisco Unified IP Phone 7912s and 7912Gs, Cisco ATA-186s, or Cisco ATA-188s:

```
Router(config)# voice register global
Router(config-register-global)# mode cme
Router(config-register-global)# file text
Router(config-register-global)# create profile
```

Related Commands

Command	Description
create profile (voice register global)	Generates the configuration profiles required for SIP phone.
reset (voice register global)	Performs a complete reboot of all SIP phones associated with a Cisco CME router.
show voice register profile	Displays the content of configuration files that are in ASCII text format.

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

final

To define the last extension (ephone-dn) in an ephone hunt group, use the **final** command in ephone-hunt configuration mode. To remove this number from the hunt group, use the **no** form of this command.

final *dn-number*

no final *dn-number*

Syntax Description

dn-number	Ephone-dn number. Can be an ephone-dn primary or secondary number, voice-mail number, pilot number of another hunt group, or FXS caller-ID number.

Defaults

No final number is defined.

Command Modes

Ephone-hunt configuration

Command History

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

Usage Guidelines

If a final number in one hunt group is configured as a pilot number of another hunt group, the pilot number of the first hunt group cannot be configured as a final number in any hunt group.

Examples

The following example defines ephone-dn 6000 as the last number of hunt group number 1:

```
Router(config)# ephone-hunt 1 sequential
Router(config-ephone-hunt)# final 6000
```

Related Commands

Command	Description
ephone-hunt	Defines an ephone hunt group and enters ephone-hunt configuration mode.
hops	Defines the number of times that a call is redirected to the next ephone-dn in a peer ephone-hunt-group list before proceeding to the final ephone-dn.
list	Defines the ephone-dns that participate in an ephone hunt group.
max-redirect	Changes the current number of allowable redirects in a Cisco CME system.
no-reg (ephone-hunt)	Specifies that the pilot number of an ephone hunt group should not register with the H.323 gatekeeper.
pilot	Defines the ephone-dn that is dialed to reach an ephone hunt group.

Command	Description
preference (ephone-hunt)	Sets preference order for the ephone-dn associated with an ephone-hunt-group pilot number.
timeout (ephone-hunt)	Sets the number of seconds after which a call that is not answered is redirected to the next number in the ephone-hunt-group list.

final (voice hunt-group)

To define the last extension in a voice hunt group, use the **final** command in voice hunt-group configuration mode. To remove this number from the hunt group, use the **no** form of this command.

final *directory-number*

no final *directory-number*

Syntax Description

directory-number	Telephone or extension number. Can be an E.164 number, voice-mail number, pilot number of another hunt group, or FXS caller-ID number.

Defaults

This command has no arguments or keywords.

Command Modes

Voice hunt-group configuration

Command History

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

Usage Guidelines

If a final number in one hunt group is configured as a pilot number of another hunt group, the pilot number of the first hunt group cannot be configured as a final number in any other hunt group.

Examples

The following example shows how to define extension 6000 as the last number of hunt group 1:

```
Router(config)# voice hunt-group 1 sequential
Router(config-voice-hunt-group)# final 6000
```

Related Commands

Command	Description
voice hunt-group	Defines a voice hunt group and enters hunt-group configuration mode.
hops (voice hunt-group)	Defines the number of times that a call is redirected to the next number in a peer hunt-group list before proceeding to the final number.
list (voice hunt-group)	Defines the numbers that participate in a voice hunt group.
max-redirect (voice register global)	Changes the current number of allowable redirects in a Cisco CME system.
timeout (voice hunt-group)	Sets the number of seconds after which a call that is not answered is redirected to the next number in the hunt-group list.

forward local-calls

To allow internal (local) calls to be forwarded, use the **forward local-calls** command in ephone-dn or ephone-hunt configuration mode. To prevent internal calls from being forwarded, use the **no** form of this command.

forward local-calls

no forward local-calls

Syntax Description

This command has no arguments or keywords.

Command Default

Internal calls are forwarded as specified in the ephone-dn or ephone-hunt configuration of the called party.

Command Modes

Ephone-dn configuration
Ephone-hunt configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Internal, or local, calls are defined as those calls that originate from other ephone-dns in the same Cisco Unified CME system.

When the **no forward local-calls** command is used in ephone-dn configuration mode, internal calls to that ephone-dn are not forwarded if the ephone-dn is busy or does not answer. If the ephone-dn is busy, the caller hears a busy signal.If the ephone-dn does not answer, the caller hears a ringback signal. The call is not forwarded even if call forwarding is enabled for the ephone-dn.

When the **no forward local-calls** command is used in ephone-hunt configuration mode, internal calls to a hunt-group pilot number are sent only to the first member of the group. If the first group member is busy, the caller hears a busy signal. If the first group member does not answer, the caller hears a ringback signal. The call is not forwarded to subsequent hunt group members.

Examples

In the following example, extension 2222 dials the pilot number 3000 and is forwarded to extension 3011. If 3011 is busy, the caller hears a busy tone. If 3011 does not answer, the caller hears ringback. The call is not forwarded, even after the timeout expires.

```
ephone-hunt 17 sequential
 pilot 3000
 list 3011, 3021, 3031
 timeout 10
 final 7600
 no forward local-calls
```

In the following example, extension 2222 calls extension 3675 and hears ringback or a busy signal. If an external caller reaches extension 3675 and there is no answer, the call is forwarded to extension 4000.

```
ephone-dn 25
 number 3675
 no forward local-calls
 call-forward noan 4000 timeout 30
```

Related Commands

Command	Description
ephone-dn	Enters ephone-dn configuration mode.
ephone-hunt	Defines an ephone hunt group and enters ephone-hunt configuration mode.

forwarding local (voice register global)

To use the forwarding-party number and name (the local number and name) in calls forwarded using local hairpin call routing on a SIP phone, use the **forwarding local** command in voice register global configuration mode. To return to the default, use the **no** form of this command.

forwarding local

no forwarding local

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled

Command Modes

Voice register global configuration

Command History

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

Usage Guidelines

This command replaces a calling-party number and name with the local forwarding-party number and name in hairpinned forwarded calls.

Examples

The following example shows how to enable local forwarding:

```
Router(config)# voice register global
Router(config-register-global)# forwarding local
```

Related Commands

Command	Description
call-forward b2bua all (voice register dn and voice register pool)	Enables call forwarding for a SIP B2BUA so that all incoming calls are forwarded to another extension.
voice register global	Enters voice register global configuration mode in order to set global parameters for all supported Cisco SIP phones in a Cisco CME or Cisco SIP SRST environment.

from-ring

To specify that on-hook time stamps for ephone hunt group agents should be updated when calls ring as well as when calls are answered in a longest-idle ephone hunt group, use the **from-ring** command in ephone-hunt configuration mode. To return to the default, use the **no** form of this command.

from-ring

no from-ring

Syntax Description

This command has no keywords or arguments.

Command Default

On-hook time stamps are updated only when calls are answered by agents.

Command Modes

Ephone-hunt configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used only with longest-idle ephone hunt groups. In a longest-idle hunt group, the algorithm for choosing the the next agent to receive a call is based on a comparison of on-hook time stamps. The agent with the smallest on-hook time stamp value is chosen when the next call comes to the hunt group.

This command can be used to specify that on-hook time stamps should be updated when calls ring agents as well as when calls are answered by agents.

The **show ephone-hunt** command displays on-hook time stamps.

Examples

The following example defines longest-idle ephone hunt group 1 with a pilot number 7501, a final number 8000, and five numbers in the list. Because the **from-ring** command is used, on-hook time stamps will be recorded when calls ring agents as well as when calls are answered. After a call is redirected three times (makes six hops), it is redirected to the final number, 8000.

```
ephone-hunt 1 longest-idle
 pilot 7501
 list 7001, 7002, 7023, 7028, 7045
 final 8000
 from-ring
 hops 3
 timeout 20

telephony-service
 max-redirect 8
```

Related Commands

Command	Description
ephone-hunt	Defines an ephone hunt group and enters ephone-hunt configuration mode.
show ephone-hunt	Displays configuration information, current status, and statistics for ephone hunt groups.

fwd-final

To specify the final destination of an unanswered call that has been transferred into a hunt group, use the **fwd-final** command in ephone-hunt configuration mode. To return to the default, use the **no** form of this command.

fwd-final {**orig-phone** | **final**}

no fwd-final {**orig-phone** | **final**}

Syntax Description

orig-phone	Returns unanswered calls to the phone that transferred them to the pilot number of the hunt group.
final	Sends unanswered calls to the final number specified in the hunt group configuration.

Command Default

Calls are sent to the final number that is specified in the hunt group configuration.

Command Modes

Ephone-hunt configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

The **fwd-final** command is used only for incoming calls to an ephone hunt group that have been previously answered by a Cisco Unified CME extension and transferred into the hunt group. If a transferred call is not answered by the hunt group, it is routed as specified in the **fwd-final** command. The **orig-phone** keyword specifies that an unanswered transferred call should be returned to the extension that originally answered the call and transferred it to the hunt group.

When you use the **final** keyword with the **fwd-final** command, unanswered calls that have been transferred into the hunt group are routed to the destination specified in the hunt-group **final** command. The **final** command specifies the last destination for incoming calls that dial the pilot number of the hunt group directly.

Examples

The following example sets up a peer hunt group with three ephone-dns to answer calls. A call that is unanswered will be returned to the ephone-dn that transferred it to the ephone hunt group pilot number. A DID call that dials the pilot number directly will be sent to extension 7600 if it is unanswered by the hunt group.

```
ephone-hunt 17 peer
 pilot 3000
 list 3011, 3021, 3031
 hops 3
```

```
final 7600
fwd-final orig-phone
```

Related Commands

Command	Description
ephone-hunt	Defines an ephone hunt group and enters ephone-hunt configuration mode.

fxo hook-flash

To enable display of a flash soft key on a Cisco IP Phones 7940 and 7940G or Cisco IP Phones 7960 and 7960G in a Cisco CallManager Express (Cisco CME) system, use the **fxo hook-flash** command in telephony-service configuration mode. To disable display of the flash soft key, use the **no** form of this command.

fxo hook-flash

no fxo hook-flash

Syntax Description

This command has no arguments or keywords.

Defaults

The flash soft key is disabled.

Command Modes

Telephony-service configuration

Command History

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

Usage Guidelines

Certain public switched telephony network (PSTN) services, such as three-way calling and call waiting, require hookflash intervention from the phone user. A soft key labeled flash provides this functionality for the Cisco IP Phones 7940 and 7940G and the Cisco IP Phones 7960 and 7960G users on foreign exchange office (FXO) lines attached to the Cisco CME system. The flash soft key is enabled using the **fxo hook-flash** command.

Once a flash soft key has been enabled on an IP phone, it is available to provide hookflash functionality during all calls except local IP-phone-to-IP-phone calls. Note that hookflash-controlled services can be activated only if they are supported by the PSTN connection that is involved in the call. The availability of the flash soft key does not guarantee that hookflash-based services are actually accessible to the phone user.

The flash soft key display is automatically disabled for local IP-phone-to-IP-phone calls.

This command must be followed by a quick reboot of the phones using the **restart all** command.

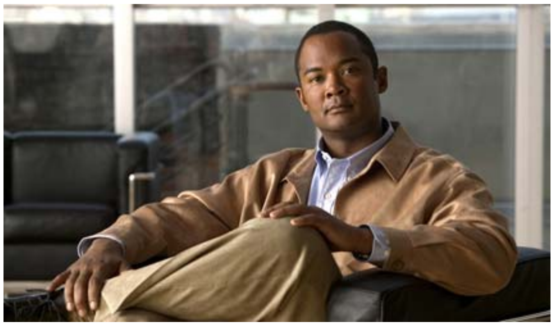
Examples

The following example enables the flash soft key on the Cisco IP Phones 7940 and 7940G and the Cisco IP Phones 7960 and 7960G:

```
Router(config)# telephony-service
Router(config-telephony)# fxo hook-flash
```

Related Commands

Command	Description
restart (ephone)	Performs a fast reboot of a single phone associated with a Cisco CME router.
restart (telephony-service)	Performs a fast reboot of one or all phones associated with a Cisco CME router.
telephony-service	Enters telephony-service configuration mode.



Cisco Unified CME Commands: G

First Published: June 18, 2007

This chapter contains commands to configure and maintain Cisco Unified Communications Manager Express (formally known as Cisco Unified CallManager Express). The commands are presented in alphabetical order. Some commands required for configuring Cisco Unified Communications Manager Express (Cisco Unified CME) may be found in other Cisco IOS command references. Use the command reference master index or search online to find these commands.

gcid

To enable Global Call ID (Gcid) for every call on an outbound leg of a VoIP dial peer for a SIP endpoint, use the **gcid** command in voice-service configuration mode. To return to the default, use the **no** form of this command.

gcid

no gcid

Syntax Description

This command has no arguments or keywords.

Command Default

Gcid is disabled.

Command Modes

Voice-service configuration (config-voi-serv)

Command History

Cisco IOS Release	Modification
12.4(11)XW1	This command was introduced.

Usage Guidelines

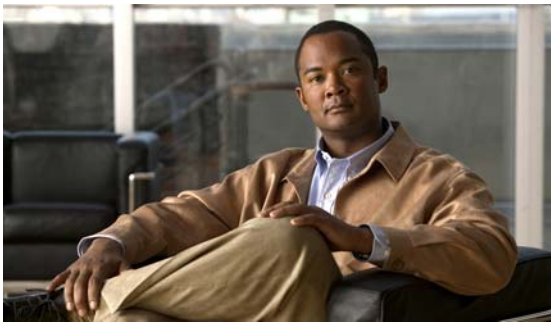
Use this command in voice-service configuration mode to enable Global Call ID (Gcid) in the SIP header for every call on an outbound leg of a VoIP dial peer for a SIP endpoint.

When a call moves around and between the SIP endpoint and the target on a VoIP network because of redirect, transfer, and conference, the SIP Call-ID continues to change. For call control purposes, a unique Gcid is issued for every outbound call leg. A single Gcid remains the same for the same call in the system, and is valid for redirect, transfer, and conference events, including 3-party conferencing when a call center phone acts as a conference host. A SIP header, Cisco_GCID, is added into SIP Invite and REFER requests and to certain other responses to pass the Gcid to the target.

Examples

The following partial output shows the configuration for the **gcid** command:

```
router# show running-configuration
!
!
!
voice service voip
 gcid
 callmonitor
 allow-connections h323 to h323
 allow-connections h323 to sip
 allow-connections sip to h323
 allow-connections sip to sip
 no supplementary-service sip moved-temporarily
 sip
  registrar server expires max 120 min 60
```



Cisco Unified CME Commands: H

Last Updated: June 19, 2006
First Published: February 27, 2006

This chapter contains commands to configure and maintain Cisco Unified Communications Manager Express (formally known as Cisco Unified CallManager Express). The commands are presented in alphabetical order. Some commands required for configuring Cisco Unified Communications Manager Express (Cisco Unified CME) may be found in other Cisco IOS command references. Use the command reference master index or search online to find these commands.

headset auto-answer line

To enable auto-answer on the specified line when the headset key is engaged, use the **headset auto-answer** command in ephone configuration mode. To disable headset auto-answer for this line, use the **no** form of this command.

headset auto-answer line *line-number*

no headset auto-answer line *line-number*

Syntax Description

line-number	Phone line that should be automatically answered.

Defaults

Headset auto-answer is not enabled.

Command Modes

Ephone configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Use this command to enable headset auto-answer on a particular line. A line, as used in this command, is *not* identical to a phone button. A line, as used in this command, represents the ability for a call connection on this phone, and the line numbers generally follow a top-to-bottom sequence starting with the number 1.

The following examples represent common situations pertaining to a button:line relationship:

- button 1:1—A single ephone-dn is associated with a single ephone button. Counts as one line.
- button 1o1,2,3,4,5—Five ephone-dns are overlaid on a single ephone button. Counts as one line.
- button 2x1—An ephone button acts as an extension for an overlaid ephone button. Counts as one line.
- Button is unoccupied or programmed for speed-dial. Does not count as a line.

Examples

The following example shows how to enable headset auto-answer for line 1 (button 1) and line 4 (button 4), which has overlaid ephone-dns but counts as a single line in this context. In this example, four (1, 2, 3, and 4) buttons are defined for ephone 3.

```
ephone 3
 button 1:2 2:4 3:6 4o21,22,23,24,25
 headset auto-answer line 1
 headset auto-answer line 4
```

The following example shows how to enable headset auto-answer for line 2 (button 2), which has overlaid ephone-dns, and line 3 (button 3), which is an overlay rollover line. In this example, three (1, 2, and 3) buttons are defined for ephone 17.

```
ephone 17
 button 1:2 2o21,22,23,24,25 3x2
 headset auto-answer line 2
 headset auto-answer line 3
```

The following example shows how to enable headset auto-answer for line 2 (button 3) and line 3 (button 5). In this case, the button numbers do not match the line numbers because buttons 2 and 4 are not used.

```
ephone 25
 button 1:2 3:4 5:6
 headset auto-answer line 2
 headset auto-answer line 3
```

hold-alert

To set a repeating audible alert notification when a call is on hold on a Cisco Unified IP phone, use the **hold-alert** command in ephone-dn or ephone-dn-template configuration mode. To disable this feature, use the **no** form of this command.

hold-alert *timeout* {**idle** | **originator** | **shared**}

no hold-alert *timeout* {**idle** | **originator** | **shared**}

Syntax Description

timeout	Interval after which an audible alert notification is repeated, in seconds. Range is from 15 to 300. There is no default.
idle	Alerts only when the phone is idle.
originator	Alerts whether the phone is idle or busy.
shared	Alerts only when the extension is idle but alerts all phones that share the line.

Command Default

Audible alert notification for on-hold calls is disabled. Only a visual indication is provided.

Command Modes

Ephone-dn configuration
Ephone-dn-template configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(2)XT	Cisco ITS 2.0	This command was introduced on the following platforms: Cisco 1750, Cisco 1751, Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	Cisco ITS 2.0	This command was implemented on the Cisco 2600-XM and Cisco 2691.
12.2(11)T	Cisco ITS 2.01	This command was implemented on the Cisco 1760.
12.4(4)XC	Cisco Unified CME 4.0	This command was made available in ephone-dn-template configuration mode.
12.4(9)T	Cisco Unified CME 4.0	This command in ephone-dn-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Use the **hold-alert** command to set an audible alert notification on a Cisco Unified IP phone to remind the phone user that a call is on hold. The *timeout* argument specifies the time interval in seconds from the time the call is placed on hold to the time the on-hold audible alert is generated. The alert is repeated every *timeout* seconds.

When the **idle** keyword is enabled, a one-second burst of ringing on the phone is generated on the IP phone that placed the call in the hold state, but only if the phone is in the idle state. If the phone is in active use, no on-hold alert is generated.

When the **originator** keyword is enabled, a one-second burst of ringing is generated on the phone that placed the call in the hold state, but only if the phone is in the idle state. If the phone is in use on another call, an audible beep (call-waiting tone) is generated.

When the **shared** keyword is enabled, a one-second ring burst is generated for all the idle phones that share the extension with the on-hold call. Phones that are in use do not receive an audio beep (call-waiting tone) alert. Only the phone that placed the call on hold hears a call-waiting beep if it is busy.

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

Examples

The following example sets audible alert notification to idle on extension 1111:

```
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 1111
Router(config-ephone-dn)# name phone1
Router(config-ephone-dn)# hold-alert 100 idle
```

The following example uses an ephone-dn template to set audible alert notification for extension 1111 to only occur when the phone is idle:

```
Router(config)# ephone-dn-template 3
Router(config-ephone-dn-template)# hold-alert 100 idle
Router(config-ephone-dn-template)# exit
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 1111
Router(config-ephone-dn)# name phone1
Router(config-ephone-dn)# ephone-dn-template 3
```

Related Commands

Command	Description
ephone-dn	Enters ephone-dn configuration mode.
ephone-dn-template	Enters ephone-dn-template configuration mode.

hold-alert (voice register global)

To set a repeating audible alert notification when a call is on hold on all supported SIP phones directly connected in Cisco Unified CME, use the command in voice register global configuration mode. To disable this feature, use the **no** form of this command.

hold-alert *timeout*

no hold-alert

Syntax Description

timeout	Interval after which an audible alert notification is repeated, in seconds. Range is from 15 to 300. There is no default.

Defaults

Audible alert notification for on-hold calls is disabled. Only a visual indication is provided.

Command Modes

Voice register global configuration

Command History

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

Usage Guidelines

Use this command to set an audible alert notification on all supported SIP phones in a Cisco Unified CME system to remind the phone user that a call is on hold. The alert is repeated after a specific interval as defined by the *timeout* argument.

Note This command does not apply to Cisco ATAs that have been configured for SIP in Cisco Unified CME.

Examples

The following example shows how to set audible alert notification on SIP phones for on-hold calls:

```
Router(config)# voice register global
Router(config-register-global)# mode cme
Router(config-register-global)# hold-alert 30
```

Related Commands

Command	Description
call-waiting (voice register pool)	Enables call waiting on a SIP phone.
mode (voice register global)	Enables the mode for provisioning SIP phones in a Cisco CallManager Express (Cisco CME) system.
voice register global	Enters voice register global configuration mode in order to set global parameters for all supported Cisco SIP phones in a Cisco CME or Cisco SIP SRST environment.

hops

To define the number of times that a call can proceed to the next ephone-dn in a peer or longest-idle ephone hunt group before the call proceeds to the final ephone-dn, use the **hops** command in ephone hunt configuration mode. To return to the default number of hops, use the **no** form of this command.

hops *number*

no hops *number*

Syntax Description

number	Number of hops before the call proceeds to the final ephone-dn. Range is from 2 to 20, but the value must be less than or equal to the number of extensions that are specified in the **list** command. Default automatically adjusts to the number of hunt group members.

Command Default

The number of hops automatically adjusts to the number of ephone hunt group members.

Command Modes

Ephone-hunt configuration

Command Modes

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.3(7)T	Cisco CME 3.1	The maximum number of hops was restricted to the number of extensions specified in the **list** command.
12.3(11)XL	Cisco CME 3.2.1	Increased maximum number of hops to 20.
12.3(14)T	Cisco CME 3.3	This command was integrated into Cisco IOS Release 12.3(14)T.
12.4(4)XC	Cisco Unified CME 4.0	The default was changed from 2 hops to automatically adjust the number of hops to the number of ephone hunt group members.
12.4(9)T	Cisco Unified CME 4.0	The modification to change the default was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is valid only for peer and longest-idle ephone hunt groups in Cisco Unified CallManager Express systems.

This command is required when you are configuring the automatic logout feature for peer and longest-idle hunt groups.

Examples

The following example sets the number of hops to 6 for peer hunt group 3:

```
Router(config)# ephone-hunt 3 peer
Router(config-ephone-hunt)# hops 6
```

Related Commands

Command	Description
ephone-hunt	Defines an ephone hunt group and enters ephone-hunt configuration mode.
final	Defines the last ephone-dn in an ephone hunt group.
list	Defines the ephone-dns that participate in an ephone hunt group.
max-redirect	Changes the current number of allowable redirects in a Cisco Unified CME system.
no-reg (ephone-hunt)	Specifies that the pilot number of this ephone hunt group should not register with the H.323 gatekeeper.
pilot	Defines the ephone-dn that is dialed to reach an ephone hunt group.
preference (ephone-hunt)	Sets preference order for the ephone-dn associated with an ephone-hunt-group pilot number.
timeout (ephone-hunt)	Sets the number of seconds after which a call that is not answered is redirected to the next number in the hunt-group list.

hops (voice hunt-group)

To define the number of times that a call can hop to the next number in a peer hunt group before the call proceeds to the final number, use the **hops** command in voice hunt-group configuration mode. To return to the default number of hops, use the **no** form of this command.

hops *number*

no hops

Syntax Description

number	Number of hops before the call proceeds to the final number. Range is 2 to 10, but the value must be less than or equal to the number of extensions that are specified in the **list** command. The default is the same number as there are destinations defined under the **list** command.

Defaults

The default is the number of *directory-number* arguments configured in the **list** command.

Command Modes

Voice hunt-group configuration

Command History

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

Usage Guidelines

This command is valid only for peer or longest-idle voice hunt groups in Cisco CallManager Express systems.

Examples

The following example shows how to set the number of hops to 6 for peer voice hunt group 1:

```
Router(config)# voice hunt-group 1 peer
Router(config-voice-hunt-group)# list 1000, 1001, 1002, 1003, 1004, 1005, 1006, 006, 1007, 
1008, 1009
Router(config-voice-hunt-group)# hops 6
```

Related Commands

Command	Description
final (voice hunt-group)	Defines the last extension in a voice hunt group.
list (voice hunt-group)	Defines the directory numbers that participate in a hunt group.
timeout (voice hunt-group)	Sets the number of seconds after which a call that is not answered is redirected to the next number in the hunt-group list and defines the last directory number in the hunt group.
voice hunt-group	Defines the type of hunt group.

hunt-group logout

To enable separate handling of DND and HLog functionality for hunt-group agents and the display of the HLog soft key on phones, use the **hunt-group logout** command in telephony-service configuration mode. To return to the default, use the **no** form of this command.

hunt-group logout [**DND** | **HLog**]

no hunt-group logout [**DND** | **HLog**]

Syntax Description

DND	When agent phones do not answer the number of calls specified in the **auto logout** command, they are automatically placed in both DND status and not-ready status. The HLog soft key is not displayed on phones.
HLog	When agent phones do not answer the number of calls specified in the **auto logout** command, they are automatically placed only in not-ready status. The HLog soft key is displayed on phones in addition to the DND soft key.

Command Default

DND and HLog functionality is not separate and the HLog soft key will not be displayed on phones.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

When Do Not Disturb (DND) functionality is activated, no calls are received at the phone, including ephone hunt group calls. DND is activated and canceled using the DND soft key or the DND feature access code (FAC).

When HLog functionality is activated, hunt-group agents are placed in not-ready status and hunt-group calls are blocked from the phone. Other calls that directly dial the phone’s extension numbers are still received at the phone. HLog is activated and canceled using the HLog soft key or an HLog FAC.

If the **auto logout** command is used, the Automatic Agent Status Not-Ready feature is invoked for an ephone hunt group. This feature is triggered when an ephone-dn member does not answer a specified number of ephone hunt group calls. The following actions take place:

- If the **hunt-group logout HLog** command has been used, the agent is placed in not-ready status. The agent’s ephone-dn will not receive further hunt group calls but will receive calls that directly dial the ephone-dn’s extension numbers. An agent in not-ready status can return to ready status by pressing the HLog soft key or by using the HLog FAC.

- If the **hunt-group logout HLog** command has not been used or if the **hunt-group logout DND** command has been used, the phone on which the ephone-dn appears is placed into DND mode, in which the ephone-dn does not receive any calls at all, including hunt-group calls. The red lamp on the phone lights to indicate DND status. An agent in DND mode can return to ready status by pressing the DND soft key or by using the DND FAC.

Note When an agent who is a dynamic member of a hunt group is in not-ready status, the agent's slot in the ephone hunt group is not relinquished. It remains reserved by the agent until the agent leaves the group.

Examples

The following example creates hunt group 3 with three agents (extensions 1001, 1002, and 1003). It specifies that after one unanswered call, an agent should be put into not-ready status but not into DND status.

```
Router(config)# telephony-service
Router(config-telephony)# hunt-group logout HLog
Router(config-telephony)# exit

Router(config)# ephone-hunt 3 peer
Router(config-ephone-hunt)# pilot 4200
Router(config-ephone-hunt)# list 1001, 1002, 1003
Router(config-ephone-hunt)# timeout 10
Router(config-ephone-hunt)# auto logout
Router(config-ephone-hunt)# final 4500
```

Related Commands

Command	Description
auto logout	Enables the automatic change of an agent's ephone-dn to not-ready status after a specified number of hunt-group calls are not answered.

hunt-group report delay hours

To delay the automatic transfer of Cisco CallManager Express (Cisco CME) basic automatic call distribution (B-ACD) call statistics to a file, use the **hunt-group report delay hours** command in telephony-service configuration mode. To remove to the delay setting, use the **no** form of this command.

hunt-group report delay *number* **hours**

no hunt-group report delay *number* **hours**

Syntax Description

number	Number of hours by which the collection of statistics can be extended for the statistics collection periods configured with the **hunt-group report every hours** command. The range is from 1 to 10.

Defaults

No hunt-group report delay is configured.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.3(11)XL	3.2.1	This command was introduced.
12.3(14)T	3.3	This command was integrated into Cisco IOS Release 12.3(14)T.

Usage Guidelines

This command is used for Cisco CME basic automatic call distribution (B-ACD) and auto-attendant (AA) service only.

The **hunt-group report delay hours** command is used as part of a statistics reporting configuration that allows Cisco CME B-ACD call statistics to be sent automatically to files using TFTP. For detailed information, see *Cisco CME B-ACD and Tcl Call-Handling Applications*.

Statistics are collected and stored (**statistics collect** command and **hunt-group report url** command) in specified intervals (**hunt-group report every hours** command). The default is for the statistics to be collected one hour after the specified interval. Because calls are counted when they end, some of the longer calls may not be counted. For example, if there is a call from 1:35 p.m. to 3:30 p.m., the interval is 1 hour, and there is no delay, TFTP will write the 1 p.m. to 2 p.m. statistics at 3 p.m. However, at 3 p.m., the 1:35 p.m. call is still active, so the call will not be counted at that time as occurring in the 1 p.m. to 2 p.m. time slot. When the call finishes at 3:30 p.m., it will be counted as occurring from 1 p.m. to 2 p.m. The **show hunt-group** command will report it, but TFTP will have already sent out its report. To include the 1:35 p.m. call, you could use the **hunt-group report delay hours** command to delay TFTP statistics reporting for an extra hour so the 1 p.m. to 2 p.m. report will be written at 4 p.m. instead of 3 p.m.

Examples

The following example shows a configuration in which statistics are reported for B-ACD calls that occur within three-hour time frames, but the collection of the statistic collection is extended for an extra hour to include calls that did not end within the three-hour time period:

```
Router(config)# telephony-service
Router(config-telephony)# hunt-group report every 3 hours
Router(config-telephony)# hunt-group report delay 1 hours
```

The following is an example of a report that the previous configuration might send to a file if the **statistics collect** command was entered at 18:20:

```
23:00:00 UTC Tue Dec 20 2004,
,
01, Tue 18:00 - 19:00, HuntGp, 02, 01, 00005, 00002, 0003, 0006, 000001, 000001, 0011,
01, Tue 19:00 - 20:00, HuntGp, 02, 02, 00000, 00000, 0000, 0000, 000000, 000000, 0000,
01, Tue 20:00 - 21:00, HuntGp, 02, 02, 00006, 00003, 0003, 0009, 000001, 000003, 0012,
```

Statistics collection has to take place for at least three hours for the statistics to be written to a file. The following is a chronology of events:

- At 19:00, the statistics collection was active for 40 minutes, so no statistics were written to file.
- At 20:00, the statistics collection was active for 1 hour and 40 minutes, so no statistics were written to file.
- At 21:00, the statistics collection was active for 2 hours and 40 minutes, so no statistics were written to file.
- At 22:00, the statistics collection was active for 3 hours and 40 minutes but there is a one-hour delay, so no statistics were written to file.
- At 23:00 the statistics were written to a file using TFTP.

Related Commands

Command	Description
ephone-hunt	Defines an ephone hunt group and enters ephone-hunt configuration mode.
hunt-group report every hours	Sets the hourly interval after which Cisco CME B-ACD call statistics are automatically transferred to a file.
hunt-group report url	Sets filename parameters and the URL path where Cisco CME B-ACD call statistics are to be sent using TFTP.
statistics collect	Enables the collection of Cisco CME B-ACD call data for an ephone hunt group.

hunt-group report every hours

To set the hourly interval at which Cisco CallManager Express (Cisco CME) basic automatic call distribution (B-ACD) call statistics are automatically transferred to a file, use the **hunt-group report every hours** command in telephony-service configuration mode. To remove the interval setting, use the **no** form of this command.

hunt-group report every *number* **hours**

no hunt-group report every *number* **hours**

Syntax Description

number	Number of hours after which auto-attendant (AA) call statistics are collected and reported. The range is from 1 to 84.

Defaults

No hourly interval is configured.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.3(11)XL	3.2.1	This command was introduced.
12.3(14)T	3.3	This command was integrated into Cisco IOS Release 12.3(14)T.

Usage Guidelines

This command is used for Cisco CME basic automatic call distribution (B-ACD) and auto-attendant (AA) service only.

The **hunt-group report every hours** command is used as part of a statistics reporting configuration that allows Cisco CME B-ACD call statistics to be sent automatically to files by means of TFTP. For detailed information, see *Cisco CME B-ACD and Tcl Call-Handling Applications*.

Because calls are counted when they end, some of the longer calls may not be counted in the report. To delay the time in which statistics are collected and transferred you may configure a delay time with the **hunt-group report delay hours** command.

Examples

The following example sets the statistics collection to occur every three hours. There is no delay.

```
Router(config)# telephony-service
Router(config-telephony)# hunt-group report every 3 hours
```

The following is an example of a report that the previous configuration might send to a file if the **statistics collect** command was entered at 18:20:

```
22:00:00 UTC Tue Dec 20 2005,
,
01, Tue 18:00 - 19:00, HuntGp, 02, 01, 00005, 00002, 0003, 0006, 000001, 000001, 0011,
01, Tue 19:00 - 20:00, HuntGp, 02, 02, 00000, 00000, 0000, 0000, 000000, 000000, 0000,
```

```
01, Tue 20:00 - 21:00, HuntGp, 02, 02, 00006, 00003, 0003, 0009, 000001, 000003, 0012,
```

Statistics collection has to take place for at least three hours for the statistics to be written to a file. The following is a chronology of events:

- At 19:00, the statistics collection was active for 40 minutes, so no statistics were written to file.
- At 20:00, the statistics collection was active for 1 hour and 40 minutes, so no statistics were written to file.
- At 21:00, the statistics collection was active for 2 hours and 40 minutes, so no statistics were written to file.
- At 22:00, the statistics collection was active for 3 hours and 40 minutes, so statistics were written to a file using TFTP.

If the previous example were configured for a delay of one hour using the **hunt-group report delay 1 hours** command, the statistics would be written one hour later at 23:00.

Related Commands

Command	Description
ephone-hunt	Defines an ephone hunt group and enters ephone-hunt configuration mode.
hunt-group report delay hours	Delays the automatic transfer of Cisco CME B-ACD call statistics to a file.
hunt-group report url	Sets filename parameters and the URL path where Cisco CME B-ACD call statistics are to be sent using TFTP.
statistics collect	Enables the collection of Cisco CME B-ACD call statistics for an ephone hunt group.

hunt-group report url

To set filename parameters and the URL path where Cisco Unified CME basic automatic call distribution (B-ACD) call statistics are to be sent using TFTP, use the **hunt-group report url** command in telephony-service configuration mode. To remove the report URL settings and stop statistics from being sent to files, use the **no** form of this command.

hunt-group report url [**prefix tftp://***ip-address***/***directory-name...***/***prefix* | **suffix** *from-number* **to** *to-number*]

no hunt-group report url [**prefix tftp://***ip-address***/***directory-name...***/***prefix* | **suffix** *from-number* **to** *to-number*]

Syntax Description

prefix	Sets the parameters for how the filenames must start.
tftp://*ip-address-path***/**	IP address to the files where AA call data is sent using TFTP.
*directory-name...***/**	Names of directories, separated by forward slashes (/) to declare the path to the files where AA call data is sent.
prefix	Specifies a common beginning to be used for the filenames.
suffix	Sets numeric parameters for unique endings for the filenames.
from-number	Number at which the suffix range starts. The range is from 0 to 1. There is no default.
to *to-number*	Number at which the suffix range ends. The range is from 1 to 200. There is no default.

Command Default

No statistics are sent to files.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)XL	Cisco CME 3.2.1	This command was introduced.
12.3(14)T	Cisco CME 3.3	This command was integrated into Cisco IOS Release 12.3(14)T.
12.4(4)XC	Cisco Unified CME 4.0	Call-hold statistics were added to the output for Cisco Unified CME B-ACD hunt groups.
12.4(9)T	Cisco Unified CME 4.0	Call-hold statistics in the output for Cisco Unified CME B-ACD hunt groups was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Use this command for Cisco Unified CME basic automatic call distribution (B-ACD) and auto-attendant (AA) service only. For detailed information, see *Cisco Unified CME B-ACD and Tcl Call-Handling Applications*.

The **hunt-group report url** command is used with the **hunt-group every hour** command to collect statistics about ephone hunt groups that are part of Cisco Unified CME B-ACD services. Data is collected for all agents combined and for individual agents. The data includes statistics on the number of calls received, the amount of time the calls had to wait to be answered, the amount of time they spent on hold or in queue, and so forth.

The **hunt-group report url** command transfers these call statistics to files using TFTP for time periods set by the **hunt-group every hours** command. Each set of statistics for each time period is sent to a different file that is named using the arguments in the **hunt-group report url** command. For example, the first set of statistics may go to a file named test001, the second set to test002, and so forth.

Prior to using the **hunt-group report url** command, you must create files with matching prefixes and suffixes. For example, for the following configuration:

```
telephony-service
 hunt-group report url prefix tftp://239.1.1.1/dirname/dirname/data
 hunt-group report url suffix 0 to 3
```

you must have files named data0, data1, data2, and data3 at the designated directory location (tftp://239.1.1.1/dirname/dirname).

For the following configuration, you must have files named data00, data01, data02, ... data50:

```
telephony-service
 hunt-group report url prefix tftp://239.1.1.1/dirname/dirname/data
 hunt-group report url suffix 0 to 50
```

For the following configuration, you must have files named data000, data002, ... data200:

```
telephony-service
 hunt-group report url prefix tftp://239.1.1.1/dirname/dirname/data
 hunt-group report url suffix 0 to 200
```

The files must be must empty read-and-write files. The following is an example of the statistics sent to a file using TFTP:

```
23:00:00 UTC Wed Apr 23 2003,

01, Wed 21:00 - 22:00, HuntGp, 02, 02, 00005, 00002, 0003, 0006, 000001, 000001, 0011,
01, Wed 21:00 - 22:00, Agent, 8001, 00002, 000001, 000001, 00002, 000002, 000002,
01, Wed 21:00 - 22:00, Agent, 8003, 00001, 000001, 000001, 00000, 000000, 000000,
01, Wed 21:00 - 22:00, Queue, 00002, 00002, 00000, 00002, 00003, 00000, 00000, 00000,
00000,
```

The order of the data fields corresponds to the order of the descriptions issued by the **show hunt-group** command. See the "Examples" section for explanations of the data fields. The *Cisco CME B-ACD and Tcl Call-Handling Applications* document discusses how hunt-group reports align with the **show hunt-group** command output. Once the statistics are in a file, they can be sent to an application, such as Microsoft Excel or Access, to be merged into a chart or graph for easier reading.

For the report mechanism to collect data, you must first issue the **statistics collect** command.

Examples

The following configuration uses TFTP to send AA call statistics to files named test00, test01, ... test90 located at tftp://239.1.1.1/dirname/dirname/test:

```
Router(config)# telephony-service
Router(config-telephony)# hunt-group report url prefix
tftp://239.1.1.1/dirname/dirname/test
Router(config-telephony)# hunt-group report url suffix 0 to 90
```

The following example displays the raw data output that was transferred to files in TFTP format after the **hunt-group report every hours** command was used. Table 9 through Table 11 describe what each number in the example represents. Table 9 explains the first line of data, Table 10 explains the second and third lines of data, and Table 11 explains the fourth line of data.

```
18:00:00 UTC Tue Apr 23 2003,
,
01, Tue 16:00 - 17:00, HuntGp, 06, 06, 00002, 00002, 00000, 0006, 0011, 000004, 000006,
0000, 00002, 000002, 000005,
01, Tue 16:00 - 17:00, Agent, 8001, 00001, 000002, 000002, 00001, 000003, 000003, 00002,
000003, 000003, 00002, 000001, 000001,
01, Tue 16:00 - 17:00, Agent, 8003, 00001, 000006, 000006, 00001, 000005, 000005, 00000,
000000, 000000, 00000, 000000, 000000,
01, Tue 16:00 - 17:00, Queue, 00002, 00002, 00000, 00001, 00001, 00000, 00000, 00000,
00000,
```

Table 9 explains the first line of TFTP-format statistics, which are the main statistics that present data for the hunt group as a whole.

Table 9 Ephone Hunt Group TFTP Format Main Statistics

Hunt Group Data Line Example (Entire Hunt Group):

```
01, Tue 16:00 - 17:00, HuntGp, 06, 06, 00002, 00002, 00000, 0006, 0011, 000004,
000006, 0000, 00002, 000002, 000005,
```

Data	Explanation
01	Statistics for hunt group 1 are provided in this line of data.
Tue 16:00 - 17:00	Period during which the statistics were collected.
HuntGp	Main statistics for a complete hunt group are provided in this line of data.
06	Maximum number of agents.
06	Minimum number of agents.
00002	Total calls.
00002	Answered calls.
00000	Abandoned calls.
0006	Average time to answer, in seconds.
0011	Longest time to answer, in seconds.
000004	Average time in call, in seconds.
000006	Longest time in call, in seconds.
0000	Average time before abandonment, in seconds.
00002	Calls on hold.
000002	Average time on hold, in seconds.
000005	Longest time on hold, in seconds.

Table 10 explains the next two lines of TFTP-format statistics in the example, which provide data for individual agents. Note that only the second line is presented in the table, but the third line follows the same format.

In the table, some statistics are marked with the following comments.

- Direct—Indicates calls that were made directly to the hunt group pilot number.
- Queue—Indicates calls that passed through a Cisco Unified CME B-ACD call queue.

Table 10 Ephone Hunt Group TFTP Format Per-Agent Statistics

Agent Data Line Example:

```
01, Tue 16:00 - 17:00, Agent, 8001, 00001, 000002, 000002, 00001, 000003, 000003,
00002, 000003, 000003, 00002, 000001, 000001,
```

Data	Explanation
01	Statistics for hunt group 1 are provided in this line of data.
Tue 16:00 - 17:00	Period during which these statistics were collected.
Agent	Hunt group statistics for a single agent are provided in this line of data.
8001	Agent number.
00001	Total calls answered (Direct).
000002	Average time in call, in seconds (Direct).
000002	Longest time in call, in seconds (Direct).
00001	Total calls on hold (Direct).
000003	Average hold time, in seconds (Direct).
000003	Longest hold time, in seconds (Direct).
00002	Total calls answered (Queue).
000003	Average time in call, in seconds (Queue).
000003	Longest time in call, in seconds (Queue).
00002	Total calls on Hold (Queue).
000001	Average hold time, in seconds (Queue).
000001	Longest hold time, in seconds (Queue).

Table 11 explains the final line of data in the example, which is the data for the B-ACD queue.

Table 11 Ephone Hunt Group TFTP Format Queue-Related Statistics

Queue-Related Data Line Example:

```
01, Tue 16:00 - 17:00, Queue, 00002, 00002, 00000, 00001, 00001, 00000, 00000,
00000, 00000,
```

Data	Explanation
01	Statistics for hunt group 1 are provided in this line of data.
Tue 16:00 - 17:00	Period during which these statistics were collected.
Queue	Queue-related statistics are provided in this line of data.
00002	Total number of calls presented to the queue.
00002	Calls answered by agents.
00000	Number of calls in the queue.
00001	Average time to answer, in seconds.
00001	Longest time to answer, in seconds.
00000	Number of abandoned calls.
00000	Average time before abandonment, in seconds.
00000	Calls forwarded to voice mail.
00000	Calls answered by voice mail.

Related Commands

Command	Description
ephone-hunt	Defines an ephone hunt group and enters ephone-hunt configuration mode.
hunt-group report delay hours	Delays the automatic transfer of Cisco Unified CME B-ACD call statistics to a file.
hunt-group report every hours	Sets the hourly interval after which Cisco Unified CME B-ACD call statistics are automatically transferred to a file.
statistics collect	Enables the collection of Cisco Unified CME B-ACD call statistics for an ephone hunt group.

huntstop (ephone-dn and ephone-dn-template)

To discontinue call hunting behavior for an extension (ephone-dn) or an extension channel, use the **huntstop** command in ephone-dn or ephone-dn-template configuration mode. To disable huntstop, use the **no** form of this command.

huntstop [**channel**]

no huntstop [**channel**]

Syntax Description

channel	(Optional) For dual-line ephone-dns, keeps incoming calls from hunting to the second channel if the first channel is busy or does not answer.

Command Default

Ephone-dn huntstop is enabled.
Channel huntstop is disabled.

Command Modes

Ephone-dn configuration
Ephone-dn-template configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco ITS 1.0	This command was introduced on the following platforms: Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(2)XT	Cisco ITS 2.0	This command was implemented on the Cisco 1750 and Cisco 1751.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	Cisco ITS 2.0	This command was implemented on the Cisco 2600-XM and Cisco 2691.
12.2(11)T	Cisco ITS 2.0.1	This command was implemented on the Cisco 1760.
12.2(15)ZJ	Cisco CME 3.0	The **channel** keyword was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.4(4)XC	Cisco Unified CME 4.0	This command was made available in ephone-dn-template configuration mode.
12.4(9)T	Cisco Unified CME 4.0	This command in ephone-dn-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

When you use the **huntstop** command without the **channel** keyword, it affects call hunting behavior that relates to ephone-dns (lines or extensions). If the huntstop attribute is set, an incoming call does not roll over (hunt) to another ephone-dn if the called ephone-dn is busy or does not answer and a hunting strategy has been established that includes this ephone-dn. A huntstop allows you to prevent hunt-on-busy from redirecting a call from a busy phone into a dial-peer setup with a catch-all default destination. Use the **no huntstop** command to disable huntstop and allow hunting for ephone-dns.

Channel huntstop works in a similar way, but it affects call hunting behavior for the two channels of a single dual-line ephone-dn. If the **huntstop channel** command is used, incoming calls do not hunt to the second channel of an ephone-dn if the first channel is busy or does not answer. For example, an incoming call might search through the following ephone-dns and channels:

ephone-dn 10 (channel 1)
ephone-dn 10 (channel 2)

ephone-dn 11 (channel 1)
ephone-dn 11 (channel 2)
ephone-dn 12 (channel 1)
ephone-dn 12 (channel 2)

When the **no huntstop channel** command is used (the default), you might have a call ring for 30 seconds on ephone-dn 10 (channel 1) and then after 30 seconds move to ephone-dn 10 (channel 2). This is usually not the behavior that you desire. Also, it is often useful to reserve the second channel of a dual-line ephone-dn for call transfer, call waiting, or conferencing. The **huntstop channel** command tells the system that if the first channel is in use or does not answer, an incoming call should hunt forward to the next ephone-dn in the hunt sequence instead of to the next channel on the same ephone-dn.

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

Examples

The following example shows how to disable huntstop for the destination dial peer with the extension 5001. The huntstop for the dial peer is set to OFF and prevents calls to extension 5001 from being rerouted to the on-net H.323 dial peer for the 5... destination when 5001 is busy (the three periods are used as wild cards).

```
ephone-dn 1
 number 5001
 no huntstop
```

The following example shows a typical configuration in which ephone-dn huntstop (default) is required:

```
ephone-dn 1
 number 5001

ephone 4
 button 1:1
 mac-address 0030.94c3.8724

dial-peer voice 5000 voip
 destination-pattern 5...
 session target ipv4:192.168.17.225
```

In the previous example, the huntstop attribute is set to ON by default and prevents calls to extension 5001 from being rerouted to the on-net H.323 dial peer for 5... when extension 5001 is busy.

The next example shows another instance in which huntstop is not desired and is explicitly disabled. In this example, ephone 4 is configured with two lines, each with the same extension number 5001. This is done in order to allow the second line to provide call-waiting notification for extension number 5001 when the first line is in use. Setting **no huntstop** on the first line (ephone-dn 1) allows incoming calls to hunt to the second line (ephone-dn 2) on ephone 4 when the ephone-dn 1 line is busy.

Ephone-dn 2 has call forwarding set to extension 6000, which corresponds to a locally attached answering machine connected to a foreign exchange station (FXS) voice port. In this example, the plain old telephone system (POTS) dial peer for extension 6000 also has the dial-peer huntstop attribute explicitly set to prevent further hunting.

```
ephone-dn 1
 number 5001
 no huntstop
 preference 1
 call-forward noan 6000

ephone-dn 2
 number 5001
 preference 2
 call-forward busy 6000
 call-forward noan 6000

ephone 4
 button 1:1 2:2
 mac-address 0030.94c3.8724

dial-peer voice 6000 pots
 destination-pattern 6000
 huntstop
 port 1/0/0
 description answering-machine
```

The next example shows a dual-line ephone-dn configuration in which calls do not hunt to the second channel of any ephone-dn, but they do hunt through the channel 1 for each ephone-dn in the order 10, 11, 12.

```
ephone-dn 10 dual-line
 number 1001
 no huntstop
 huntstop channel

ephone-dn 11 dual-line
 number 1001
 no huntstop
 huntstop channel
 preference 1

ephone-dn 12 dual-line
 number 1001
 no huntstop
 huntstop channel
 preference 2
```

The next example uses an ephone-dn-template in a dual-line ephone-dn configuration to keep calls from hunting to the second channel of any ephone-dn. The calls do hunt through the first channels for each ephone-dn in the order 10, 11, 12.

```
ephone-dn-template 2
 huntstop channel

ephone-dn 10 dual-line
 number 1001
 no huntstop
 ephone-dn-template 2

ephone-dn 11 dual-line
 number 1001
 no huntstop
 ephone-dn-template 2
 preference 1

ephone-dn 12 dual-line
 number 1001
 no huntstop
 ephone-dn-template 2
 preference 2
```

Related Commands

Command	Description
ephone-dn	Enters ephone-dn configuration mode.
ephone-dn-template	Enters ephone-dn-template configuration mode.
huntstop (dial-peer)	Disables all further dial-peer hunting if a call fails using hunt groups.

huntstop (voice register dn)

To disable call hunting behavior for an extension on a SIP phone, use the **huntstop** command in voice register dn configuration mode. To return to the default, use the **no** form of this command.

huntstop

no huntstop

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled

Command Modes

Voice register dn configuration

Command History

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

Usage Guidelines

If this command is enabled, an incoming call does not roll over (hunt) to another directory number if the called directory number is busy, or does not answer, and a hunting strategy has been established that includes this directory number. A huntstop allows you to prevent hunt-on-busy from redirecting a call from a busy phone into a dial-peer setup with a catch-all default destination. Use the **no huntstop** command to disable huntstop and allow hunting for directory numbers (default).

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

Examples

The following example shows a typical configuration in which huntstop is required. The **huntstop** command is enabled and prevents calls to extension 5001 from being rerouted to the on-net H.323 dial peer for 5... when extension 5001 is busy (three periods are used as wild cards).

```
voice register dn 1
 number 5001
 huntstop

voice register pool 4
 button 1:1
 mac-address 0030.94c3.8724

dial-peer voice 5000 voip
 destination-pattern 5...
 session target ipv4:192.168.17.225
```

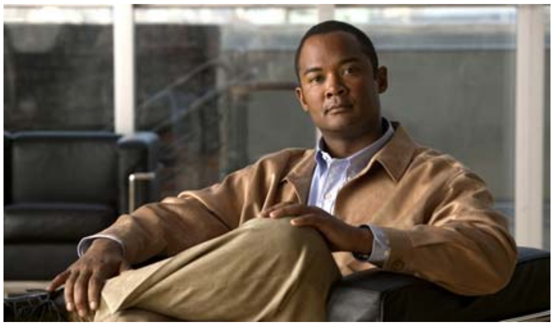
The next example shows an example in which huntstop is not desired (default). In this example, directory number 4 is configured with two lines, each with the same extension number 5001. This is done to allow the second line to provide call-waiting notification for extension number 5001 when the first line is in use. Not enabling huntstop on the first line (directory number 1) allows incoming calls to hunt to the second line (directory number 2) on phone 4 when the directory number 1 line is busy.

directory number 2 has call forwarding set to extension 6000, which corresponds to a locally attached answering machine connected to a foreign exchange station (FXS) voice port. In this example, the plain old telephone system (POTS) dial peer for extension 6000 also has the dial-peer huntstop attribute explicitly set to prevent further hunting.

```
voice register dn 1
 number 5001
 preference 1
 call-forward noan 6000

voice register dn 2
 number 5001
 preference 2
 call-forward busy 6000
 call-forward noan 6000

voice register pool 4
 button 1:1 2:2
 mac-address 0030.94c3.8724

dial-peer voice 6000 pots
 destination-pattern 6000
 huntstop
 port 1/0/0
 description answering-machine
```

Related Commands

Command	Description
voice register dn	Enters voice register dn configuration mode to define an extension for a SIP phone line.
huntstop (dial-peer)	Disables all further dial-peer hunting if a call fails on the dial peer.



Cisco Unified CME Commands: I

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This chapter contains commands to configure and maintain Cisco Unified Communications Manager Express (formally known as Cisco Unified CallManager Express). The commands are presented in alphabetical order. Some commands required for configuring Cisco Unified Communications Manager Express (Cisco Unified CME) may be found in other Cisco IOS command references. Use the command reference master index or search online to find these commands.

id (voice register pool)

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

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

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

Syntax Description

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

Command Default

None

Command Modes

Voice register pool configuration

Command History

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

Usage Guidelines

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

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

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

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

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

Examples

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

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

Related Commands

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

intercom (ephone-dn)

To create an intercom by programming a pair of extensions (ephone-dns) to automatically call and answer each other, use the **intercom** command in ephone-dn configuration mode. To remove an intercom, use the **no** form of this command.

intercom *extension-number* [**barge-in** [**no-mute**] [**label** *label*]] [**no-auto-answer**] [**label** *label*] [**no-mute**]

no intercom *extension-number*

Syntax Description

extension-number	Telephone number to which intercom calls are placed.
barge-in	(Optional) Allows inbound intercom calls to force an existing call into the call-hold state and allows the intercom call to be answered immediately.
label *label*	(Optional) Defines an alphanumeric label for the intercom, of up to 30 characters.
no-auto-answer	(Optional) Disables the intercom auto-answer feature.
no-mute	Allows an intercom call to be answered without deactivating a speaker's mute key.

Defaults

Intercom functionality is disabled.

Command Modes

Ephone-dn configuration

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.2(2)XT	2.0	This command was introduced on the following platforms: Cisco 1750, Cisco 1751, Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(8)T	2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	2.0	This command was implemented on the Cisco 2600-XM and Cisco 2691.
12.2(11)T	2.01	This command was implemented on the Cisco 1760.
12.3(11)XL	3.2.1	The **no-mute** keyword was added.
12.3(14)T	3.3	This command was integrated into Cisco IOS Release 12.3(14)T.

Usage Guidelines

This command is used to dedicate a pair of Cisco ephone-dns for use as a “press to talk” two-way intercom between Cisco IP phones. Intercom lines cannot be used in shared-line configurations. If an ephone-dn is configured for intercom operation, it must be associated with one Cisco IP phone only. The intercom attribute causes an IP extension (ephone-dn) to operate in autodial fashion for outbound calls and auto answer-with-mute for inbound calls.

The **barge-in** keyword allows inbound intercom calls to force an existing call on the called phone into the call-hold state to allow the intercom call to be answered immediately. The **no-auto-answer** keyword creates for the IP phone line a connection that resembles a private line, automatic ringdown (PLAR). The **label** keyword defines a text label for the intercom.

Following this command, the intercom ephone-dns are assigned to ephones using the **button** command. Following the **button** command, the **restart** command must be used to initiate a quick reboot of the phones to which this intercom is assigned.

The default **intercom** command behavior is speakers are set to mute automatically when phones receive intercom calls. For example, if phone user 1 places an intercom call and connects to phone user 2, user 2 will hear user 1, but user 1 will not hear user 2. To be heard, user 2 must first disable the speaker’s mute function. The benefit is people who receive intercom calls can use the mute button to control when they will be heard initially.

The **no-mute** keyword deactivates the speaker mute function when IP phones receive intercom calls. For example, if phone user 1 makes an intercom call to phone user 2, both users will hear each other upon connection. The benefit is that people who receive intercom calls do not have to disable their speaker’s mute function to be heard, *but* their conversations and nearby background sounds will be heard the moment an intercom call to them is connected—regardless of whether they are ready to take a call or not.

Examples

The following example sets the intercom on Cisco IP phone directory number 1:

```
Router(config)# ephone-dn 1
Router(config-ephone-dn) number A5001
Router(config-ephone-dn) name “intercom”
Router(config-ephone-dn) intercom A5002 barge-in
```

The following example shows intercom configuration between two Cisco IP phones:

```
ephone-dn 18
 number A5001
 name “intercom”
 intercom A5002 barge-in
ephone-dn 19
 number A5002
 name “intercom”
 intercom A5001 barge-in
ephone 4
 button 1:2 2:4 3:18
 ephone 5
 button 1:3 2:6 3:19
```

In the example, ephone-dn 18 and ephone-dn 19 are set as an intercom pair. Ephone-dn 18 is associated with button 3 of Cisco IP phone (ephone) 4, and ephone-dn 19 is associated with button number 3 of Cisco IP phone (ephone) 5. Button 3 on Cisco IP phone 4 and button 3 on Cisco IP phone 5 are set as a pair to provide intercom service to each other.

The intercom feature acts as a combination speed-dial PLAR and auto answer-with-mute. If the **barge-in** keyword is set on the ephone-dn that receives the intercom call, the existing call is forced into the hold state, and the intercom call is accepted. If the phone user has the handset off hook (that is, not in speakerphone mode), the user hears a warning beep, and the intercom call is immediately connected with two-way audio. If the phone user is using speakerphone mode, the intercom connects with the microphone mute activated.

Note Any caller can dial in to an intercom extension, and a call to an intercom extension that is originated by a nonintercom caller triggers an automatic answer exactly like a legitimate intercom call. To prevent nonintercom originators from manually dialing an intercom destination, you can use alphabetic characters when you assign numbers to intercom extensions using the **number** command. These characters cannot be dialed from a normal phone but can be dialed by preprogrammed intercom extensions whose calls are made by the router.

Related Commands

Command	Description
button	Associates ephone-dns with individual buttons on Cisco IP phones and specifies ring behavior per button.
ephone-dn	Enters ephone-dn configuration mode.
number	Associates a telephone or extension number with an extension (ephone-dn).
restart (ephone)	Performs a fast reboot of a single phone associated with a Cisco CME router.
restart (telephony-service)	Performs a fast reboot of one or all phones associated with a Cisco CME router.

ip source-address (credentials)

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

ip source-address *ip-address* [**port** [*port*]]

no ip source-address

Syntax Description

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

Command Default

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

Command Modes

Credentials configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(14)T	Cisco SRST 3.3	This command was introduced for Cisco SRST.
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced for Cisco Unified CME.
12.4(9)T	Cisco Unified CME 4.0	This command for Cisco Unified CME was integrated in Cisco IOS Release 12.4(9)T.

Usage Guidelines

Cisco Unified CME

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

Cisco Unified SRST

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

Examples

Cisco Unified CME

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

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

Cisco Unified SRST

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

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

Related Commands

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

ip source-address (telephony-service)

To identify the IP address and port through which IP phones communicate with a Cisco Unified CME router, use the **ip source-address** command in telephony-service configuration mode. To disable the router from receiving messages from Cisco Unified IP phones, use the **no** form of this command.

ip source-address *ip-address* [**port** *port*] [**secondary** *ip-address* [**rehome** *seconds*]] [**any-match** | **strict-match**]

no ip source-address

Syntax Description

ip-address	Preexisting router IP address, typically one of the addresses of the Ethernet port of the router.
port *port*	(Optional) TCP/IP port number to use for Skinny Client Control Protocol (SCCP). Range is from 2000 to 9999. Default is 2000.
secondary	(Optional) Second Cisco Unified CME router with which phones can register if the primary Cisco Unified CME router fails.
rehome *seconds*	(Optional) Used only by Cisco Unified IP phones that have registered with a Cisco Unified SRST router. This keyword defines a delay that is used by phones to verify the stability of their primary SCCP controller (Cisco Unified CallManager or Cisco Unified CME) before the phones reregister with it. This parameter is ignored by phones unless they are registered to a secondary Cisco Unified SRST router. The range is from 0 to 65535 seconds. The default is 120 seconds. **Note** The use of this parameter is a phone behavior and is subject to change, based on the phone type and phone firmware version.
any-match	(Optional) Disables strict IP address checking for registration. This is the default.
strict-match	(Optional) Requires strict IP address checking for registration.

Command Default

IP address for communicating with phones is not defined.

Command Modes

Telephony-service configuration

Command History

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

Cisco IOS Release	Cisco Product	Modification
12.2(8)T1	Cisco ITS 2.0	This command was implemented on the Cisco 2600-XM and Cisco 2691.
12.2(11)T	Cisco ITS 2.01	This command was implemented on the Cisco 1760.
12.4(4)XC	Cisco Unified CME 4.0	The **secondary** *ip-address* and **rehome** *seconds* keyword-argument pairs were added.

Usage Guidelines

Note Prior to version 4.0, the name of this product was Cisco CallManager Express (Cisco CME). Prior to version 3.0, the name was Cisco IOS Telephony Services (Cisco ITS).

This mandatory command enables a router to receive messages from Cisco Unified IP phones through the specified IP address and port. The Cisco Unified CME router cannot communicate with the Cisco Unified CME phones if the IP address is not provided. If the port number is not provided, the default is port 2000. The IP address is usually the IP address of the Ethernet port to which the phones are connected.

Use the **any-match** keyword to instruct the router to permit Cisco Unified IP phone registration, and use the **strict-match** keyword to instruct the router to reject IP phone registration attempts if the IP server address used by the phone does not match the source address exactly.

Prior to Cisco IOS Telephony Services (Cisco ITS) V2.1, this command helped the router to automatically generate the SEPDEFAULT.cnf file, which was stored in the flash memory of the router. The SEPDEFAULT.cnf file contains the IP address of one of the Ethernet ports of the router to which the phone should register. In ITS V2.1, Cisco CME 3.0, and later versions, the configuration files have been moved to system:/its/. The file named Flash:SEPDEFAULT.cnf that was used with previous Cisco ITS versions is now obsolete, but is retained as system:/its/SEPDEFAULT.cnf to support upgrades from older phone firmware.

For systems using Cisco ITS V2.1, Cisco CME 3.0, or later versions, the IP phones receive their initial configuration information and phone firmware from the TFTP server associated with the router. In most cases, the phones obtain the IP address of their TFTP server using the **option 150** command and Dynamic Host Configuration Protocol (DHCP). For Cisco ITS or Cisco CME operation, the TFTP server address obtained by the Cisco Unified IP phones should point to the router IP address. The Cisco IP phones attempt to transfer a configuration file called XmlDefault.cnf.xml. This file is automatically generated by the router through the **ip source-address** command and is placed in router memory. The XmlDefault.cnf.xml file contains the IP address that the phones use to register for service, using the Skinny Client Control Protocol (SCCP). This IP address should correspond to a valid Cisco CME router IP address (and may be the same as the router TFTP server address).

Similarly, when an analog telephone adapter (ATA) such as the ATA-186 is attached to the Cisco Unified CME router, the ATA receives very basic configuration information and firmware from the TFTP server XmlDefault.cnf.xml file. The XmlDefault.cnf.xml file is automatically generated by the Cisco Unified CME router with the **ip source-address** command and is placed in the router's flash memory.

By specifying a second Cisco Unified CME router in the **ip source-address** command, you improve the failover time for phones.

Examples

The following example sets the IP source address and port:

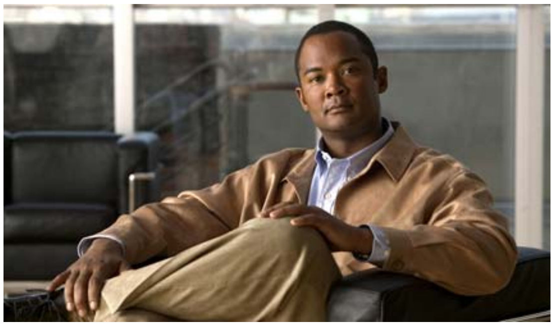
```
Router(config)# telephony-service
Router(config-telephony)# ip source-address 10.6.21.4 port 2000 strict-match
```

The following example establishes the router at 10.5.2.78 as a secondary router:

```
Router(config)# telephony-service
Router(config-telephony)# ip source-address 10.0.0.1 port 2000 secondary 10.5.2.78
```

Related Commands

Command	Description
telephony-service	Enters telephony-service configuration mode.



Cisco Unified CME Commands: K

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This chapter contains commands to configure and maintain Cisco Unified Communications Manager Express (formally known as Cisco Unified CallManager Express). The commands are presented in alphabetical order. Some commands required for configuring Cisco Unified Communications Manager Express (Cisco Unified CME) may be found in other Cisco IOS command references. Use the command reference master index or search online to find these commands.

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keepalive (ephone and ephone-template)

To set the length of the time interval between successive keepalive messages from the Cisco Unified CME router to a particular IP phone, use the **keepalive** command in ephone or ephone-template configuration mode. To reset this length to the default value, use the **no** form of this command.

keepalive *seconds*

no keepalive

Syntax Description

seconds	Interval time, in seconds. Range is from 10 to 65535. Default is 30.

Defaults

30 seconds

Command Modes

Ephone configuration
Ephone-template configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)T	Cisco CME 2.1	This command was introduced.
12.4(4)XC	Cisco Unified CME 4.0	This command was made available in ephone-template configuration mode.
12.4(9)T	Cisco Unified CME 4.0	This command in ephone-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

If the router fails to receive three successive keepalive messages, it considers the phone to be out of service until the phone reregisters.

This command allows the keepalive interval to be set for individual phones so that wireless phone batteries are not run down too quickly by overly frequent keepalive signals.

If the **keepalive (telephony-service)** command and the **keepalive (ephone)** command are set to different time intervals, the value that has been set using the **keepalive (ephone)** command is used for that phone only.

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

Examples

The following example sets the keepalive interval to 300 seconds:

```
Router(config)# ephone 1
Router(config-ephone)# keepalive 300
```

Related Commands

Command	Description
ephone	Enters ephone configuration mode.
keepalive (telephony-service)	Sets the time interval for keepalive messages between IP phones and the Cisco Unified CME router.

keepalive (telephony-service)

To set the length of the time interval between successive keepalive messages from the Cisco CallManager Express router to IP phones, use the **keepalive** command in telephony-service configuration mode. To reset this length to the default value, use the **no** form of this command.

keepalive *seconds*

no keepalive

Syntax Description

seconds	Interval time, in seconds. Range is from 10 to 65535. Default is 30.

Defaults

30 seconds

Command Modes

Telephony-service configuration

Command History

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

Usage Guidelines

If the router fails to receive three successive keepalive messages, it considers the phone to be out of service until the phone reregisters.

If the **keepalive (telephony-service)** command and the **keepalive (ephone)** command are set to different time intervals, the value that is set for the **keepalive (ephone)** command is used for that phone only.

Examples

The following example sets the keepalive time interval to 40 seconds:

```
Router(config)# telephony-service
Router(config-telephony)# keepalive 40
```

Related Commands

Command	Description
keepalive (ephone)	Sets the time interval for keepalive messages between a particular IP phone and the Cisco CME router.
telephony-service	Enters telephony-service configuration mode.

keepalive (voice register session-server)

To define the duration for registrations of external feature servers after which the registration expires, use the **keepalive** command in voice register session-server configuration mode. To return to default, use the **no** form of this command.

keepalive *seconds*

no keepalive

Syntax Description

seconds	Duration for registration, in seconds. Range: 60 t0 3600. Default: 300.

Command Default

300 seconds.

Command Modes

Voice register session-server configuration (config-register-fs)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XW1	Cisco Unified CME 4.2	This command was introduced.

Usage Guidelines

Use this command to define the duration for registration, in seconds, after which the registration expires unless the feature server reregisters before the registration expiry.

Examples

The following partial output shows the configuration for a session manager for an external feature server, including a keepalive expiry of 360 seconds:

```
router# show running-configuration
!
!
voice register session-server 1
 register-id CSR1
 keepalive 360
```

Related Commands

Command	Description
register id	Creates an ID for explicitly identifying an external feature server during Register requests

keep-conference

To allow conference initiators to exit from conference calls and to either end or maintain the conference for the remaining parties, use the **keep-conference** command in ephone or ephone-template configuration mode. To return to the default, use the **no** form of this command.

keep-conference [**drop-last**] [**endcall**] [**local-only**]

no keep-conference

Syntax Description

drop-last	(Optional) The action of the Confrn soft key is changed; the conference initiator can press the Confrn soft key (IP phone) or hookflash (analog phone) to drop the last party. **Note** Analog phones connected to the Cisco Unified CME system through a Cisco VG 224 require Cisco IOS Release 12.3(11)YL1 or a later release to use this feature.
endcall	(Optional) The action of the EndCall soft key is changed; the conference initiator can hang up or press the EndCall soft key to leave the conference and keep the other two parties connected. **Note** If this option is not enabled, pressing the EndCall soft key terminates the conference and disconnects all parties.
local-only	(Optional) The conference initiator can hang up to end the conference and leave the other two parties connected only if one of the remaining parties is local to the Cisco Unified CME system (an internal extension).

Defaults

The default is that the **keep-conference** command is not enabled. A conference initiator can hang up or press the EndCall soft key to end a conference and disconnect all parties or press the Confrn soft key to drop only the last party that was connected to the conference.

Command Modes

Ephone configuration
Ephone-template configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)T	Cisco CME 3.2	This command was introduced.
12.4(4)XC	Cisco Unified CME 4.0	The **drop-last** and **local-only** keywords were added, and this command was made available in ephone-template configuration mode.
12.4(9)T	Cisco Unified CME 4.0	The **drop-last** and **local-only** keywords, and this command in ephone-template configuration mode were integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

> **Note** This feature uses call transfer to connect the two remaining parties of a conference when a conference initiator leaves the conference. To use this feature, you must configure the **transfer-system** command using the **full-blind**, **full-consult**, or **full-consult dss** keywords.

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

If the **keep-conference** command is configured with no keywords, a conference initiator can hang up to leave the conference and the other two parties will remain connected. Alternatively, the conference initiator can use the EndCall soft key to terminate the conference and disconnect all parties.

If the **keep-conference** command is configured with no keywords, a conference initiator can use the Confrn soft key (IP phone) or hookflash (analog phone) to break up the conference but stay connected to both parties. The oldest call will be put on hold, and the most recent call will be actively connected to the initiator. The conference initiator can navigate between the two parties by pressing the Hold soft key or the appropriate line button on the phone.

If the **endcall** keyword is used, the conference initiator can hang up or press the EndCall soft key to leave the conference with the other two parties remaining connected.

In Cisco CME 3.2.3 and later versions, if the **keep-conference** command is not configured (the default) or if the **no keep-conference** command is used, a conference initiator can drop the last party that was added to the conference by pressing the Confrn soft key (IP phone) or hookflash (analog phone).

> **Note** Analog phones connected to the Cisco Unified CME system through a Cisco VG 224 require Cisco IOS Release 12.3(11)YL1 or a later release to use this feature.

Examples

In the following example, extension 3555 initiates a three-way conference. After the conference is established, extension 3555 can press the Confrn soft key to disconnect the last party that was connected and remain connected to the first party that was connected. If extension 3555 hangs up from the conference, the other two parties remain connected if one of them is local to the Cisco Unified CME system.

```
ephone-dn 35
 number 3555

ephone 24
 button 1:35
 keep-conference drop-last local-only
```

In the following example, extension 3666 initiates a three-way conference. After the conference is established, extension 3666 can press the Confrn soft key to disconnect the last party that was connected and remain connected to the first party that was connected. Also, extension 3666 can hang up from a three-way conference to terminate the conference and disconnect all parties or can press the EndCall soft key to leave the conference and keep the other two parties connected.

```
ephone-dn 36
 number 3666

ephone 25
 button 1:36
 keep-conference drop-last endcall
```

In the following example, extension 3777 initiates a three-way conference. After the conference is established, extension 3777 can press the Confrn soft key to disconnect the last party that was connected and remain connected to the first party that was connected. Also, extension 3777 can hang up from a three-way conference to terminate the conference and disconnect all parties or press the EndCall soft key to leave the conference and keep the other two parties connected only if one of the two parties is local to the Cisco Unified CME system.

```
ephone-dn 38
 number 3777

ephone 27
 button 1:38
 keep-conference drop-last endcall local-only
```

In the following example, extension 3999 initiates a three-way conference. After the conference is established, extension 3999 can hang up to terminate the conference and disconnect all parties or press the EndCall soft key to leave the conference and keep the other two parties connected only if one of the two parties is local to the Cisco Unified CME system.

```
ephone-dn 39
 number 3999

ephone 29
 button 1:39
 keep-conference endcall local-only
```

Related Commands

Command	Description
max-conferences	Sets the maximum number of three-party conferences simultaneously supported by the Cisco Unified CME router.
transfer-system	Specifies the call transfer method for IP phone extensions that use the ITU-T H.450.2 standard.

keep-conference (voice register pool)

To allow IP phone conference initiators to exit from conference calls and keep the remaining parties connected, use the **keep-conference** command in voice register pool configuration mode. To disable the keep-conference feature, use the **no** form of this command.

keep-conference

no keep-conference

Syntax Description

This command has no arguments or keywords.

Defaults

Enabled

Command Modes

Voice register pool configuration

Command History

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

Usage Guidelines

When the conference initiator hangs up, Cisco CallManager Express (Cisco CME) executes a call transfer to connect the two remaining lines. The remaining calls are transferred without consultation. To facilitate call transfer, the **transfer-attended** command or **transfer-blind** command must be enabled.

Conference initiators can disconnect from their conference calls by pressing the Confrn (conference) soft key. When an initiator uses the Confrn soft key to disconnect from the conference call, the oldest call leg is put on hold, leaving the initiator connected to the most recent call leg. The conference initiator can then navigate between the two separate parties by pressing either the Hold soft key or the line buttons to select the desired call.

Examples

The following example shows how to configure Cisco IP phones in Cisco CME to keep remaining conference legs after the conference initiator hangs up.

```
Router(config)# voice register pool 1
Router(config-register-pool)# keep-conference
```

Related Commands

Command	Description
conference (voice register template)	Enables a soft key for conference in a SIP phone template.
max-conferences	Sets the maximum number of three-party conferences simultaneously supported by the Cisco CME router.
template (voice register pool)	Applies a template to a SIP phone.

Command	Description
transfer-attended (voice register template)	Enables a soft key for attended transfer in a SIP phone template.
transfer-blind (voice register template)	Enables a soft key for blind transfer in a SIP phone template.
voice register template	Enters voice register template configuration mode and defines a template of common parameters for SIP phones.

keygen-retry

To specify the number of times that a CAPF server sends a key generation request, use the **keygen-retry** command in CAPF-server configuration mode. To return to the default, use the **no** form of this command.

keygen-retry *number*

no keygen-retry

Syntax Description

number	Number of retries. Range is from 0 to 100. Default is 3.

Command Default

Number of retries is 3.

Command Modes

CAPF-server configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used with Cisco Unified CME phone authentication.

Examples

The following example specifies that the key generation process should be tried 5 times.

```
Router(config)# capf-server
Router(config-capf-server)# source address 10.10.10.1
Router(config-capf-server)# trustpoint-label server25
Router(config-capf-server)# cert-oper upgrade all
Router(config-capf-server)# cert-enroll-trustpoint server12 password 0 x8oWiet
Router(config-capf-server)# auth-mode auth-string
Router(config-capf-server)# auth-string generate all
Router(config-capf-server)# port 3000
Router(config-capf-server)# keygen-retry 5
Router(config-capf-server)# keygen-timeout 45
Router(config-capf-server)# phone-key-size 2048
```

keygen-timeout

To specify the number of minutes that the CAPF server waits for a key-generation response from a phone, use the **keygen-timeout** command in CAPF-server configuration mode. To return to the default, use the **no** form of this command.

keygen-timeout *minutes*

no keygen-timeout

Syntax Description

minutes	Number of minutes before the generation process times out. Range is from 1 to 120. Default is 30.

Command Default

30 minutes

Command Modes

CAPF-server configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used with Cisco Unified CME phone authentication.

Examples

The following example specifies a period of 45 minutes before the key generation process times out.

```
Router(config)# capf-server
Router(config-capf-server)# source address 10.10.10.1
Router(config-capf-server)# trustpoint-label server25
Router(config-capf-server)# cert-oper upgrade all
Router(config-capf-server)# cert-enroll-trustpoint server12 password 0 x8oWiet
Router(config-capf-server)# auth-mode auth-string
Router(config-capf-server)# auth-string generate all
Router(config-capf-server)# port 3000
Router(config-capf-server)# keygen-retry 5
Router(config-capf-server)# keygen-timeout 45
Router(config-capf-server)# phone-key-size 2048
```

keypad-normalize

To impose a 200-millisecond delay before each keypad message from an IP phone, use the **keypad-normalize** command in ephone or ephone-template configuration mode. To return to the default, use the **no** form of this command.

keypad-normalize

no keypad-normalize

Syntax Description

This command has no keywords or arguments.

Command Default

Keypad messages are handled as fast as the system can handle them, without an imposed delay.

Command Modes

Ephone configuration
Ephone-template configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command normalizes the processing of incoming keypad messages from an IP phone so that one message is processed every 200 milliseconds. This is useful for handling the personal speed dial (fastdial) feature when the destination of the call tends to be slower in accepting the digits, or when converting keypad messages into appropriate digit events on the network side, such as RFC 2833 digits.

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

Examples

The following example normalizes the sending of digits from ephone 43.

```
ephone 43
 button 1:29
 keypad-normalize
```

keyphone

To designate a Cisco Unified IP phone as a marked or “key” phone when using the Cisco Unified CME eXtensible Markup Language (XML) application program interface (API), use the **keyphone** command in ephone or ephone-template configuration mode. To remove the keyphone designation, use the **no** form of this command.

keyphone

no keyphone

Syntax Description

This command has no arguments or keywords.

Defaults

The phone that is being configured is not a “key” phone.

Command Modes

Ephone configuration
Ephone-template configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.4(4)XC	Cisco Unified CME 4.0	This command was made available in ephone-template configuration mode.
12.4(9)T	Cisco Unified CME 4.0	This command in ephone-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used with the XML API to mark a Cisco Unified IP phone as a “key” phone to be tracked while using the XML API. The XML API can be instructed to report the status of only the “key” phones in the system for network management purposes, for example.

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

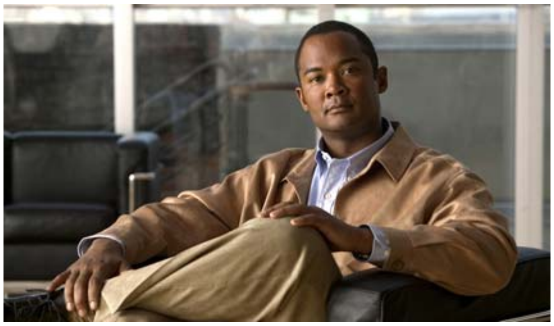
Examples

The following example sets the phone with the phone tag of 1 as a “key” phone for the XML API:

```
Router(config)# ephone 1
Router(config-ephone)# keyphone
```

Related Commands

Command	Description
ephone	Enters ephone configuration mode.



Cisco Unified CME Commands: L

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This chapter contains commands to configure and maintain Cisco Unified Communications Manager Express (formally known as Cisco Unified CallManager Express). The commands are presented in alphabetical order. Some commands required for configuring Cisco Unified Communications Manager Express (Cisco Unified CME) may be found in other Cisco IOS command references. Use the command reference master index or search online to find these commands.

label

To create a text identifier instead of a phone-number display for an extension on an IP phone console, use the **label command in** ephone-dn configuration mode. To delete a label, use the **no** form of this command.

label *string*

no label *string*

Syntax Description

string	Alphanumeric string of up to 30 characters.

Defaults

No label is defined.

Command Modes

Ephone-dn configuration

Command History

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

Usage Guidelines

One label is allowed per extension (ephone-dn). The ephone-dn must already have a number that was set using the **number** command before a label can be created for it.

This command must be followed by a quick reboot of the phone on which the label appears, using the **restart** command.

Examples

The following example creates three phone labels to appear in place of three phone numbers on IP phone console displays:

```
Router(config)# ephone-dn 10
Router(config-ephone-dn)# label user10
Router(config-ephone-dn)# exit

Router(config)# ephone-dn 20
Router(config-ephone-dn)# label user20
Router(config-ephone-dn)# exit

Router(config)# ephone-dn 30
Router(config-ephone-dn)# label user30
Router(config-ephone-dn)# exit
```

Related Commands

Command	Description
ephone-dn	Enters ephone-dn configuration mode.
number	Associates a telephone or extension number with an ephone-dn in a Cisco CME system.
restart (ephone)	Performs a fast reboot of a single phone associated with a Cisco CME router.
restart (telephony-service)	Performs a fast reboot of one or all phones associated with a Cisco CME router.

label (voice register dn)

To create a text identifier instead of a phone-number display for an extension on a SIP phone console, use the **label command in** voice register dn configuration mode. To delete a label, use the **no** form of this command.

label *string*

no label *string*

Syntax Description

string	Alphanumeric string of up to 30 characters.

Defaults

No default behavior or values

Command Modes

Voice register dn configuration

Command History

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

Usage Guidelines

One label is allowed per extension (directory number). The directory number must already have a number that was set by using the **number** command before a label can be created for it.

After you configure this command, restart the phone by using the **reset** command.

Examples

The following example shows how to create three phone labels to appear in place of three phone numbers on Cisco IP phone console displays:

```
Router(config)#  voice register dn 10
Router(config-register-dn)# label user10
Router(config-register-dn)# exit

Router(config)# voice register dn 20
Router(config-register-dn)# label user20
Router(config-register-dn)# exit

Router(config)# voice register dn 30
Router(config-register-dn)# label user30
Router(config-register-dn)# exit
```

Related Commands

Command	Description
number (voice register dn)	Associates a telephone or extension number with a directory number in a Cisco CME system.
reset (voice register pool)	Performs a compete reboot of a single SIP phone associated with a Cisco CME router.

Command	Description
reset (voice register global)	Performs a complete reboot of all SIP phones associated with a Cisco CME router.
voice register dn	Enters voice register dn configuration mode to define an extension for a SIP phone line.

list (ephone-hunt)

To create a list of extensions that are members of a Cisco Unified CME ephone hunt group, use the **list** command in ephone-hunt configuration mode. To remove a list from the router configuration, use the **no** form of this command.

list *number*[**,** *number...*]

no list

Syntax Description

number	Preconfigured extension or E.164 number. An asterisk (*) can take the place of an extension number to represent a wildcard slot. An agent at an authorized ephone-dn can dynamically join and leave a hunt group if a wildcard slot is available. There can be up to 20 wildcard slots in a hunt group.

Command Default

No list is defined.

Command Modes

Ephone-hunt configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.3(11)XL	Cisco CME 3.2.1	The number of ephone-dns allowed was increased to 20.
12.3(14)T	Cisco CME 3.3	This command was integrated into Cisco IOS Release 12.3(14)T.
12.4(4)XC	Cisco Unified CME 4.0	The use of wildcard asterisks (*) in the *dn-number* argument was introduced.
12.4(9)T	Cisco Unified CME 4.0	The use of wildcard asterisks (*) in the *dn-number* argument was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Use this command to create a list of member numbers for defining a hunt group.

List must contain 1 to 20 numbers.

A number cannot be added to a list unless it was already defined by using the **number** command.

Add or delete all numbers in a hunt group list at one time. You cannot add or single number to an existing list or remove one number from a list.

Any number in the list cannot be a pilot number of a parallel hunt group.

To allow dynamic membership in a hunt group, use asterisks to represent wildcard slots in the **list** command. To allow an ephone-dn to use one of the wildcard slots to dynamically join a hunt group, use the **ephone-hunt login** command under that ephone-dn. Ephone-dns are disallowed from joining hunt groups by default, so you have to explicitly allow this behavior for each ephone-dn that you want to be able to log into hunt groups.

The **show ephone-hunt** command displays the numbers associated to ephone-dns that are joined to groups at the time that the command is run, in addition to static members of the hunt group. Static hunt group members are the numbers that are explicitly named in the **list** command.

Examples

The following example creates sequential hunt group number 7, which contains four static members (ephone-dns):

```
Router(config)# ephone-hunt 7 sequential
Router(config-ephone-hunt)# list 7711, 7712, 7713, 7714
```

The following example creates five ephone-dns and a hunt group that includes the first two ephone-dns as static members and two wildcard slots for dynamic hunt group members. The last three ephone-dns are enabled for dynamic membership in the hunt group. Any of them can join the hunt group whenever one of the wildcard slots is available. Once an ephone-dn has joined a hunt group, it can leave at any time.

```
ephone-dn 22
 number 4566

ephone-dn 23
 number 4567

ephone-dn 24
 number 4568
 ephone-hunt login

ephone-dn 25
 number 4569
 ephone-hunt login

ephone-dn 26
 number 4570
 ephone-hunt login

ephone-hunt 1 peer
 list 4566,4567,*,*
 final 7777
```

Related Commands

Command	Description
ephone-hunt login	Allows an ephone-dn to dynamically join and leave an ephone hunt group.
final	Defines the last ephone-dn in an ephone hunt group.
hops	Defines the number of times that a call is redirected to the next ephone-dn in a peer ephone-hunt-group list before proceeding to the final ephone-dn.
max-redirect	Changes the current number of allowable redirects in a Cisco CME system.
no-reg (ephone-hunt)	Specifies that the pilot number of this ephone hunt group should not register with the H.323 gatekeeper.
number (ephone-dn)	Associates an extension or telephone number with a directory number.

Command	Description
pilot	Defines the ephone-dn that is dialed to reach an ephone hunt group.
preference (ephone-hunt)	Sets preference order for the ephone-dn associated with an ephone-hunt-group pilot number.
show ephone-hunt	Displays ephone-hunt group configuration, current status, and statistics.
timeout (ephone-hunt)	Sets the number of seconds after which a call that is not answered is redirected to the next number in the hunt-group list.

list (voice hunt-group)

To create a list of extensions that are members of a Cisco CallManager Express (Cisco CME) voice hunt group, use the **list** command in voice hunt-group configuration mode. To remove a list, use the **no** form of this command.

list *number*, *number*[, *number*...]

no list

Syntax Description

number	Preconfigured telephone or extension number assigned to supported SIP phones in Cisco CME.

Defaults

No default behavior or values

Command Modes

Voice hunt-group configuration

Command History

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

Usage Guidelines

Use this command to create a list of numbers for defining a hunt group.

List must contain 2 to 10 numbers.

A number cannot be added to a list unless it was already defined by using the **number** command.

Add or delete all numbers in a hunt group list at one time. You cannot add or single number to an existing list or remove one number from a list.

Any number in the list cannot be a pilot number of a parallel hunt group.

Examples

The following example shows how to create a sequential hunt group containing four directory numbers:

```
Router(config)# voice hunt-group 1 sequential
Router(config-voice-hunt-group)# list 7711, 7712, 7713, 7714
```

Related Commands

Command	Description
final (voice hunt-group)	Defines the last extension in a voice hunt group.
hops (voice hunt-group)	Defines the number of times that a call is redirected to the next directory number in a peer hunt-group list before proceeding to the final directory number.
number (voice register dn)	Associates an extension or telephone number with a directory number.

Command	Description
number (voice register pool)	Assigns a directory number to a SIP phone.
pilot (voice hunt-group)	Defines the voice dn that callers dial to reach a Cisco CallManager Express (Cisco CME) voice hunt group.
timeout (voice hunt-group)	Sets the number of seconds after which a call that is not answered is redirected to the next number in the hunt-group list and defines the last directory number in the hunt group.

load (telephony-service)

To associate a type of Cisco Unified IP phone with a phone firmware file, use the **load** command in telephony-service configuration mode. To disassociate a type of phone from a phone firmware file, use the **no** form of this command.

load *phone-type firmware-file*

no load *phone-type*

Syntax Description

phone-type	Type of Cisco Unified IP phone. The following choices are valid: • **7902**—Cisco Unified IP Phone 7902G. • **7905**—Cisco Unified IP Phone 7905G. • **7910—**Cisco Unified IP Phone 7910 and 7910G. • **7911**—Cisco Unified IP Phone 7911G. • **7912**—Cisco Unified IP Phone 7912G. • **7914—**Cisco Unified IP Phone 7914 Expansion Module. • **7920**—Cisco Unified Wireless IP Phone 7920. • **7921**—Cisco Unified Wireless IP Phone 7921G. • **7931**—Cisco Unified IP Phone 7931G. • **7935**—Cisco Unified IP Conference Station 7935. • **7936—**Cisco Unified IP Conference Station 7936. • **7941**—Cisco Unified IP Phone 7941G. • **7941GE**—Cisco Unified IP Phone 7941G-GE. • **7960-7940**—Cisco Unified IP Phones 7960 and 7960G and Cisco Unified IP Phone 7940 and 7940G. • **7961**—Cisco Unified IP Phone 7961G. • **7961GE—**Cisco Unified IP Phone 7961G-GE. • **7970—**Cisco Unified IP Phone 7970G. • **7971**—Cisco Unified IP Phone 7971G-GE. • **ATA—**Cisco ATA-186 and Cisco ATA-188.
firmware-file	Filename for the IP phone firmware to be associated with the IP phone type. For every phone type except analog telephone adapter (ATA), do not use any file extension. Filenames are case-sensitive.

Command Default

Firmware files are not associated with phone types.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco ITS 1.0	This command was introduced on the following platforms: Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(2)XT	Cisco ITS 2.0	This command was implemented on the Cisco 1750 and Cisco 1751.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	Cisco ITS 2.0	This command was implemented on the Cisco 2600-XM and Cisco 2691.
12.2(11)T	Cisco ITS 2.01	This command was implemented on the Cisco 1760.
12.2(11)YT	Cisco ITS 2.1	Support was added for the Cisco IP Phone 7914 Expansion Module.
12.2(15)T	Cisco ITS 2.1	This command was integrated into Cisco IOS Release 12.2(15)T.
12.2(15)ZJ	Cisco CME 3.0	The following keywords were added to this command: **7902**, **7905**, and **7912**.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.3(7)T	Cisco CME 3.1	The **7920** and **7936** keywords were added.
12.3(11)XL	Cisco CME 3.2.1	The **7970** keyword was added.
12.3(14)T	Cisco CME 3.3	The **7971** keyword was added, and this command was integrated into Cisco IOS Release 12.3(14)T.
12.4(4)XC	Cisco Unified CME 4.0	The **7911**, **7941**, **7941GE**, **7961**, and **7961GE** keywords were added.
12.4(9)T	Cisco Unified CME 4.0	The **7911**, **7941**, **7941GE**, **7961**, and **7961GE** keywords were integrated into Cisco IOS Release 12.4(9)T.
12.4(6)XE	Cisco Unified CME 4.0(2)	The **7931** keyword was added.
12.4(4)XC4	Cisco Unified CME 4.0(3)	The **7931** keyword was added.
12.4(11)T	Cisco Unified CME 4.0(3)	The **7931** keyword was integrated into Cisco IOS Release 12.4(11)T.
12.4(11)XJ2	Cisco Unified CME 4.1	The **7921** keywords was added.
12.4(15)T	Cisco Unified CME 4.1	The **7921** keyword was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

This command updates the Cisco Unified CME configuration file for the specified type of Cisco Unified IP phone to add the name of the correct firmware file that the phone should load. This filename also provides the version number for the phone firmware that is in the file. Later, whenever a phone is started up or rebooted, the phone reads the configuration file to determine the name of the firmware file that it should load and then looks for that firmware file on the TFTP server.

Cisco Unified IP phones update themselves with new phone firmware whenever they are started up or rebooted.

A separate **load** command is needed for each type of phone. The Cisco Unified IP Phones 7940 and 7940G and the Cisco Unified IP Phones 7960 and 7960G have the same phone firmware and share the **7960-7940** keyword.

When specifying the phone firmware filename in this command, for every phone type except **ATA**, do not use the .bin file extension. For example, if the firmware file for Cisco Unified IP Phone 7914 Expansion Modules is named W05473955.bin, enter the **load 7914 W05473955** command.

Following the **load** command, you use the **tftp-server** command to enable TFTP access to the file by Cisco Unified IP phones. Note that the **tftp-server** command does require that you use the file extension as part of the filename.

The **load** command must be followed by a reboot of the phones using the **reset** command.

Examples

The following example identifies the Cisco Unified IP phone firmware file that is used by the Cisco Unified IP Phones 7960 and 7960G and Cisco Unified IP Phone 7910G and then defines the Cisco CME router flash memory as the location of the phone firmware file. Note that no file extension is used with the **load** command, and the file extension is used with the **tftp-server** command.

```
Router(config)# telephony-service
Router(config-telephony)# load 7960-7940 P00303020209
Router(config-telephony)# load 7910 P00403020209
Router(config-telephony)# exit
Router(config)# tftp-server flash:P00303020209.bin
Router(config)# tftp-server flash:P00403020209.bin
```

Related Commands

Command	Description
reset	Resets a Cisco Unified IP phone.
tftp-server	Enables TFTP access to firmware files on the TFTP server.

load (voice register global)

To associate a type of IP phone with a phone firmware file, use the **load** command in voice register global configuration mode. To disassociate a type of phone from a phone firmware file, use the **no** form of this command.

load *phone-type firmware-file*

no load *phone-type*

Syntax Description

phone-type	Type of IP phone. The following choices are valid: • **3951**—Cisco Unified IP Phone 3951 • **7905**—Cisco Unified Phone 7905 and 7905G. • **7911**—Cisco Unified Phone 7911G. • **7912**—Cisco Unified Phone 7912 and 7912G. • **7941**—Cisco Unified Phone 7941G. • **7941GE**—Cisco Unified Phone 7941GE. • **7960–7940**—Cisco Unified Phone 7940 and 7940G and Cisco IP Phone 7960 and 7960G. • **7961**—Cisco Unified Phone 7961G. • **7961GE**—Cisco Unified Phone 7961GE. • **7970**—Cisco Unified Phone 7970G. • **7971**—Cisco Unified Phone 7971GE. • **ATA**—Cisco ATA-186 and Cisco ATA-188.
firmware-file	Filename for the Cisco Unified IP phone firmware to be associated with the IP phone type. Do not use the .bin or .load file extension, except for the Cisco Unified IP phone 7905, 7912, or ATA. Filenames are case sensitive.

Command Default

The firmware file is not associated with the type of phone.

Command Modes

Voice register global configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.
12.4(11)XJ	Cisco Unified CME 4.1	The **3951**, **7911**, **7941**, **7941GE**, **7961**, **7961GE**, **7970** and **7971** keywords were added.
12.4(15)T	Cisco Unified CME 4.1	The **3951**, **7911**, **7941**, **7941GE**, **7961**, **7961GE**, **7970** and **7971** keywords were added and this command was integrated into Cisco IOS Release 12.4(6th)T.

Usage Guidelines

This command updates the Cisco Unified CME configuration file for the specified type of IP phone to add the name of the correct firmware file that the phone should load. This filename also provides the version number for the phone firmware that is in the file. Later, whenever a phone is started up or rebooted, the phone reads the configuration file to determine the name of the firmware file that it should load and then looks for that firmware file on the TFTP server.

A separate **load** command is needed for each type of phone. The Cisco Unified IP Phones 7940 and 7940G and Cisco Unified IP Phones 7960 and 7960G have the same phone firmware and share the **7960-7940** keyword.

For Java-based IP phones, such as the Cisco Unified IP Phone 7906G, 7911G, 7941G, 7941GE, 7961G, 7961GE, 7970G, and 7971G, there are multiple firmware files. For these phones, use the TERMnn.x-y-x-w.loads or SCCPnn.x-y-x-w.loads firmware filename for the **load** command, without the.loads file extension. For these phones, you do *not* configure the **load** command for any firmware file other than the TERM.loads or SIP.loads firmware file.

Following the **load** command, use the **tftp-server** command to enable TFTP access to the file by Cisco Unified IP phones. The file extension is required when using the **tftp-server** command.

The **load** command must be followed by a reboot of the phones. Plug in a new IP phone or use the **reset** command to reboot an IP phone that is already connected to the Cisco router.

Examples

The following example shows how to configure the **load** command to indicate which phone firmware is to be used by a Cisco Unified IP Phone 7960 and 7960G, a Cisco Unified IP Phone 7912 and 7912G, and a Cisco Unified IP Phone 7941GEs. The **tftp-server** command is used to specify the location of the phone firmware files, including all firmware files for the Java-based Cisco Unified IP Phone 7941GE. Note that while no file extension is used with the **load** command, the file extension is required when using the **tftp-server** command.

```
Router(config)# voice register global
Router(config-voice-register)# load 7960-7940 P00303020209
Router(config-voice-register)# load 7912 P00403020209
Router(config-voice-register)# load 7941 TERM41.7-0-3-0S
Router(config-voice-register)# exit
Router(config)# tftp-server flash:P00303020209.bin
Router(config)# tftp-server flash:P00403020209.bin
Router(config)# tftp-server flash:SIP41.8-0-3-0S.loads
Router(config)# tftp-server flash:term61.default.loadsterm
Router(config)# tftp-server flash:41.default.loads
Router(config)# tftp-server flash:CVM41.2-0-2-26.sbn
Router(config)# tftp-server flash:cnu41.2-7-6-26.sbn
Router(config)# tftp-server flash:Jar41.2-9-2-26.sbn
```

Related Commands

Command	Description
reset (voice register global)	Performs a complete reboot of all SIP phones associated with a Cisco Unified CME router.
show voice register global	Displays all global configuration parameters associated with SIP phones.
tftp-server	Enables TFTP access to firmware files on the TFTP server.

Command	Description
type (voice register pool)	Defines a phone type for a SIP phone.
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.

load-cfg-file

To load configuration files on the TFTP server and to sign configuration files that are not created by Cisco Unified CME, use the **load-cfg-file** command in telephony-service configuration mode. To return to the default, use the **no** form of this command.

load-cfg-file *file-url* **alias** *file-alias* [**sign**] [**create**]

no load-cfg-file *file-url* **alias** *file-alias*

Syntax Description

file-url	Complete path of a configuration file in a local directory.
alias *file-alias*	Name of the file on the TFTP server.
sign	Signs the file and serves it on the TFTP server.
create	Creates the signed file in the local directory.

Command Default

A file is not loaded on the TFTP server.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used with Cisco Unified CME phone authentication to sign configuration files that are not created by Cisco Unified CME. This command also loads the signed and unsigned versions of the file on the TFTP server. To simply serve an already signed file on the TFTP server, use this command without the **sign** and **create** keywords.

The **create** keyword should be used with the **sign** keyword the first time that this command is used for each file. The **create** keyword is not maintained in the running configuration; this prevents signed files from being recreated during every reload.

Examples

The following example creates a file called ringlist.xml.sgn in slot0 and serves both ringlist.xml and ringlist.xml.sgn on the TFTP server.

```
telephony-service
 load-cfg-file slot0:Ringlist.xml alias Ringlist.xml sign create
```

The following example serves P00307010200.sbn on the TFTP server without creating a signed file.

```
telephony-service
 load-cfg-file slot0:P00307010200.sbn alias P00307010200.sbn
```

log password

Effective with Cisco Unified CME 4.0, the **log password** command was replaced by the **xml user** command in telephony-service configuration mode. See the **xml user** command for more information.

For Cisco CME 3.4 and earlier versions, to set a local password for an eXtensible Markup Language (XML) Application Programming Interface (API) query, use the **log password** command in telephony-service configuration mode. To remove the password definition, use the **no** form of this command.

log password *password-string*

no log password *password-string*

Syntax Description

password-string	Character string that is a password for XML API queries. Maximum length is 28 characters. Longer strings are truncated.

Command Default

No password is defined.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.4(4)XC	Cisco Unified CME 4.0	This command was replaced by the **xml user** command.
12.4(9)T	Cisco Unified CME 4.0	This command was replaced by the **xml user** command.

Usage Guidelines

The local password is used to authenticate XML API requests on the network management server. If the password is not set, an XML API query fails local authentication.

The password string is stored as plain text. No encryption is supported.

Examples

The following example defines a local password for XML API requests:

```
Router(config)# telephony-service
Router(config-telephony)# log password ewvpil
```

Related Commands

Command	Description
telephony-service	Enters telephony-service configuration mode.

log table

To set parameters for the table used to capture phone events used for the eXtensible Markup Language (XML) Application Programming Interface (API), use the **log table** command in telephony-service configuration mode. To reset parameters to their default values, use the **no** form of this command.

log table {**max-size** *entries* | **retain-timer** *minutes*}

no log table {**max-size** | **retain-timer**}

Syntax Description

max-size *entries*	Number of entries in the log table. Range is from 0 to 1000. Default is 150.
retain-timer *minutes*	Number of minutes to retain entries in the log table. Range is from 2 to 500. Default is 15.

Defaults

max-size: 150
retain-timer: 15

Command Modes

Telephony-service configuration

Command History

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

Usage Guidelines

Cisco CallManager Express (Cisco CME) captures and time-stamps events, such as phones registering and unregistering and extension status, and stores them in an internal buffer. This command sets the maximum number of events, or entries, that can be stored in the table. One event equals one entry. The **retain-timer** keyword sets the number of minutes that events are kept in the buffer before they are deleted.

The event table can be viewed using the **show fb-its-log** command.

Examples

The following example sets the maximum size of the table at 750 events and sets the retention time at 30 minutes:

```
Router(config)# telephony-service
Router(config-telephony)# log table max-size 750
Router(config-telephony)# log table retain-timer 30
```

Related Commands

Command	Description
show fb-its-log	Displays information about the Cisco CME XML API configuration, statistics on XML API queries, and event logs.
telephony-service	Enters telephony-service configuration mode.

login (telephony-service)

To define when users of IP phones in a Cisco CallManager Express (Cisco CME) system are logged out automatically, use the **login** command in telephony-service configuration mode. To revert to the default length of time before automatic logout, use the **no** form of this command.

login [**timeout** [*minutes]*] [**clear** *time*]

no login

Syntax Description

timeout	(Optional) Deactivates user login after a phone is idle for a given number of minutes.
minutes	(Optional) Number of minutes for which an IP phone can be idle before it is logged out automatically. Range is from 5 to 1440. Default is 60.
clear *time*	(Optional) Deactivates user login for all IP phones at the specified time of day, using 00:00 to 24:00 on a 24-hour clock. For example, 10:30 p.m. is 22:30. Default is 24:00 (midnight).

Defaults

minutes: 60
clear *time*: 24:00 (midnight)

Command Modes

Telephony-service configuration

Command History

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

Usage Guidelines

The **login** command is used in conjunction with the **pin** command to define the capability for individual phone users to override call blocking. Call blocking on IP phones is defined in the following way. First, one or more patterns of outgoing digits are defined using the **after-hours block pattern** command. Next, one or more time periods during which calls that match those patterns are to be blocked are defined using the **after-hours date** or **after-hours day** command or both. By default, all IP phones in a Cisco CME system are restricted if at least one pattern and at least one time period are defined. Individual phones can be exempted from call blocking using the **after-hour exempt** command. Individual users on specified phones can override call blocking by logging in with their PINs. Those logins are terminated at a specific time or after a specified period of idleness, as directed by the parameters in this command.

The **login** command applies only to IP phones that have soft keys, such as the Cisco IP Phones 7940 and 7940G and the Cisco IP Phones 7960 and 7960G.

This command must be followed by a quick reboot of the phones using the **restart all** command.

When a Cisco CME router is rebooted, the login status for all phones is reset to the default.

Examples

The following example sets the login deactivation to occur after a 2-hour idle time and after 11:30 p.m.

```
Router(config)# telephony-service
Router(config-telephony)# login timeout 120 clear 2330
```

Related Commands

Command	Description
after-hour exempt	Specifies that an IP phone does not have any of its outgoing calls blocked even though call blocking has been defined.
after-hours block pattern	Defines a pattern of digits for blocking outgoing calls from IP phones.
after-hours date	Defines a recurring period based on date during which outgoing calls that match defined block patterns are blocked on IP phones.
after-hours day	Defines a recurring period based on day of the week during which outgoing calls that match defined block patterns are blocked on IP phones.
pin	Sets a global/individual PIN for phone users to deactivate call blocking during nonwork hours.
restart (ephone)	Performs a fast reboot of a single phone associated with a Cisco CME router.
restart (telephony-service)	Performs a fast reboot of one or all phones associated with a Cisco CME router.
show ephone login	Displays the login states of all phones.
telephony-service	Enters telephony-service configuration mode.

logo (voice register global)

To specify a file to display on SIP phones, use the **logo** command in voice register global configuration mode. To disable the display of the file, use the **no** form of this command.

logo *url*

no logo

Syntax Description

url	URL as defined in RFC 2396.

Defaults

No file is specified for display on idle phones.

Command Modes

Voice register global configuration

Command History

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

Usage Guidelines

Use this command to define the URL for the file to be used by SIP phones connected in Cisco Unified CME. The file that is displayed must be encoded in eXtensible Markup Language (XML) by using the Cisco XML document type definition (DTD). For more information about Cisco DTD formats, see the *Cisco IP Phone Services Application Development Notes*.

After you configure this command, restart the phones by using the **reset** command.

Examples

The following example shows how to specify that the file logo.xml should be displayed on SIP phones:

```
Router(config)# voice register global
Router(config-register-global)# logo http://mycompany.com/files/logo.xml
```

Related Commands

Command	Description
reset (voice register pool)	Performs a complete reboot of one phone associated with a Cisco CME router.
reset (voice register global)	Performs a complete reboot of one or all phones associated with a Cisco CME router.
voice register global	Enters voice register global configuration mode in order to set global parameters for all supported Cisco SIP phones in a Cisco CME or Cisco SIP SRST environment.

logout-profile

To enable an IP phone for extension mobility and apply a default logout profile to the phone, use the **logout-profile** command in ephone configuration mode. To disable extension mobility, use the **no** form of this command.

logout-profile *profile-tag*

no logout-profile *profile-tag*

Syntax Description

profile-tag	Unique identifier for a default logout profile to be applied. Previously created by using the **voice logout-profile** command in voice logout-profile configuration mode. Range: 1 to maximum number of phones supported by platform.

Command Default

IP phone is not enabled for extension mobility.

Command Modes

Ephone configuration (config-ephone)

Command History

Cisco IOS Release	Cisco Product	Modification
12.5(11)XW1	Cisco Unified CME 4.2	This command was introduced.

Usage Guidelines

Use this command in ephone configuration mode to enable a supported IP phone registered in Cisco Unified CME for extension mobility and to apply a default logout profile to the ephone being configured.

In Cisco Unified CME 4.2, extension mobility is supported only on SCCP IP phones.

Extension mobility is not supported on non-display IP phones.

Extension mobility is not supported for analog devices.

Before using this command, you must create a logout profile to be applied to this phone by using the **voice logout-profile** command.

You cannot apply more than one logout profile to an ephone. If you attempt to apply a second logout profile to an ephone to which a profile has already been applied, the second profile will overwrite the first logout profile configuration.

Examples

The following example shows the ephone configuration for three different Cisco Unified IP phones. All three phones are enabled for extension mobility and share the same logout profile number 1, to be downloaded when these phones boot and when no phone user is logged into these phone:

```
ephone 1
 mac-address 000D.EDAB.3566
 type 7960
 logout-profile 1
```

```
ephone 2
 mac-address 0012.DA8A.C43D
 type 7970
 logout-profile 1

ephone 3
 mac-address 1200.80FC.9B01
 type 7911
 logout-profile 1
```

Related Commands

Command	Description
voice logout-profile	Enters voice profile configuration mode for the purpose of configuring a default logout profile for extension mobility.

loopback-dn

To create a virtual loopback voice port (loopback-dn) to establish a demarcation point for VoIP calls and supplementary services, use the **loopback-dn** command in ephone-dn configuration mode. To delete a loopback-dn configuration, use the **no** form of this command.

loopback-dn *dn-tag* [**forward** *number-of-digits* | **strip** *number-of-digits*] [**prefix** *prefix-digit-string*] [**suffix** *suffix-digit-string*] [**retry** *seconds*] [**auto-con**] [**codec** {**g711alaw** | **g711ulaw**}]

no loopback-dn

Syntax Description

dn-tag	Unique sequence number that identifies the ephone-dn that is being paired for loopback with the ephone-dn that is currently being configured. The paired ephone-dn must be one that is already defined in the system.
forward *number-of-digits*	(Optional) Number of digits in the original called number to forward to the other ephone-dn in the loopback-dn pair. Range is from 1 to 32 digits. Default is to forward all digits.
strip *number-of-digits*	(Optional) Number of leading digits to be stripped from the original called number before forwarding to the other ephone-dn in the loopback-dn pair. Range is from 1 to 32 digits. Default is not to A-law strip any digits.
prefix *prefix-digit-string*	(Optional) Defines a string of digits to add in front of the forwarded called number. Maximum number of digits in the string is 32. Default is that no prefix is defined.
suffix *suffix-digit-string*	(Optional) Defines a string of digits to add to the end of the forwarded called number. Maximum number of digits in the string is 32. Default is that no suffix is defined. If you add a suffix that starts with the pound character (#), the string must be enclosed in quotation marks.
retry *seconds*	(Optional) Number of seconds to wait before retrying the loopback target when it is busy or unavailable. Range is from 0 to 32767. Default is that retry is disabled and appropriate call-progress tones are passed to the call originator.
auto-con	(Optional) Immediately connects the call and provides in-band alerting while waiting for the far-end destination to answer. Default is that automatic connection is disabled.
codec	(Optional) Explicitly forces the G.711 A-law or G.711 mu-law voice coding type to be used for calls that pass through the loopback-dn. This overrides the G.711 coding type that is negotiated for the call and provides mu-law to A-law conversion if needed. Default is that Real-Time Transport Protocol (RTP) voice packets are passed through the loopback-dn without considering the G.711 coding type negotiated for the calls.
g711alaw	G.711 A-law, 64000 bits per second, for T1.
g711ulaw	G.711 mu-law, 64000 bits per second, for E1.

Defaults

All calls are set to forward all digits and not to strip any digits.
Prefix is not defined.
Suffix is not defined.
Retry is disabled.

Automatic connection is disabled.

RTP voice packets are passed through the loopback-dn without considering the G.711 coding type negotiated for the call.

Command Modes

Ephone-dn configuration

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.2(2)XT	2.0	This command was introduced on the following platforms: Cisco 1750, Cisco 1751, Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(2)XT3	2.0	The **suffix** keyword was added.
12.2(8)T	2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745 routers. The **auto-con** keyword was added.
12.2(8)T1	2.0	This command was implemented on the Cisco 2600-XM and Cisco 2691 routers.
12.2(11)T	2.01	This command was integrated into Cisco IOS Release 12.2(11)T and the **suffix** keyword was added.
12.2(11)YT	2.1	This command was integrated into Cisco IOS Release 12.2(11)YT and the **strip** keyword was added.
12.2(11)YT2	2.1	The **codec** keyword was added.
12.2(15)T	2.1	This command was integrated into Cisco IOS Release 12.2(15)T.

Usage Guidelines

The **loopback-dn** command is used to configure two ephone-dn virtual voice ports as back-to-back-connected voice-port pairs. A call presented on one side of the loopback-dn pair is reoriginated as a new call on the opposite side of the loopback-dn pair. The **forward**, **strip, prefix**, and **suffix** keywords can be used to manipulate the original called number that is presented to the incoming side of the loopback-dn pair to generate a modified called number to use when reoriginating the call at the opposite side of the loopback-dn pair. For loopback-dn configurations, you must always configure ephone-dn virtual voice ports as cross-coupled pairs.

Note Use of loopback-dn configurations within a VoIP network should be restricted to resolving critical network interoperability service problems that cannot otherwise be solved. Loopback-dn configurations are intended to be used in VoIP network interworking situations in which the only other alternative would be to make use of back-to-back-connected physical voice ports. Loopback-dn configurations emulate the effect of a back-to-back physical voice-port arrangement without the expense of the physical voice-port hardware. A disadvantage of loopback-dn configurations is that, because digital signal processors (DSPs) are not involved in a loopback-dn arrangement, the configuration does not support interworking or transcoding between calls that use different voice codecs. In many cases, the use of back-to-back physical voice ports that do use DSPs to resolve VoIP network interworking issues is preferred, because it introduces fewer restrictions in terms of supported codecs and call flows. Also, loopback-dns do not support T.38 fax relay.

Note We recommend that you create the basic ephone-dn configuration for both ephone-dn entries before configuring the loopback-dn option under each ephone-dn. The loopback-dn mechanism should be used only in situations where the voice call parameters for the calls on either side of the loopback-dn use compatible configurations; for example, compatible voice codec and dual tone multifrequency (DTMF) relay parameters. Loopback-dn configurations should be used only for G.711 voice calls.

The loopback-dn arrangement allows an incoming telephone call to be terminated on one side of the loopback-dn port pair and a new pass-through outgoing call to be originated on the other side of the loopback-dn port pair. The loopback-dn port pair normally works with direct cross-coupling of their call states; the alerting call state on the outbound call segment is associated with the ringing state on the inbound call segment.

The loopback-dn mechanism allows for call operations (such as call transfer and call forward) that are invoked for the call segment on one side of the loopback-dn port pair to be isolated from the call segment that is present on the opposite side of the loopback-dn port pair. This approach is useful when the endpoint devices associated with the two different sides have mismatched call-transfer and call-forwarding capabilities. The loopback-dn arrangement allows for call-transfer and call-forward requests to be serviced on one side of the loopback-dn port pair by creating hairpin-routed calls when necessary. The loopback-dn arrangement avoids the propagation of call-transfer and call-forward requests to endpoint devices that do not support these functions.

The **loopback-dn** command provides options for controlling the called-number digits that are passed through from the incoming side to the outgoing side. The available digits can be manipulated with the **forward**, **strip**, **prefix**, and **suffix** keywords.

The **forward** keyword defines the number of digits in the original called number to forward to the other ephone-dn in the loopback-dn pair. The default is set to forward all digits. The **strip** keyword defines the number of leading digits to be stripped from the original called number before forwarding to the other ephone-dn in the loopback-dn pair. The default is set to not strip any digits. The **forward** and **strip** commands are mutually exclusive and can be used with any combination of the **prefix** and **suffix** keywords.

The **prefix** keyword defines a string of digits to add in front of the forwarded number.

The **suffix** keyword is most commonly used to add a terminating "#" (pound-sign) character to the end of the forwarded number to indicate that no more digits should be expected. The pound-sign character indicates to the call-routing mechanism that is processing the forwarded number that the forwarded number is complete. Providing an explicit end-of-number character also avoids a situation in which the call-processing mechanism waits for the interdigit timeout period to expire before routing the call onward using the forwarded number.

Note The Cisco IOS command-line interface (CLI) requires that arguments with character strings that start with the pound-sign (#) character be enclosed within quotation marks; for example, "#".

The **retry** keyword is used to suppress a far-end busy indication on the outbound call segment. Instead of returning a busy signal to the call originator (on the incoming call segment), a loopback-dn presents an alerting or ringing tone to the caller and then periodically retries the call to the final far-end destination (on the outgoing call segment). This is not bidirectional. To prevent calls from being routed into the idle outgoing side of the loopback-dn port pair during the idle interval that occurs between successive outgoing call attempts, configure the outgoing side of the loopback-dn without a number so that there is no number to match for the inbound call.

The **auto-con** keyword is used to configure a premature trigger for a connected state for an incoming call segment while the outgoing call segment is still in the alerting state. This setup forces the voice path to open for the incoming call segment and support the generation of in-band call progress tones for busy, alerting, or ringback. The disadvantage of the **auto-con** keyword is premature opening of the voice path during the alerting stage and also triggering of the beginning of billing for the call before the call has been answered by the far end. These disadvantages should be considered carefully before you use the **auto-con** keyword.

The **codec** keyword is used to explicitly select the A-law or mu-law type of G.711 and to provide A-law to mu-law conversion if needed. Setting the codec type on one side of the loopback-dn forces the selection of A-law or mu-law for voice packets that are transmitted from that side of the loopback-dn. To force the A-law or mu-law G.711 codec type for both voice packet directions, set the codec type on both sides of the loopback-dn. Loopback-dn configurations are used only with G.711 calls. Other voice codec types are not supported.

Examples

The following example creates a loopback-dn configured with the **forward** and **prefix** keywords:

```
Router(config)# ephone-dn 7
Router(config-ephone-dn)# loopback-dn 15 forward 5 prefix 41
```

The following example creates a loopback-dn that appends the pound-sign (#) character to forwarded numbers to indicate the end of the numbers:

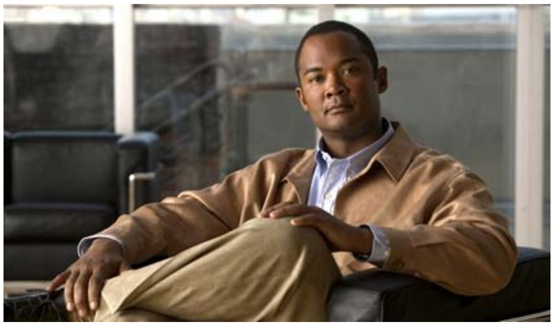
```
Router(config)# ephone-dn 7
Router(config-ephone-dn)# loopback-dn 16 suffix "#"
```

The following example shows a loopback-dn configuration that pairs ephone-dns 15 and 16. An incoming call (for example, from VoIP) to 4085550101 matches ephone-dn 16. The call is then reoriginated from ephone-dn 15 and sent to extension 50101. Another incoming call (for example, from a local IP phone) to extension 50151 matches ephone-dn 15. It is reoriginated from ephone-dn 16 and sent to 4085550151.

```
ephone-dn 15
 number 5015.
 loopback-dn 16 forward 5 prefix 40855
 caller-id local
 no huntstop
!
ephone-dn 16
 number 408555010.
 loopback-dn 15 forward 5
 caller-id local
 no huntstop
```

Related Commands

Command	Description
ephone-dn	Enters ephone-dn configuration mode.
show ephone-dn loopback	Displays information about loopback ephone-dns that have been created in a Cisco CME system.



Cisco Unified CME Commands: M

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This chapter contains commands to configure and maintain Cisco Unified Communications Manager Express (formally known as Cisco Unified CallManager Express). The commands are presented in alphabetical order. Some commands required for configuring Cisco Unified Communications Manager Express (Cisco Unified CME) may be found in other Cisco IOS command references. Use the command reference master index or search online to find these commands.

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mac-address (ephone)

To associate the MAC address of a Cisco IP phone with an ephone configuration in a Cisco CallManager Express (Cisco CME) system, use the **mac-address** command in ephone configuration mode. To disassociate the MAC address from an ephone configuration, use the **no** form of this command.

mac-address [*mac-address*]

no mac-address

Syntax Description

mac-address	Identifying MAC address of an IP phone, which is found on a sticker located on the bottom of the phone.

Defaults

No default behavior or values

Command Modes

Ephone configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco ITS 1.0	This command was introduced on the following platforms: Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(2)XT	Cisco ITS 2.0	This command was implemented on the Cisco 1750 and Cisco 1751.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	Cisco ITS 2.0	This command was implemented on the Cisco 2600-XM and Cisco 2691.
12.2(11)T	Cisco ITS 2.01	This command was implemented on the Cisco 1760.
12.2(15)ZJ	Cisco CME 3.0	The *mac-address* argument was made optional to enable automatic MAC address assignment after registration of phones.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Usage Guidelines

Use this command to specify the MAC address of a specific Cisco IP phone in order to physically identify the Cisco IP phone in a Cisco CME configuration. The MAC address of each Cisco IP phone is printed on a sticker that is placed on the bottom of the phone.

If you choose to register phones before configuring them, the **mac-address** command can be used during configuration without entering the *mac-address* argument. The Cisco CME system detects MAC addresses and automatically populates phone configurations with their corresponding MAC addresses and phone types. This capability is not supported for voice-mail ports and is supported only by Cisco CME 3.0 and later versions. To use this capability, enable Cisco CME by using the following commands: **max-ephones**, **max-dn**, **create cnf-files**, and **ip source-address.** After these commands have been used, phones can start to register. Then, when you are configuring a registered ephone and you use the **mac-address** command with no argument, the MAC address of the phone is automatically read into the configuration. The equivalent functionality is available through the Cisco CME graphic user interface (GUI).

If you choose to configure phones before registering them, the MAC address for each ephone must be entered during configuration.

Examples

The following example associates the MAC address CFBA.321B.96FA with the IP phone that has phone-tag 22:

```
Router(config)# ephone 22
Router(config-ephone)# mac-address CFBA.321B.96FA
```

Related Commands

Command	Description
create cnf-files	Builds the XML configuration files that are required for IP phones used with Cisco IOS Telephony Services V2.1, Cisco CallManager Express 3.0, or later versions.
ephone	Enters ephone configuration mode.
ip source-address	Identifies the IP address and port through which IP phones communicate with a Cisco CME router.
max-dn	Sets the maximum number of ephone-dns to be supported by a Cisco CME router.
max-ephones	Sets the maximum number of ephones to be supported by a Cisco CME router.
show ephone registered	Displays status and information for registered IP phones.

mailbox-selection (dial-peer)

To set a policy for selecting a mailbox for calls from a Cisco Unified CME system that are diverted before being sent to a Cisco Unity Express or PBX voice-mail pilot number, use the **mailbox-selection** command in dial-peer configuration mode. To return to the default, use the **no** form of this command.

mailbox-selection {**last-redirect-num** | **orig-called-num**}

no mailbox-selection

Syntax Description

last-redirect-num	(PBX voice mail only) The mailbox to which the call will be sent is the number that diverted the call to the voice-mail pilot number (the last number to divert the call).
orig-called-num	(Cisco Unity Express only) The mailbox to which the call will be sent is the number that was originally dialed before the call was diverted.

Command Default

Cisco Unity Express uses the last number to which the call was diverted before it was sent to voice mail as the mailbox number. Some legacy PBX systems use the originally called number as the mailbox number.

Command Modes

Dial-peer configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4()T.

Usage Guidelines

When Cisco Unified CME diverts a call, it captures the reroute information which will be used to compose a reroute request. A dial-peer match will be performed against the diverted-to number. If this is the voice mail pilot number and the **mailbox-selection** command has been used to install a policy, the reroute information will be amended as directed by the command. The originator will pick up the modified reroute request, build the diversion information and include it in the new diverted call to the voice-mail pilot number.

This command should be used on the outbound dial peer for the pilot number of the voice-mail system.

This command might not work properly in certain network topologies, including the following cases:

- When the last redirecting endpoint is not hosted on Cisco Unified CME. This may rarely occur with a PBX.
- When a call is forwarded across several SIP trunks. Multiple SIP Diversion Headers (stacking hierarchy) are not supported in Cisco IOS software.
- When a call is forwarded across non Cisco voice gateways that do not support the optional H450.3 originalCalledNr field.

Examples

The following example shows how to set a policy to select the mailbox of the originally called number when a call is diverted to a Cisco Unity Express or PBX voice-mail system with the pilot number 7000.

```
dial-peer voice 7000 voip
 destination-pattern 7000
 session target ipv4:10.3.34.211
 codec g711ulaw
 no vad
 mailbox-selection orig-called-num
```

mailbox-selection (ephone-dn)

To set a policy for selecting a mailbox for calls that are diverted before being sent to a Cisco Unity voice-mail pilot number, use the **mailbox-selection** command in ephone-dn configuration mode. To return to the default, use the **no** form of this command.

mailbox-selection {l**ast-redirect-num**}

no mailbox-selection

Syntax Description

last-redirect-num	The mailbox to which the call will be sent is the last number to divert the call.

Command Default

Cisco Unity uses the originally called number as the mailbox number.

Command Modes

Ephone-dn configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used on the ephone-dn associated with the voice-mail pilot number.

This command can only be used with SCCP phones.

This command might not work properly in certain network topologies, including the following cases:

- When the last redirecting endpoint is not hosted on Cisco Unified CME. This may rarely occur with a PBX.
- When a call is forwarded across several SIP trunks. Multiple SIP Diversion Headers (stacking hierarchy) are not supported in Cisco IOS software.
- When a call is forwarded across non Cisco voice gateways that do not support the optional H450.3 originalCalledNr field.

Examples

The following example sets a policy to select the mailbox of the last redirecting number when a call is diverted to a Cisco Unity voice-mail system with the pilot number 8000.

```
ephone-dn 2583
 number 8000
 mailbox-selection last-redirect-num
```

max-conferences

To set the maximum number of three-party conferences that are supported simultaneously by the Cisco CallManager Express (Cisco CME) router, use the **max-conferences** command in telephony-service configuration mode. To reset this number to the default, use the **no** form of this command.

max-conferences *max-conference-number* [**gain -6** | **0** | **3** | **6**]

no max-conferences

Syntax Description

max-conference number	Maximum number of three-party conferences that are supported simultaneously by the router. This number is platform-dependent, and the default is half the maximum for each platform. The following are the maximum values for this argument: • Cisco 1700 series, Cisco 2600 series, Cisco 2801—8 • Cisco 2811, Cisco 2821, Cisco 2851, Cisco 3600 series, Cisco 3700 series—16 • Cisco 3800 series—24 (requires Cisco IOS Release 12.3(11)XL or higher) **Note** Each individual Cisco IP phone can host a maximum of one conference at a time. You cannot create a second conference on the phone if you already have an existing conference on hold.
gain	(Optional) Increases the sound volume of VoIP and public switched telephony network (PSTN) parties joining a conference call. The allowable decibel units are -6 db, 0 db, 3 db, and 6 db. The default is -6 db.

Defaults

Half the maximum number of simultaneous three-party conferences for each platform

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(2)XT	Cisco ITS 2.0	This command was introduced on the following platforms: Cisco 1750, Cisco 1751, Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	Cisco ITS 2.0	This command was implemented on the Cisco 2600-XM and Cisco 2691.
12.2(11)T	Cisco ITS 2.01	This command was implemented on the Cisco 1760.

Cisco IOS Release	Cisco Product	Modification
12.3(11)XL1	Cisco CME 3.2.1	The **gain** keyword was added.
12.3(14)T	Cisco CME 3.3	This command was integrated into Cisco IOS Release 12.3(14)T.

Usage Guidelines

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

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

Examples

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

```
Router(config)# telephony-service
Router(config-telephony)# max-conferences 4 gain 6
```

Related Commands

Command	Description
telephony-service	Enters telephony-service configuration mode.

max-dn

To set the maximum number of extensions (ephone-dns) to be supported by a Cisco Unified CME router, use the **max-dn** command in telephony-service configuration mode. To reset this number to the default value, use the **no** form of this command.

max-dn *max-directory-numbers* [**preference** *preference-order*] [**no-reg** {**primary** | **both**}]

no max-dn

Syntax Description

max-directory-numbers	Maximum number of extensions (ephone-dns) to allow in the Cisco CME system. The maximum you can set depends on the software version, router platform, and amount of memory that you have installed. Type **?** to display range. The default is 0.
preference *preference-order*	(Optional) Sets a preference value for the primary number of an ephone-dn. Refer to CLI help for a range of numeric options, where 0 is the highest preference. Default is 0.
no-reg	(Optional) Globally disables ephone registration with an H.323 gatekeeper or SIP proxy.
primary	Primary ephone-dn numbers only.
both	Both primary and secondary ephone-dn numbers.

Command Default

The maximum number of extensions is 0.

Command Modes

Telephony-service configuration

Command History

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

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified 4.0	The **preference**, **no-reg**, **primary**, and **both** keywords were introduced.
12.4(9)T	Cisco Unified 4.0	The **preference**, **no-reg**, **primary**, and **both** keywords were integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

The **max-dn** command limits the number of extensions (ephone-dns) available in a Cisco Unified CME system. The maximum number of ephone-dns that you can create depends on the software version, router platform, and amount of memory that you have installed. For the maximum number of ephone-dns and recommended memory for each platform, see the *Cisco CallManager Express Supported Firmware, Platforms, Memory, and Voice Products* for your Cisco Unified CME version.

The **max-ephones** command similarly limits the number of IP phones in a Cisco Unified CME system.

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

If registration with an H.323 gatekeeper or SIP proxy is enabled globally (the default), you can override the setting per extension by using the **no-reg** keyword in the **number** command for individual ephone-dns.

After using this command, you can provision individual extensions using the Cisco Unified CME graphic user interface (GUI) or the router CLI in ephone-dn configuration mode.

Examples

The following example sets the maximum number of extensions (ephone-dns) to 12:

```
Router(config)# telephony-service
Router(config-telephony)# max-dn 12
```

The following example sets the maximum number of extensions to 150 and specifies that the primary number of each extension should receive a dial-peer preference order of 1:

```
Router(config)# telephony-service
Router(config-telephony)# max-dn 150 preference 1
```

The following example sets the maximum number of extensions to 200 and specifies that they should not register both primary and secondary numbers with the H.323 gatekeeper:

```
Router(config)# telephony-service
Router(config-telephony)# max-dn 200 no-reg both
```

The following example sets the maximum number of extensions to 200 and specifies that ephone-dn 36 should not register its primary number with the gatekeeper:

```
Router(config)# telephony-service
Router(config-telephony)# max-dn 200
Router(config-telephony)# exit
Router(config)# ephone-dn 36
Router(config-ephone-dn)# number 75373 no-reg primary
```

Related Commands

Command	Description
ephone-dn	Enters ephone-dn configuration mode.
max-ephones	Sets the maximum number of phones supported by the router.
number	Associates a telephone or extension number with an ephone-dn.
telephony-service	Enters telephony-service configuration mode.

max-dn (voice register global)

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

max-dn *max-directory-numbers*

no max-dn

Syntax Description

max-directory-numbers	Maximum number of extensions (ephone-dns) supported by the Cisco router. The maximum number is version and platform dependent; type **?** to display range. Range is 1 to 150. Default is 150.

Defaults

150

Command Modes

Voice register global configuration

Command History

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

Usage Guidelines

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

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

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

Examples

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

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

Related Commands

Command	Description
voice register dn	Enters voice register dn configuration mode to define an extension for a SIP phone line.

Command	Description
max-pool (voice register global)	Sets the maximum number of SIP voice register pools that are supported in a Cisco SIP SRST or Cisco CME environment.
voice register global	Enters voice register global configuration mode in order to set global parameters for all supported Cisco SIP phones in a Cisco CME or Cisco SIP SRST environment.

max-ephones

To set the maximum number of Cisco IP phones to be supported by a Cisco CallManager Express (Cisco CME) router, use the **max-ephones** command in telephony-service configuration mode. To reset this number to the default value, use the **no** form of this command.

max-ephones *max-phones*

no max-ephones

Syntax Description

max-phones	Maximum number of phones supported by the Cisco CME router. The maximum number is version- and platform-dependent; refer to Cisco IOS command-line interface (CLI) help. Default is 0.

Defaults

0 phones

Command Modes

Telephony-service configuration

Command History

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

Usage Guidelines

The **max-ephones** command limits the number of Cisco IP phones supported on the router. The maximum number you can set is platform- and version-dependent. Use CLI help to determine the maximum number of ephones you can set, as shown in this example:

```
Router(config-telephony)# max-ephones ?
  <1-48>  Maximum phones to support
```

The **max-dn** command similarly limits the number of extensions (ephone-dns) in a Cisco CME system.

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

After using this command, configure phones by using the Cisco CME graphic user interface (GUI) or the router CLI in ephone configuration mode.

Examples

The following example sets the maximum number of Cisco IP phones in a Cisco CME system to 24:

```
Router(config)# telephony-service
Router(config-telephony)# max-ephones 24
```

Related Commands

Command	Description
ephone	Enters ephone configuration mode.
max-dn	Sets the maximum number of extensions (ephone-dns) that can be supported by the router.
telephony-service	Enters telephony-service configuration mode.

maximum bit-rate (telephony-video)

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

maximum bit-rate *value*

no maximum bit-rate

Syntax Description

value	Sets the maximum IP phone video bandwidth, in kbps. The range is 0 to 10000000. The default value is 10000000.

Command Default

Maximum bit-rate is 1000000.

Command Modes

Telephony-video configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Use this command to set the maximum bit-rate for all video-capable phones associated with a Cisco Unified CME router.

Examples

The following example sets a maximum bit-rate of 256 kbps.

```
Router(config)# telephony-service
Router(config-telephony)# video
Router(conf-tele-video)# maximum bit-rate 256
```

Related Commands

Command	Description
maximum bit-rate (cm-fallback-video)	Sets the maximum bit-rate for all video-capable phones in a Cisco Unified SRST system.

max-pool (voice register global)

To set the maximum number of Session Initiation Protocol (SIP) voice register pools that are supported in a Cisco Unified SIP Survivable Remote Site Telephony (SRST) or Cisco Unified CallManager Express (Cisco Unified CME) environment, use the **max-pool** command in voice register global configuration mode. To reset the maximum number to the default, use the **no** form of this command.

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

no max-pool

Syntax Description

max-voice-register-pools	Maximum number of SIP voice register pools supported by the Cisco router. The upper limit of voice register pools is version- and platform-dependent; type **?** for range. Default is 0.

Command Default

Default is 0 pools.

Command Modes

Voice register global configuration

Command History

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

Usage Guidelines

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

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

Examples

The following example shows how to set the maximum number of Cisco SIP IP phones in a Cisco Unified SIP SRST or Cisco Unified CME environment to 24:

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

Related Commands

Command	Description
max-dn (voice register global)	Set the maximum number of SIP phone directory numbers (extensions) that are supported by a Cisco Unified CME router.

max-redirect

To change the number of times that a call can be redirected by call forwarding or transfer within a Cisco CallManager Express (Cisco CME) system, use the **max-redirect** command in telephony-service configuration mode. To revert to the default number of redirects, use the **no** form of this command.

max-redirect *number*

no max-redirect

Syntax Description

number	Number of permissible redirects. Range is from 5 to 20. Default is 5.

Defaults

Number of redirects is 5.

Command Modes

Telephony-service configuration

Command History

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

Usage Guidelines

This command supports Cisco CME ephone hunt groups by allowing calls to be redirected more than the default 5 times.

Examples

The following example sets the maximum number of redirects to 8:

```
Router(config)# telephony-service
Router(config-telephony)# max-redirect 8
```

Related Commands

Command	Description
telephony-service	Enters telephony-service configuration mode.

max-subscription

To set the maximum number of concurrent watch sessions that are allowed, use the **max-subscription** command in presence configuration mode. To return to the default, use the **no** form of this command.

max-subscription *number*

no max-subscription

Syntax Description

number	Maximum watch sessions. Range: 100 to 500. Default: 100.

Command Default

Maximum subscriptions is 100.

Command Modes

Presence

Command History

Release	Modification
12.4(11)XJ	This command was introduced.
12.4(15)T	This command was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

This command sets the maximum number of concurrent presence subscriptions for both internal and external subscribe requests.

Examples

The following example shows the maximum subscriptions set to 150:

```
Router(config)# presence
Router(config-presence)# max-subscription 150
```

Related Commands

Command	Description
allow watch	Allows a directory number on a phone registered to Cisco Unified CME to be watched in a presence service.
allow subscribe	Allows internal watchers to monitor external presence entities (directory numbers).
presence	Enables presence service on the router and enters presence configuration mode.
presence enable	Allows incoming presence requests from SIP trunks.
server	Specifies the IP address of a presence server for sending presence requests from internal watchers to external presence entities.
watcher all	Allows external watchers to monitor internal presence entities (directory numbers).

max-timeout

To set the maximum combined timeout for the no-answer periods for all ephone-dns in the ephone-hunt list, use the **max-timeout** command in ephone-hunt configuration mode. To return this value to the default, use the **no** form of this command.

max-timeout *seconds*

no max-timeout *seconds*

Syntax Description

seconds	Number of seconds. Range is from 3 to 60000. Default is unlimited.

Command Default

Number of seconds is unlimited.

Command Modes

Ephone-hunt configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Examples

The following example shows how to set different no-answer timeouts for each ephone-dn in the hunt-group list and no maximum timeout. The first call to the hunt group rings extension 1001. If that extension does not answer in 7 seconds, the call is forwarded to extension 1002. If that extension does not answer after 10 seconds, the call is forwarded to extension 1003. However, if extension 1003 does not answer after 8 seconds, the call is sent to the final number, extension 4500, because the maximum timeout of 25 seconds has been reached.

```
ephone-hunt 3 peer
 pilot 4200
 list 1001, 1002, 1003
 hops 3
 timeout 7, 10, 15
 max-timeout 25
 final 4500
```

Related Commands

Command	Description
ephone-hunt	Defines an ephone hunt group and enters ephone-hunt configuration mode.

mode (voice register global)

To enable the mode for configuring SIP phones in a Cisco Unified CallManager Express (Cisco Unified CME) system, use the **mode cme** command in voice register global configuration mode. To return to the default, use the **no** form of this command.

mode cme

no mode cme

Syntax Description

cme	Only valid keyword is **cme**. This mode determines the commands that are available to configure SIP phones.

Defaults

SIP SRST mode

Command Modes

Voice register global configuration

Command History

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

Usage Guidelines

This command enables Cisco Unified CME on the router for configuration purposes. The router is enabled for Cisco SIP SRST by default. Enable this command before configuring SIP phones in Cisco Unified CME to ensure that all required commands are available.

Examples

The following example shows how to set the mode to Cisco CME:

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

Related Commands

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

moh (ephone-dn)

To enable music on hold (MOH) from an external live audio feed (standard line-level audio connection) connected directly to the router by an foreign office exchange (FXO) or an E&M analog voice port, use the **moh** command in ephone-dn configuration mode. To disable MOH from a live feed or to disable the outcall number or multicast capability, use the **no** form of this command.

moh [**out-call** *outcall-number*] [**ip** *ip-address* **port** *port-number* [**route** *ip-address*]]

no moh [**out-call** *outcall-number* | **ip**]

Syntax Description

out-call *outcall-number*	(Optional) Sets up a call to the outcall number in order to connect to the MOH feed. If this keyword is not used, the live feed is assumed to derive from an incoming call to the ephone-dn under which this command is used.
ip *ip-address*	(Optional) Indicates that this audio stream is to be used as a multicast source as well as the MOH source and specifies the destination IP address for multicast.
port *port-number*	(Optional) Specifies the media port for multicast. Range is from 2000 to 65535. Port 2000 is recommended because this port is already used for normal Real-Time Transport Protocol (RTP) media transmissions between IP phones and the Cisco CallManager Express router.
route *ip-address*	(Optional) Indicates the specific router interface on which to transmit the IP multicast packets. The default is that the MOH multicast stream is automatically output on the interface that corresponds to the address that was configured with the **ip source-address** command.

Defaults

MOH is disabled on an extension.

Command Modes

Ephone-dn configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(11)YT	Cisco ITS 2.1	This command was introduced.
12.2(15)T	Cisco ITS 2.1	This command was integrated into Cisco IOS Release 12.2(15)T.
12.2(15)ZJ	Cisco CME 3.0	The **ip**, **port**, and **route** keywords were added.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Usage Guidelines

This command takes the specified live-feed audio stream and uses it as MOH for a Cisco CallManager Express (CME) system. The connection for the live-feed audio stream is established as an automatically connected voice call. If the **out-call** keyword is used, the type of connection can include VoIP calls if voice activity detection (VAD) is disabled. The typical operation is for the MOH ephone-dn to establish a call to a local router E&M voice port.

Connection via E&M is the recommended mechanism because it requires minimal external components. The E&M port must be placed in 4-wire operation, using E&M immediate signaling and with the **auto-cut-through** option enabled. You directly connect a line-level audio feed (standard audio jack) to pins 3 and 6 of an E&M RJ-45 connector. The E&M WAN interface card (WIC) has a built-in audio transformer that provides appropriate electrical isolation for the external audio source. (The audio connection on the E&M port does not require loop current.) The **signal immediate** and **auto-cut-through** commands disable E&M signaling on this voice port. A G.711 audio packet stream is generated by the digital signal processor (DSP) on the E&M port.

If you are using an FXO voice port for live-feed MOH instead of an E&M port, connect the MOH source to the FXO voice port. This connection requires an external adapter to supply normal telephone company (telco) battery voltage with the correct polarity to the tip-and-ring leads of the FXO port. The adapter must also provide transformer-based isolation between the external audio source and the tip-and-ring leads of the FXO port.

Music from a live feed is continuously fed into the MOH playout buffer instead of being read from an audio file in flash memory. There is typically a two-second delay with live-feed MOH.

If the **out-call** keyword is used, an outbound call to the MOH live-feed source is attempted (or reattempted) every 30 seconds until the call is connected to the ephone-dn (extension) that has been configured for MOH. Note that this ephone-dn is not associated with any physical phone.

If the **moh** (ephone-dn) command is used without any keywords or arguments, the ephone-dn will accept an incoming call and use the audio stream from the call as the source for the MOH stream, displacing any audio stream that is available from a flash file. To accept an incoming call, the ephone-dn must have an extension or phone number configured for it. A typical usage would be for an external H.323-based server device to call the ephone-dn to deliver an audio stream to the Cisco CME system. Normally, only a single ephone-dn would be configured like this. If there is more than one ephone configured to accept incoming calls for MOH, the first ephone-dn that is successfully connected to a call (incoming or outgoing) is the MOH source for the system.

MOH can also be derived from an audio file when you use the **moh** command in telephony-service configuration mode with the *filename* argument. There can be only one MOH stream at a time in a Cisco CME system, and if both an audio file and a live feed have been specified for the MOH stream, the router seeks the live feed from the **moh (ephone-dn)** command first. If the live feed is found, the router displaces the audio file source. If the live feed is not found or fails at any time, the router falls back to the audio file source that was specified in the **moh (telephony-service)** command.

If you use the **ip** keyword to specify a multicast address in this command, the audio stream is sent to the multicast address in addition to serving as the MOH source. Additionally, if you specify a different multicast address using the **multicast moh** command under telephony-service configuration mode, the audio stream is also sent to the multicast address that you named in that command. It is therefore possible to send the live-feed audio stream to MOH and to two different multicast addresses: the one that is directly configured under the **moh (ephone-dn)** command and the one that is indirectly configured under the **multicast moh** (telephony-service) command.

A related command, the **feed** command, provides the ability to multicast an audio stream that is not the MOH audio stream.

Note IP phones do not support multicast at 224.x.x.x addresses.

Examples

The following example establishes a live music-on-hold source by setting up a call to extension 7777:

```
Router(config)# ephone-dn 55
Router(config-ephone-dn)# moh out-call 7777
```

Related Commands

Command	Description
auto-cut-through	Enables call completion when an M-lead response is not provided.
ephone-dn	Enters ephone-dn configuration mode to set directory numbers and parameters for individual Cisco IP phone extensions.
feed	Enables multicast of an audio stream that is different from the music-on-hold audio stream.
ip source-address	Identifies the IP address and port through which IP phones communicate with a Cisco CME router.
moh (telephony-service)	Enables music on hold from an audio file.
multicast moh	Enables multicast of the music-on-hold audio stream.
signal	Specifies the type of signaling for a voice port.

moh (telephony-service)

To generate an audio stream from a file for music on hold (MOH) in a Cisco CallManager Express (Cisco CME) system, use the **moh** command in telephony-service configuration mode. To disable the MOH audio stream from this file, use the **no** form of this command.

moh *filename*

no moh

Syntax Description

filename	Name of the audio file to use for the MOH audio stream. The file must be copied to flash memory on the Cisco CME router.

Defaults

Tone on hold (a periodic beep is played to the caller)

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(2)XT	Cisco ITS 2.0	This command was introduced on the Cisco 1750, Cisco 1751, Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	Cisco ITS 2.0	This command was implemented on the Cisco 2600-XM and Cisco 2691.

Usage Guidelines

This command enables MOH from .au and .wav format music files. MOH is played for G.711 callers and on-net VoIP and PSTN callers who are on hold in a Cisco CME system. Local callers within a Cisco CME system hear a repeating tone while they are on hold.

Audio files that are used for MOH must be copied to the Cisco CME router flash memory. A MOH file can be in .au or .wav file format; however, the file format must contain 8-bit 8-kHz data in A-law or mu-law data format.

If you want to replace or modify the audio file that is currently specified, you must first disable the MOH capability using the **no moh** command. The following example replaces file1 with file2:

```
Router(config-telephony)# moh file1
Router(config-telephony)# no moh
Router(config-telephony)# moh file2
```

If you specify a second file without first removing the original file, the MOH mechanism stops working and may require a router reboot to clear the problem.

A related command, the **moh** command in ephone-dn configuration mode, can be used to establish a MOH audio stream from a live feed. If you configure both commands, MOH falls back to playing music from the audio file if the live music feed is interrupted.

The **multicast moh** command allows you to use the MOH stream for a multicast broadcast.

When the **multicast moh** and **debug ephone moh** commands are both enabled, if you also use the **no moh** command, the debug output can be excessive and flood the console. Multicast MOH should be disabled before using the **no moh** command when the **debug ephone moh** command is enabled.

Examples

The following example enables music on hold and specifies a music file:

```
Router(config)# telephony-service
Router(config-telephony)# moh minuet.wav
```

Related Commands

Command	Description
debug ephone moh	Displays diagnostic information for music on hold.
moh (ephone-dn)	Enables music on hold from a live audio feed.
multicast moh	Enables multicast of the music-on-hold audio stream.
telephony-service	Enters telephony-service configuration mode.

mtp

To send media packets from an IP phone to the Cisco Unified CME router, use the **mtp** command in ephone or ephone-template configuration mode. To return to the default, use the **no** form of this command.

mtp

no mtp

Syntax Description

This command has no keywords or arguments.

Command Default

An IP phone in a call with another IP phone in the same Cisco Unified CME system sends media packets directly to the other phone.

Command Modes

Ephone configuration
Ephone-template configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Normally, media packets (RTP) packets that are sent between IP phones in the same Cisco Unified CME system go directly to the other phone and do not travel through the Cisco Unified CME router. When these packets are sent from a remote IP phone to another IP phone in the same Cisco Unified CME system, they may be obstructed by a firewall. The **mtp** command instructs a phone to always send its media packets to the Cisco Unified CME router, which acts as a proxy and forwards the packets to the destination. Firewalls can then be easily configured to pass the RTP packets because the router uses a specified UDP port for media packets. In this way, RTP packets from remote IP phones can be delivered to IP phones on the same system. The default is that this function is off and that RTP packets that are sent from one IP phone to another IP phone in the same Cisco Unified CME system go directly to the other phone.

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

Examples

The following example sends media packets from ephone 437 to the Cisco Unified CME router for all calls:

```
ephone 437
 button 1:29
 mtp
```

multicast moh

To use the music-on-hold (MOH) audio stream as a multicast source in a Cisco CME system, use the **multicast moh** command in telephony-service configuration mode. To disable multicast use of the MOH stream, use the **no** form of this command.

multicast moh *ip-address* **port** *port-number* [**route** *ip-address-list*]

no multicast moh

Syntax Description

ip-address	Specifies the destination IP address for multicast.
port *port-number*	Specifies the media port for multicast. Range is from 2000 to 65535. Port 2000 is recommended because this port is already used for normal Real-Time Transport Protocol (RTP) media transmissions between IP phones and the Cisco CME router.
route *ip-address-list*	(Optional) Indicates specific router interfaces over which to transmit the IP multicast packets. Up to four IP addresses can be listed, each separated from the other by a space. The default is that the MOH multicast stream is automatically output on the interfaces that correspond to the address that was configured with the **ip source-address** command.

Defaults

No multicast is enabled.

Command Modes

Telephony-service configuration

Command History

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

Usage Guidelines

This command enables multicast of the audio stream that is designated for MOH in a Cisco CME system. A related command, the **moh (ephone-dn)** command, creates a MOH audio stream from an external live feed and optionally enables multicast on that stream. These two commands can be used concurrently to provide multicast of a live-feed MOH audio stream to two different multicast addresses.

Another related command, the **feed** command, enables multicast of an audio stream that is not the MOH audio stream.

When the **multicast moh** and **debug ephone moh** commands are both enabled, if you also use the **no moh** command, the debug output can be excessive and flood the console. Multicast MOH should be disabled before using the **no moh** command when the **debug ephone moh** command is enabled.

Note IP phones do not support multicast at 224.x.x.x addresses.

Examples

The following example enables multicast of the MOH audio stream at multicast address 239.10.16.4 and names two router interfaces over which to send the multicast packets.

```
Router(config)# telephony-service
Router(config-telephony)# moh minuet.au
Router(config-telephony)# multicast moh 239.10.16.4 port 2000 route 10.10.29.17
10.10.29.33
```

Related Commands

Command	Description
feed	Enables multicast of an audio stream that is not the music-on-hold audio stream.
ip source-address	Identifies the IP address and port through which IP phones communicate with a Cisco CME router.
moh (ephone-dn)	Enables music on hold from a live audio feed.
moh (telephony-service)	Enables music on hold from an audio file.
telephony-service	Enters telephony-service configuration mode.

multicast-moh

To enable multicast music on hold (MOH) on a phone in a Cisco Unified CME system, use the **multicast-moh** command in ephone or ephone-template configuration mode. To disable multicast MOH per phone, use the **no** form of this command.

multicast-moh

no multicast-moh

Syntax Description

This command has no keywords or arguments.

Command Default

Multicast MOH is enabled.

Command Modes

Ephone configuration
Ephone-template configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is enabled by default.

The **no** form of this command is used to disable multicast MOH for phone types that do not support IP multicast and therefore do not support multicast MOH.

Examples

The following example shows how to disable multicast MOH for ephone 71:

```
Router(config)# ephone 71
Router(config-ephone)# no multicast-moh
```

The following example shows how to use an ephone template to disable multicast MOH for ephone 2:

```
Router(config)# ephone-template 1
Router(config-ephone-template)# no multicast-moh
Router(config-ephone-template)# exit
Router(config)# ephone 2
Router(config-ephone)# button 1:21 2:22
Router(config-ephone)# ephone-template 1
```

Related Commands

Command	Description
multicast moh	Enables multicast of the music-on-hold audio stream.

mwi (ephone-dn and ephone-dn-template)

To enable a specific Cisco Unified IP phone extension (ephone-dn) to receive message-waiting indication (MWI) notification from an external voice-messaging system, use the **mwi** command in ephone-dn or ephone-dn-template configuration mode. To disable this feature, use the **no** form of this command.

mwi {**off** | **on** | **on-off**}

no mwi {**off** | **on** | **on-off**}

Syntax Description

off	Sets a Cisco Unified IP phone extension to process MWI to OFF, using either the main or secondary phone number.
on	Sets a Cisco Unified IP phone extension to process MWI to ON, using either the main or secondary phone number.
on-off	Sets a Cisco Unified IP phone extension to process MWI to both ON and OFF, using either the main or secondary phone number.

Command Default

MWI notification is disabled on an extension.

Command Modes

Ephone-dn configuration
Ephone-dn-template configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(2)XT	Cisco ITS 2.0	This command was introduced on the following platforms: Cisco 1750, Cisco 1751, Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	Cisco ITS2.0	This command was implemented on the Cisco 2600-XM and Cisco 2691.
12.4(4)XC	Cisco Unified CME 4.0	This command was made available in ephone-dn-template configuration mode.
12.4(9)T	Cisco Unified CME 4.0	This command in ephone-dn-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command enables a Cisco Unified IP phone extension to receive MWI notification from an external voice-messaging system for all the Cisco Unified IP phones connected to the Cisco Unified CME router. This extension is a “dummy” extension and is not associated with any physical phone. The external voice-messaging system is able to communicate MWI status by making telephone calls to the dummy extension number, with the MWI information embedded in either the called or calling-party IP phone number.

This command cannot be used unless the **number** command is already configured for this extension (ephone-dn).

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

Examples

The following example sets MWI to on:

```
Router(config)# ephone-dn 1
Router(config-ephone-dn) number 8000
Router(config-ephone-dn) mwi on
```

The following example sets MWI to off:

```
Router(config)# ephone-dn 2
Router(config-ephone-dn) number 8001
Router(config-ephone-dn) mwi off
```

The following example sets MWI to both on and off for the primary and secondary number, where the MWI information is embedded in the calling-party number. A call placed by the voice-mail system to 8002 turns on the MWI light for the extension number indicated by the calling-party number for the MWI call. A call placed to 8003 turns off the MWI light.

```
Router(config)# ephone-dn 3
Router(config-ephone-dn) number 8002 secondary 8003
Router(config-ephone-dn) mwi on-off
```

The following example sets MWI to both on and off for the primary and secondary number, where the MWI information is embedded in the called-party number. A call placed by the voice-mail system to 8000*5001*1 turns on the MWI light for extension 5001. A call placed to 8000*5001*2 turns off the MWI light.

```
Router(config)# ephone-dn 20
Router(config-ephone-dn) number 8000*....*1 secondary 8000*....*2
Router(config-ephone-dn) mwi on-off
```

The following example uses an ephone-dn-template to set MWI to on:

```
Router(config)# ephone-dn-template 4
Router(config-ephone-dn-template) mwi on
Router(config-ephone-dn-template)# exit
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 8000
Router(config-ephone-dn)# ephone-dn-template 4
```

Related Commands

Command	Description
ephone-dn	Enters ephone-dn configuration mode.
mwi expires	Sets the expiration timer for registration for either the client or the server.
mwi sip (ephone-dn)	Subscribes an extension in a Cisco Unified CME router to receive MWI notification from a SIP MWI server.

Command	Description
mwi sip-server (telephony-service)	Configures the IP address and port number for an external SIP-based MWI server.
number	Associates a telephone or extension number with an extension (ephone-dn) in a Cisco Unified CME system.

mwi (voice register dn)

To enable a specific Cisco IP phone extension (ephone-dn) using a SIP phone to receive message-waiting indication (MWI) notification, use the **mwi** command in voice register dn configuration mode. To return to the default, use the **no** form of this command.

mwi

no mwi

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled

Command Modes

Voice register dn configuration

Command History

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

Examples

The following example shows how to enable MWI:

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

Related CommandsR

Command	Description
voice register dn	Enters voice register dn configuration mode to define an extension for a SIP phone line.

mwi expires

To set the expiration timer for registration for the message-waiting indication (MWI) client or server, use the **mwi expires** command in telephony-service configuration mode. To disable the timer, use the **no** form of this command.

mwi expires *seconds*

no mwi expires *seconds*

Syntax Description

seconds	Expiration time, in seconds. Range is from 600 to 99999. Default is 86400 (24 hours).

Defaults

86400 seconds (24 hours)

Command Modes

Telephony-service configuration

Command History

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

Examples

The following example sets the expiration timer to 1000 seconds:

```
Router(config)# telephony-service
Router(config-telephony)# mwi expires 1000
```

Related Commands

Command	Description
mwi relay (telephony-service)	Enables the Cisco CME router to relay MWI information to remote Cisco IP phones.
mwi sip-server (telephony-service)	Configures the IP address and port number for the external SIP-based MWI server.
telephony-service	Enters telephony-service configuration mode.

mwi prefix

To specify a prefix for an extension that will receive unsolicited message-waiting indication (MWI) from an external SIP-based MWI server, use the **mwi prefix** command in telephony-service configuration mode. To return to the default, use the **no** form of this command.

mwi prefix *prefix-string*

no mwi prefix

Syntax Description

prefix-string	Digits at the beginning of a number that will be recognized as a prefix before a Cisco Unified CME extension number. The maximum prefix length is 32 digits.

Command Default

A prefix is not defined.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Central voice-messaging servers that provide mailboxes for several Cisco Unified CME sites may use site codes or prefixes to distinguish among similarly numbered ranges of extensions at different sites. In Cisco Unified CME 4.0 and later versions, you can specify that your Cisco Unified CME system should accept unsolicited SIP Notify messages for message-waiting indication (MWI) that include a prefix string as a site identifier.

For example, an MWI message might indicate that the central mailbox number 5551234 has a voice message. In this example, the digits 555 are set as the prefix string or site identifier using the **mwi prefix** command. The local Cisco Unified CME system is able to convert 5551234 to 1234 and deliver the MWI to the correct phone. Without this prefix string manipulation, the system would reject an MWI indication for 5551234 as not matching the local Cisco Unified CME extension 1234.

Examples

The following example identifies the SIP server for MWI notification at the IP address 172.16.14.22. It states that the Cisco Unified CME system will accept unsolicited SIP Notify messages for known mailbox numbers (extension numbers) that are prefixed with the digits 555.

```
sip-ua
 mwi-server 172.16.14.22 unsolicited

telephony-service
 mwi prefix 555
```

Related Commands

Command	Description
mwi (ephone-dn)	Configures specific Cisco Unified IP phone directory numbers to receive MWI notification from an external voice-mail system.
mwi-server	Configures MWI server parameters.
mwi sip (ephone-dn)	Subscribes an extension in a Cisco Unified CME router to receive MWI notification from a SIP MWI server.
telephony-service	Enters telephony-service configuration mode.

mwi qsig

To enable Cisco Unified CME to interrogate a QSIG message center for the message-waiting indication (MWI) status of an IP phone extension, use the **mwi qsig** command in ephone-dn or ephone-dn-template configuration mode. To return to the default, use the **no** form of this command.

mwi qsig

no mwi qsig

Syntax Description

This command has no arguments or keywords.

Command Default

An extension is not subscribed to receive MWI using QSIG.

Command Modes

Ephone-dn configuration
Ephone-dn-template configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

The **transfer-system** command must be used with the **full-consult** or **full-blind** keyword to enable H.450 call forwarding.

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

Examples

In the following example, a voice mail extension (7000) and a normal extension (7582) are defined. Calls are forwarded to voice mail when extension 7582 is busy or does not answer. The message-waiting indicator (MWI) on extension 7582's phone is subscribed to receive notifications from the QSIG message center.

```
ephone-dn 25
 number 7582
 mwi qsig
 call-forward busy 7000
 call-forward noan 7000 timeout 20

telephony-service
 voicemail 7000
 transfer-system full-consult
```

Related Commands

Command	Description
transfer-system	Specifies the call transfer method for Cisco Unified CME extensions.
voicemail	Defines the telephone number that is speed-dialed when the Messages button on a Cisco Unified IP phone is pressed.

mwi reg-e164

To register E.164 numbers rather than extension numbers with a Session Interface Protocol (SIP) proxy or registrar, use the **mwi reg-e164** command in telephony-service configuration mode. To return to the default, use the **no** form of this command.

mwi reg-e164

no mwi reg-e164

Syntax Description

This command has no keywords or arguments.

Command Default

Registering extension numbers with the SIP proxy or registrar.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)T7 12.4	Cisco CME 3.3	This command was introduced.

Usage Guidelines

This command is used when setting up extensions to use an external SIP-based message-waiting indication (MWI) server. The **mwi-server** command in SIP user-agent configuration mode specifies other settings for MWI service.

Examples

The following example specifies that E.164 numbers should be used for registration with the SIP proxy or registrar:

```
telephony-service
 mwi reg-e164
```

Related Commands

Command	Description
mwi-server (SIP user-agent)	Specifies voice-mail server settings on a voice gateway or user agent (UA).

mwi reg-e164 (voice register global)

To configure a gateway to register or deregister a fully-qualified dial-peer E.164 address with a gatekeeper, use the **mwi reg-164** command in voice register global configuration mode. To return to the default, use the **no** form of this command.

mwi reg-e164

no mwi reg-e164

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled

Command Modes

Voice register global configuration

Command History

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

Examples

The following example shows how to enable MWI stutter:

```
Router(config)# voice register global
Router(config-register-global)# mwi reg-e164
```

Related Commands

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

mwi relay

To enable a Cisco CallManager Express (Cisco CME) router to relay message-waiting indication (MWI) notification to remote Cisco IP phones, use the **mwi relay** command in telephony-service configuration mode. To disable MWI relay, use the **no** form of this command.

mwi relay

no mwi relay

Syntax Description

This command has no arguments or keywords.

Defaults

MWI is not enabled.

Command Modes

Telephony-service configuration

Command History

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

Usage Guidelines

Use this command to enable the Cisco CME router to relay MWI notification to remote Cisco IP phones. The router at the central site acts as a notifier after this command is used.

Examples

The following example enables MWI relay:

```
Router(config)# telephony-service
Router(config-telephony)# mwi relay
```

Related Commands

Command	Description
mwi expires	Sets the expiration timer for registration for the client or the server.
show mwi relay clients	Displays registration information for MWI relay clients.
telephony-service	Enters telephony-service configuration mode.

mwi sip

To subscribe an extension in a Cisco Unified CME system to receive message-waiting indication (MWI) from a SIP-based MWI server, use the **mwi sip** command in ephone-dn or ephone-dn-template configuration mode. To remove the configuration, use the **no** form of this command.

mwi sip

no mwi sip

Syntax Description

This command has no arguments or keywords.

Command Default

An extension is not subscribed to receive MWI.

Command Modes

Ephone-dn configuration
Ephone-dn-template configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(2)XT	Cisco ITS 2.0	This command was introduced on the following platforms: Cisco 1750, Cisco 1751, Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	Cisco ITS 2.0	This command was implemented on the Cisco 2600-XM and Cisco 2691.
12.4(4)XC	Cisco Unified CME 4.0	This command was made available in ephone-dn-template configuration mode.
12.4(9)T	Cisco Unified CME 4.0	This command in ephone-dn-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Use this command to subscribe an extension in a Cisco Unified CME router to receive MWI notification from a SIP-based MWI server, and use the **mwi sip-server** command to specify the IP address and port number for the external SIP-based MWI server. This function integrates a Cisco Unified CME router with a SIP-protocol-based MWI service.

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

Examples

The following example subscribes extension 5001 to receive MWI notification from an external Session Initiation Protocol (SIP) MWI server and requests the SIP MWI server to send MWI notification messages through SIP to the Cisco Unified CME router for extension 5001:

```
Router(config) ephone-dn 1
Router(config-ephone-dn) number 5001
Router(config-ephone-dn) name MWI
Router(config-ephone-dn) mwi sip

Router(config) telephony-service
Router(config-telephony) mwi sip-server 172.30.0.5
```

Related Commands

Command	Description
ephone-dn	Enters ephone-dn configuration mode.
ephone-dn-template	Enters ephone-dn-template configuration mode.
mwi sip-server (telephony-service)	Configures the IP address and port number for the external SIP-based MWI server.
show mwi relay clients	Displays registration information for MWI relay clients.

mwi sip-server

To configure parameters associated with an external SIP-based message-waiting indication (MWI) server, use the **mwi sip-server** command in telephony-service configuration mode. To disable MWI server functionality, use the **no** form of this command.

mwi sip-server *ip-address* [**transport tcp** | **transport udp**] [**port** *port-number*] [**reg-e164**] [**unsolicited** [**prefix** *prefix-string*]]

no mwi sip-server *ip-address* [**transport tcp** | **transport udp**] [**port** *port-number*] [**reg-e164**] [**unsolicited** [**prefix** *prefix-string*]]

Syntax Description

ip-address	IP address of the MWI server.
transport tcp	(Optional) Selects TCP as the transport layer protocol. This is the default transport protocol.
transport udp	(Optional) Selects UDP as the transport layer protocol. The default if these keywords are not used is TCP.
port *port-number*	(Optional) Specifies port number for the MWI server. Range is from 2000 to 9999. Default is 5060 (SIP standard port).
reg-e164	(Optional) Registers an E.164 number with a Session Interface Protocol (SIP) proxy or registrar rather than an extension number. Registering with an extension number is the default.
unsolicited	(Optional) Sends SIP Notify message for MWI without any need to send a Subscribe message from the Cisco Unified CME router.
prefix *prefix-string*	(Optional) Allows the specified digits to be present before a recognized Cisco Unified CME extension number. The maximum prefix length is 32 digits.

Command Default

An external SIP-based MWI server is not defined.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(2)XT	Cisco ITS 2.0	This command was introduced on the following platforms: Cisco 1750, Cisco 1751, Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	Cisco ITS 2.0	This command was implemented on the Cisco 2600-XM and Cisco 2691.
12.2(15)ZJ	Cisco CME 3.0	The **unsolicited** keyword was added.

Cisco IOS Release	Cisco Product	Modification
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.4(4)XC	Cisco Unified CME 4.0	The **prefix** *prefix-string* keyword-argument pair was added.
12.4(9)T	Cisco Unified CME 4.0	The **prefix** *prefix-string* keyword-argument pair was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Use this command to configure the IP address of an external SIP MWI server. This IP address is used with the **mwi sip** (ephone-dn) command to subscribe individual ephone-dn extension numbers to the notification list of the MWI SIP server. A SIP MWI client runs TCP by default.

The **transport tcp** keyword is the default setting. The **transport udp** keyword allows you to integrate with a SIP MWI client. The optional **port** keyword is used to specify a port number other than 5060, the default. The default registration is with an extension number, so the **reg-e164** keyword allows you to register with an E.164 ten-digit number.

Central voice-messaging servers that provide mailboxes for several Cisco Unified CME sites may use site codes or prefixes to distinguish among similarly numbered ranges of extensions at different sites. In Cisco CME 3.2.3 and later versions, you can specify that your Cisco Unified CME system should accept unsolicited SIP Notify messages for message-waiting indication (MWI) that include a prefix string as a site identifier.

Examples

The following example sets MWI for the SIP server and sets individual ephone-dn extension numbers to the MWI SIP server's notification list:

```
Router(config) ephone-dn 1
Router(config-ephone-dn) number 5001
Router(config-ephone-dn) name Accounting
Router(config-ephone-dn) mwi sip
Router(config-ephone-dn) exit
Router(config) telephony-service
Router(config-telephony) mwi sip-server 192.168.0.5 transport udp
```

The following example identifies the SIP server for MWI notification at the IP address 172.16.14.22. It states that the Cisco Unified CME system will accept unsolicited SIP Notify messages that include the prefix 555 as a site identifier.

```
telephony-service
 mwi sip-server 172.16.14.22 unsolicited prefix 555
```

Related Commands

Command	Description
mwi (ephone-dn)	Configures specific Cisco Unified IP phone directory numbers to receive MWI notification from an external voice-mail system.
mwi expires	Sets the expiration timer for registration for the client or the server.
mwi sip (ephone-dn)	Subscribes an extension in a Cisco Unified CME router to receive MWI notification from a SIP MWI server.
show mwi relay clients	Displays the registration information for MWI relay clients.
telephony-service	Enters telephony-service configuration mode.

mwi stutter (voice register global)

To generate a stutter tone for message-waiting indication (MWI) in a Cisco CallManager Express (Cisco CME) system using SIP, use the **mwi stutter** command in voice register global configuration mode. To disable MWI stutter, use the **no** form of this command.

mwi stutter

no mwi stutter

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled

Command Modes

Voice register global configuration

Command History

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

Usage Guidelines

Use this command to enable the Cisco CME router to relay MWI notification to remote Cisco IP phones. The router at the central site acts as a notifier after this command is used.

Examples

The following example shows how to enable MWI stutter:

```
Router(config)# voice register global
Router(config-register-global)# mwi stutter
```

Related Commands

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

mwi-line

To designate a line other than the primary line of an ephone to be associated with the ephone's message waiting indicator (MWI) lamp, use the **mwi-line** command in ephone configuration mode. To return to the default, use the **no** form of this command.

mwi-line *line-number*

no mwi-line

Syntax Description

line-number	Line number to be associated with the MWI lamp. Range is from 1 to 34. For information about line numbers, see the "Usage Guidelines" and "Examples" sections.

Command Default

A phone's MWI lamp is lit only when there is a message waiting for the phone's primary line (line 1).

Command Modes

Ephone configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command designates a phone line other than the primary line to activate the MWI lamp on the phone. When a message is waiting for an ephone-dn associated with the designated line, the MWI lamp is turned on. When the message is heard, the MWI lamp is turned off. For phone lines other than the line that is designated to receive MWI, an envelope icon is displayed next to them when there is a message waiting.

Note that a logical phone "line" is not the same as a phone button. A line is a button that has one or more ephone-dns assigned to it. A button that has no ephone-dns assigned to it does not count as a line. For examples of line numbers in different phone configurations, see the "Examples" section.

In most cases, one ephone-dn is assigned to one button on an ephone. When you set the **mwi-line** command to that button, the MWI lamp is turned on when there is a message waiting for that ephone-dn. When you set the **mwi-line** command to a button with a more complex configuration, the following rules apply:

- When a button has a single ephone-dn with primary and secondary numbers, the MWI lamp is turned on only when there is a message waiting for the primary number.
- When a button has several ephone-dns overlaid on it, the MWI lamp is turned on only when there is a message waiting for the first number in the list of ephone-dns.
- When a button is an overflow button for an overlay button, the MWI lamp is not turned on for any extension that might overflow to this button. If you set the **mwi-line** command to this button, the command is ignored.

Examples

The following example enables MWI on ephone 18 for line 2 (button 2), which has overlaid ephone-dns. The MWI lamp on this phone will be lit only if there is a message waiting for extension 2021. Button 4 is unused. The line numbers in this example are as follows:

- Line 1—Button 1—Extension 2020
- Line 2—Button 2—Extension 2021, 2022, 2023, 2024, 2025
- Line 3—Button 3—Extension 2021, 2022, 2023, 2024, 2025 (rollover line)
- Button 4—Unused
- Line 4—Button 5—Extension 2026

```
ephone-dn 20
 number 2020

ephone-dn 21
 number 2021

ephone-dn 22
 number 2022

ephone-dn 23
 number 2023

ephone-dn 24
 number 2024

ephone-dn 25
 number 2025

ephone-dn 26
 number 2026

ephone 18
 button 1:20 2o21,22,23,24,25 3x2 5:26
 mwi-line 2
```

The following example enables MWI on ephone 17 for line 3 (extension 609). In this case, the button numbers do not match the line numbers because buttons 2 and 4 are not used.

- Line 1—Button 1—Extension 607
- Button 2—Unused
- Line 2—Button 3—Extension 608
- Button 4—Unused
- Line 3—Button 5—Extension 609

```
ephone-dn 17
 number 607

ephone-dn 18
 number 608

ephone-dn 19
 number 609

ephone 25
 button 1:17 3:18 5:19
 mwi-line 3
```

mwi-type

To specify the type of message-waiting indication (MWI) notification that a directory number can receive and process, use the **mwi-type** command in ephone-dn or ephone-dn-template configuration mode. To disable this feature, use the **no** form of this command.

mwi-type {**visual** | **audio** | **both**}

no mwi-type {**visual** | **audio** | **both**}

Syntax Description

visual	Sets a directory number to process visual MWI, using either the main or secondary phone number.
audio	Sets a directory number to process audible MWI (AMWI), using either the main or secondary phone number.
both	Sets a directory number to process both visual and audible MWI, using either the main or secondary phone number.

Command Default

If MWI is enabled for a directory number, directory number will receive visual MWI.

Command Modes

Ephone-dn configuration
Ephone-dn-template configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(6)XE	Cisco Unified CME 4.0(2)	This command was introduced.
12.4(4)XC4	Cisco Unified CME 4.0(3)	This command was introduced.
12.4(11)T	Cisco Unified CME 4.0(3)	This command was integrated into Cisco IOS Release 12.4(11)T.

Usage Guidelines

This command enables a directory number to receive audible, visual, or both audible and visual MWI notification from an external voice-messaging system. The external voice-messaging system is able to communicate MWI status by making telephone calls to the dummy extension, with the MWI information embedded in either the called or calling-party IP phone number.

The Cisco Unified CME applies the following logic based on the capabilities of the IP phone and how the **mwi-type** command is configured:

- If the phone supports (visual) MWI and MWI is configured for the phone, turn on the Message Waiting light
- If the phone supports (visual) MWI only, turn on the Message Waiting light regardless of the configuration.
- If the phone supports AMWI and AMWI is configured for the phone, send the stutter dial tone to the phone when it goes off-hook.

- If the phone supports AMWI only and AMWI is configured, send the stutter dial tone to the phone when it goes off hook regardless of the configuration.
- If a phone supports (visual) MWI and AMWI and both options are configured for the phone, turn on the Message Waiting light and send the stutter dial tone to the phone when it goes off-hook.

Before using this command:

- Create the directory number to be configured by using the **number** command
- Enable MWI on this directory number by using the **mwi** command.

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

Examples

The following example shows how to enable AMWI on extension 8000, assuming that the phone to which this directory number is assigned supports AMWI. Otherwise, a call placed by the voice-mail system to 8001 turns on the MWI light for the extension number indicated by the calling-party number for the MWI call.

```
Router(config)# ephone-dn 1
Router(config-ephone-dn) number 8000
Router(config-ephone-dn) MWI on
Router(config-ephone-dn) MWI-type audible
```

The following example shows how to enable both audible and visual MWI. A call placed by the voice-mail system to 8001 turns on the MWI light for the extension number indicated by the calling-party number for the MWI call. When the phone user takes the phone off hook, they hear a stutter dial tone:

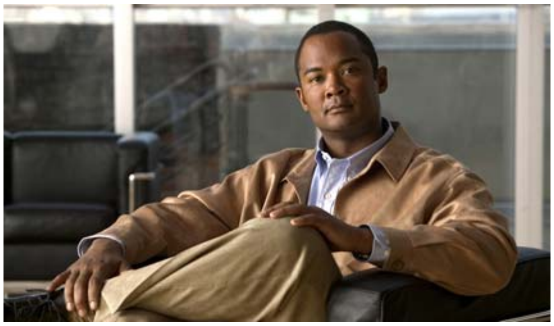
```
Router(config)# ephone-dn 2
Router(config-ephone-dn) number 8001
Router(config-ephone-dn) MWI on
Router(config-ephone-dn) MWI-type both
```

The following example shows how to use an ephone-dn-template to set MWI type:

```
Router(config)# ephone-dn-template 4
Router(config-ephone-dn-template) MWI-type both
Router(config-ephone-dn-template)# exit
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 8000
Router(config-ephone-dn)# ephone-dn-template 4
```

Related Commands

Command	Description
mwi (ephone and ephone template)	Enables a directory number to receive MWI.
number	Associates a telephone or extension number with a directory number in a Cisco Unified CME system.



Cisco Unified CME Commands: N

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This chapter contains commands to configure and maintain Cisco Unified Communications Manager Express (formally known as Cisco Unified CallManager Express). The commands are presented in alphabetical order. Some commands required for configuring Cisco Unified Communications Manager Express (Cisco Unified CME) may be found in other Cisco IOS command references. Use the command reference master index or search online to find these commands.

name (ephone-dn)

To associate a name with a Cisco CallManager Express (Cisco CME) extension (ephone-dn), use the **name** command in ephone-dn configuration mode. To disassociate a name from an extension, use the **no** form of this command.

name *name*

no name

Syntax Description

name	Name of the person associated with a given extension (ephone-dn). Name must follow the order specified in the **directory (telephony-service)** command, either **first-name-first** or **last-name-first**.

Defaults

No default behavior or values

Command Modes

Ephone-dn configuration

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.1(5)YD	1.0	This command was introduced on the following platforms: Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(2)XT	2.0	This command was implemented on the Cisco 1750 and Cisco 1751.
12.2(8)T	2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the following platforms: Cisco 2600XM, Cisco 2691, Cisco 3725, and Cisco 3745.

Usage Guidelines

The *username* argument is used to provide caller ID for calls originating from a Cisco CME extension. This command is also used to generate directory information for the local directory that is accessed from the Directories button on a Cisco IP phone.

Examples

The following example configures the username John Smith with the pattern **first-name-first**:

```
Router(config)# ephone-dn 1
Router(config-ephone-dn) name John Smith
```

The following example configures the username Jane Smith with the pattern **last-name-first**:

```
Router(config)# ephone-dn 1
Router(config-ephone-dn) name Smith, Jane
```

Related Commands

Command	Description
directory (telephony-service)	Defines the name order for the local directory of Cisco IP phone users.
ephone-dn	Enters ephone-dn configuration mode.

name (voice register dn)

To associate a name with a Cisco CallManager Express (Cisco CME) extension (directory number), use the **name** command in voice register dn configuration mode. To disassociate a name from an extension, use the **no** form of this command.

name *name*

no name

Syntax Description

name	Name of the person associated with a given extension. Name must follow the order specified in the **directory** (telephony-service) command, either **first-name-first** or **last-name-first**.

Defaults

No default behavior or values

Command Modes

Voice register dn configuration

Command History

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

Usage Guidelines

The *name* argument is used to provide caller ID for calls originating from a Cisco CME extension. This command is also used to generate directory information for the local directory that is accessed from the Directories button on a Cisco IP phone.

Examples

The following example shows how to configure the username John Smith with the pattern **first-name-first**:

```
Router(config)# voice register dn 1
Router(config-register-dn) name John Smith
```

The following example shows how to configure the username Jane Smith with the pattern **last-name-first**:

```
Router(config)# voice register dn 1
Router(config-register-dn) name Smith, Jane
```

Related Commands

Command	Description
directory (telephony-service)	Defines the name order for the local directory of Cisco IP phone users.
voice register dn	Enters voice register dn configuration mode to define an extension for a SIP phone line.

name (voice user-profile)

To create an authentication credential be used by extension mobility services in Cisco Unified CME, use the username command in voice logout-profile configuration mode. To remove the credential, use the **no** form of this command.

name *username* **password** *password*

no name

Syntax Description

username	Credential to be used by individual phone user to log into a Cisco Unified IP phone.
password	Password to be used with this user name for authentication purposes.
password	Alphanumeric string.

Command Default

Credential does not exist.

Command Modes

Voice logout-profile configuration (config-logout-profile)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XW1	Cisco Unified CME 4.2	This command was introduced.

Usage Guidelines

Use this command in voice user-profile configuration mode to create a credential to be authenticated by Cisco Unified CME before a phone user can log into a Cisco Unified IP phone that is enabled for extension mobility.

When a user logs into an extension mobility enabled phone, Cisco Unified CME will retrieve the appropriate user profile, based on user name and password match, and replace the phone's default logout profile with the user's profile.

Examples

The following example shows the configuration to be downloaded after a user enters the username and password configured in this profile, and Cisco Unified CME matches the entry to the credentials in a user profile database:

```
voice user-profile 1
 pin  12345
 user me password pass123
 number 2001 type silent-ring
 number 2002 type beep-ring
 number 2003 type feature-ring
 number 2004 type monitor-ring
 number 2005,2006 type overlay
 number 2007,2008 type cw-overly
 speed-dial 1 3001
 speed-dial 2 3002 blf
```

network-locale (ephone-template)

To specify a locale tag identifier in an ephone template, use the **network-locale** command in ephone-template configuration mode. To use the default user locale, use the **no** form of this command.

user-locale *language-tag*

no user-locale

Syntax Description

language-tag	Language tag identifier that was assigned to an alternative network locale using the **network-locale (telephony-service)** command.

Command Default

The default network locale (network locale 0) is used.

Command Modes

Ephone-template configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

To apply network locale tag identifiers to individual ephones, you must specify per-phone configuration files using the **cnf-file perphone** command and define the locale tags using the **network-locale (telephony-service)** command.

After creating an ephone template that contains a locale tag identifier, use the **ephone-template (ephone)** command to apply the template to individual ephones.

Examples

The following example defines three alternative locales: JP (Japan), FR (France), and ES (Spain). The default is US for all phones that do not have the alternatives applied using ephone templates. In this example, ephone 11 uses JP for its locales, ephone 12 uses FR, ephone 13 uses ES, and ephone 14 uses the default, US.

```
telephony-service
 cnf-file location flash:
 cnf-file perphone
 create cnf-files
 user-locale 1 JP
 user-locale 2 FR
 user-locale 3 ES
 network-locale 1 JP
 network-locale 2 FR
 network-locale 3 ES

ephone-template 1
 user-locale 1
```

```
 network-locale 1

ephone-template 2
 user-locale 2
 network-locale 2

ephone-template 3
 user-locale 3
 network-locale 3

ephone 11
 button 1:25
 ephone-template 1

ephone 12
 button 1:26
 ephone-template 2

ephone 13
 button 1:27
 ephone-template 3

ephone 14
 button 1:28
```

Related Commands

Command	Description
cnf-file	Specifies the type of configuration files that phones use.
ephone-template (ephone)	Applies an ephone template to an ephone.
network-locale (telephony-service)	Sets the locale for geographically specific tones and cadences.

network-locale (telephony-service)

To select a code for a geographically specific set of tones and cadences on supported phone types, use the **network-locale** command in telephony-service configuration mode. To disable selection of a code, use the **no** form of this command.

network-locale [*network-locale-tag* [*user-defined-code*]] *locale-code*

no network-locale *network-locale-tag*

Syntax Description

network-locale-tag	(Optional) Assigns a locale identifier to the locale code. Range is 0 to 4. Default is 0.
user-defined code	(Optional) Assigns one of the user-defined codes to the specified locale code. Valid codes are **U1**, **U2**, **U3**, **U4**, and **U5**. There is no default.
locale-code	Locale files for the following ISO 3166 codes are predefined in system storage for supported phone types: • **AT**—Austria • **CA**—Canada • **CH**—Switzerland • **DE**—Germany • **DK**—Denmark • **ES**—Spain • **FR**—France • **GB**—United Kingdom • **IT**—Italy • **JP**—Japan • **NL**—Netherlands • **NO**—Norway • **PT**—Portugal • **RU**—Russian Federation • **SE**—Sweden • **US**—United States (default) **Note** You can also assign any valid ISO 3166 code that is not listed above to a user-defined code (U1 through U5), but you must first copy the appropriate XML tone files to flash, slot 0, or an external TFTP server and use the **cnf-files perphone** command to specify the use of per-phone configuration files.

Command Default

The default locale code is **US** (United States).

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(11)YT	Cisco ITS 2.1	This command was introduced.
12.2(15)T	Cisco ITS 2.1	This command was integrated into Cisco IOS Release 12.2(15)T.
12.4(4)XC	Cisco Unified CME 4.0	The *network-locale-tag* and *user-defined-code* arguments were added.
12.4(9)T	Cisco Unified CME 4.0	The *network-locale-tag* and *user-defined-code* arguments were integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

The **show telephony-service tftp-bindings** command displays the locale-specific call-progress tone files that are accessible to IP phones using TFTP.

This command must be followed by a complete phone reboot using the **reset** command.

Network locale 0 always holds the default locale, which is used for all phones that are not assigned alternative network locales or user-defined network locales. The system default is US, but you can define a different code to be the default, as shown in the "Examples" section.

Alternative Network Locales

The *network-locale-tag* argument allows you to specify up to five alternative network locales for use in a system using Cisco Unified CME 4.0 or a later release. For example, a company can specify network-locale France for phones A, B, and C; network-locale Germany for phones D, E, and F; and network-locale United States for phones G, H, and I.

Each one of the five alternative network locales that you can use in a multi-locale system is identified with a locale tag identifier. The identifier 0 always holds the default locale, although you can define this default to be any locale code that is supported in the system and is listed in the CLI help for the command. For example, if you define network locale 0 to be JP (Japanese), the default network locale for the router is JP. If you do not specify a locale for the identifier 0, the default is US (United States).

To apply alternative network locales to different phones, you must use the **cnf-files** command to specify per-phone configuration files. When you use per-phone configuration files, a phone's configuration file automatically uses the default locales in user locale 0 and network locale 0. You can override this default for individual ephones by assigning alternative locale tag identifiers to the alternative locale codes that you want to use and then creating ephone templates to assign the locale tag identifiers to individual ephones. For example, you can give the alternative locale tag of 2 to the locale code DK (Denmark).

After using the **network-locale (telephony-service)** command to associate a locale tag identifier with a locale code, use the **network-locale** command in ephone-template mode to apply the locale tag to an ephone template. Then use the **ephone-template** command in ephone configuration mode to apply the template to the ephones that should use the alternative network locale. For an example, see the Alternative Network Locale Example.

User-Defined Network Locales

XML files for user locales and network locales that are not currently provided in the system must be downloaded to use this feature. Beginning in Cisco Unified CME 4.0, you can install the files to support a particular user and network locale in flash, slot 0, or an external TFTP server. You cannot install these files in the system location. These user-locale and network-locale files can then be used as default or alternative locales for all or some phones.

For example, if you have a site at which the phones should use the displays and tones for Traditional Chinese, which is not one of the predefined locales, you must download and install the XML files for Traditional Chinese on the phones that need to use this locale.

Examples

The following example sets the default locale tag 0 to France:

```
telephony-service
 network-locale FR
```

The following example sets the default locale tag 0 to France. It shows another way to change the default network locale:

```
telephony-service
 network-locale 0 FR
```

The following example sets the alternative locale tag 1 to Germany:

```
telephony-service
 network-locale 1 DE
```

Alternative Network Locale Example

The following example defines three alternative locales: JP (Japan), FR (France), and ES (Spain). The default is US for all phones that do not have the alternatives applied using ephone templates. In this example, ephone 11 uses JP for its locales, ephone 12 uses FR, ephone 13 uses ES, and ephone 14 uses the default, US.

```
telephony-service
 cnf-file location flash:
 cnf-file perphone
 user-locale 1 JP
 user-locale 2 FR
 user-locale 3 ES
 network-locale 1 JP
 network-locale 2 FR
 network-locale 3 ES
 create cnf-files

ephone-template 1
 user-locale 1
 network-locale 1

ephone-template 2
 user-locale 2
 network-locale 2

ephone-template 3
 user-locale 3
 network-locale 3

ephone 11
 button 1:25
 ephone-template 1
```

```
ephone 12
 button 1:26
 ephone-template 2

ephone 13
 button 1:27
 ephone-template 3

ephone 14
 button 1:28
```

User-Defined Network Locale Example

The following example applies the alternative locale tag 4 to the user-defined code U1, which is defined as ZH. ZH is the code that represents Traditional Chinese in ISO 639, the Language Code Reference. Because the code for Traditional Chinese is not one of those is provided in the system, the user must download the appropriate XML files to support this language.

In addition to the user-defined code, the example also defines three alternative locales: JP (Japan), FR (France), and ES (Spain). The default is US for all phones that do not have the alternatives applied using ephone templates. In this example, ephone 11 uses JP for its locales; ephone 12 uses FR; ephone 13 uses ES; ephone 14 uses the default, US; and ephone 15 uses the user-defined language, ZH (Traditional Chinese).

```
telephony-service
 cnf-file location flash:
 cnf-file perphone
 user-locale 1 JP
 user-locale 2 FR
 user-locale 3 ES
 user-locale 4 U1 ZH
 network-locale 1 JP
 network-locale 2 FR
 network-locale 3 ES
 network-locale 4 U1 ZH
 create cnf-files

ephone-template 1
 user-locale 1
 network-locale 1

ephone-template 2
 user-locale 2
 network-locale 2

ephone-template 3
 user-locale 3
 network-locale 3

ephone-template 4
 user-locale 4
 network-locale 4

ephone 11
 button 1:25
 ephone-template 1

ephone 12
 button 1:26
 ephone-template 2
```

```
ephone 13
 button 1:27
 ephone-template 3

ephone 14
 button 1:28

ephone 15
 button 1:29
 ephone-template 4
```

Related Commands

Command	Description
cnf-files	Specifies the type of phone configuration files to be created.
ephone-template (ephone)	Applies an ephone template to an ephone.
network-locale (ephone-template)	Applies a locale tag identifier to an ephone template.
reset (ephone)	Performs a complete reboot of one phone associated with a Cisco Unified CME router.
reset (telephony-service)	Performs a complete reboot of one or all phones associated with a Cisco Unified CME router.
show telephony-service tftp-bindings	Displays the current configuration files that are accessible to IP phones.
telephony-service	Enters telephony-service configuration mode.
user-locale (telephony-service)	Sets the language for displays on supported phone types.

night-service bell

To mark an IP phone to receive night-service bell notification when incoming calls are received on ephone-dns that are marked for night service during night-service time periods, use the **night-service bell** command in ephone or ephone-template configuration mode. To remove night-service notification capability from a phone, use the **no** form of this command.

night-service bell

no night-service bell

Syntax Description

This command has no arguments or keywords.

Command Default

A phone is not marked for night-service bell notification.

Command Modes

Ephone configuration
Ephone-template configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.4(4)XC	Cisco Unified CME 4.0	This command was made available in ephone-template configuration mode.
12.4(9)T	Cisco Unified CME 4.0	This command in ephone-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

When an ephone-dn is marked for night-service treatment using the **night-service bell (ephone-dn)** command, incoming calls that ring during the night-service time period on that ephone-dn send an alert indication to all IP phones that are marked to receive night-service bell notification with this command. The alert notification is in the form of a splash ring (not associated with any of the individual lines on the IP phone) and a visible display of the ephone-dn extension number. The phone user retrieves the call by pressing a PickUp or GPickUp soft key and dialing the appropriate digits.

Night-service periods are defined using the **night-service date** and **night-service day** commands. Night service can be manually disabled or reenabled from a phone configured with ephone-dns in night-service mode if the **night-service code** command has been set.

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

Examples

The following example designates the IP phone that is being configured as a phone that will receive night-service bell notification when ephone-dns marked for night service receive incoming calls during a night-service period:

```
Router(config)# ephone 4
Router(config-ephone)# night-service bell
```

Related Commands

Command	Description
ephone	Enters ephone configuration mode.
night-service bell (ephone-dn)	Marks an ephone-dn to send night-service bell notification to designated IP phones during night-service time periods.
night-service code	Defines a code to disable or reenable night service on IP phones.
night-service date	Defines a recurring time period associated with a month and day during which night service is active.
night-service day	Defines a recurring time period associated with a day of the week during which night service is active.

night-service bell (ephone-dn)

To mark an ephone-dn for night-service treatment, use the **night-service bell** command in ephone-dn configuration mode. To remove the night-service treatment from the ephone-dn, use the **no** form of this command.

night-service bell

no night-service bell

Syntax Description

This command has no arguments or keywords.

Defaults

An ephone-dn is not marked for night service.

Command Modes

Ephone-dn configuration

Command History

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

Usage Guidelines

When an ephone-dn is marked for night-service treatment using this command, incoming calls that ring during the night-service time period on that ephone-dn send an alert indication to all IP phones that are marked to receive night-service bell notification using the **night-service bell (ephone)** command. The alert notification is in the form of a splash ring (not associated with any of the individual lines on the IP phone) and a visible display of the ephone-dn extension number. The phone user retrieves the call by pressing a PickUp or GPickUp soft key and dialing the appropriate digits.

Night-service periods are defined using the **night-service date** and **night-service day** commands. Night service can be manually disabled or reenabled from a phone configured with ephone-dns in night-service mode if the **night-service code** command has been set.

Examples

The following example marks an ephone-dn as a line that will ring on IP phones designated to receive night-service bell notification when incoming calls are received on this ephone-dn during night-service periods:

```
Router(config)# ephone-dn 16
Router(config-ephone-dn)# night-service bell
```

Related Commands

Command	Description
ephone-dn	Enters ephone-dn configuration mode.
night-service bell (ephone)	Marks an IP phone to receive night-service bell notification when incoming calls are received on ephone-dns that are marked for night service during night-service time periods.
night-service code	Defines a code to disable or reenable night service on IP phones.
night-service date	Defines a recurring time period associated with a month and day during which night service is active.
night-service day	Defines a recurring time period associated with a day of the week during which night service is active.

night-service code

To define a code to disable or reenable night service on IP phones, use the **night-service code** command in telephony-service configuration mode. To remove the code, use the **no** form of this command.

night-service code *digit-string*

no night-service code *digit-string*

Syntax Description

digit-string	Digit code that a user enters at an IP phone to disable or reenable night service. The code must begin with an asterisk (*). The maximum number of characters is 16, including the asterisk.

Defaults

No code is defined.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.2(15)ZJ	3.0	This command was introduced.
12.3(4)T	3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.3(14)T	3.3	The action of this command was changed so that all night-service ephone-dns are activated or deactivated when the code is used rather than just the phone on which the code is input.

Usage Guidelines

Night-service periods are defined with the **night-service date** and **night-service day** commands. When an ephone-dn is marked for night-service treatment using the **night-service bell (ephone-dn)** command, incoming calls that ring during the night-service time period on that ephone-dn send an alert indication to all IP phones that are marked to receive night-service bell notification using the **night-service bell (ephone)** command. The alert notification is in the form of a splash ring (not associated with any of the individual lines on the IP phone) and a visible display of the ephone-dn extension number. The phone user retrieves the call by pressing a PickUp or GPickUp soft key and dialing the appropriate digits.

When a night-service code has been defined using the **night-service code** command, night service for all night-service ephone-dns can be manually activated or deactivated from any phone that is configured with a night-service ephone-dn.

Examples

The following example defines a night-service code of *2985:

```
Router(config)# telephony-service
Router(config-telephony)# night-service code *2985
```

Related Commands

Command	Description
night-service bell (ephone)	Marks an IP phone to receive night-service bell notification when incoming calls are received on ephone-dns that are marked for night service during night-service time periods.
night-service bell (ephone-dn)	Marks an ephone-dn to send night-service bell notification to designated IP phones during night-service time periods.
night-service date	Defines a recurring time period associated with a month and day during which night service is active.
night-service day	Defines a recurring time period associated with a day of the week during which night service is active.
telephony-service	Enters telephony-service configuration mode.

night-service date

To define a recurring time period associated with a date during which night service is active, use the **night-service date** command in telephony-service configuration mode. To delete the defined time period, use the **no** form of this command.

night-service date *month day start-time stop-time*

no night-service date *month day start-time stop-time*

Syntax Description

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

Defaults

No time period based on date is defined for night service.

Command Modes

Telephony-service configuration

Command History

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

Usage Guidelines

After you define night-service periods using this command and the **night-service day** command, use the **night-service bell (ephone-dn)** command to specify the extensions that will ring on other phones and the **night-service bell (ephone)** command to specify the phones on which the extensions will ring during the designated night-service periods.

Examples

The following example defines a night-service time period for the entire day of January 1:

```
Router(config)# telephony-service
Router(config-telephony)# night-service date jan 1 00:00 00:00
```

Related Commands

Command	Description
night-service bell (ephone)	Marks an IP phone to receive night-service-bell notification when incoming calls are received on ephone-dns that are marked for night service during night-service time periods.
night-service bell (ephone-dn)	Marks an ephone-dn to send night-service bell notification to designated IP phones during night-service time periods.
night-service code	Defines a code to disable or reenable night service on IP phones.
night-service day	Defines a recurring time period associated with a day of the week during which night service is active.
telephony-service	Enters telephony-service configuration mode.

night-service day

To define a recurring time period associated with a day of the week during which night service is active, use the **night-service day** command in telephony-service configuration mode. To delete the defined time period, use the **no** form of this command.

night-service day *day start-time stop-time*

no night-service day *day start-time stop-time*

Syntax Description

day	Day of the week abbreviation. The following are valid day abbreviations: **sun**, **mon**, **tue**, **wed**, **thu**, **fri**, **sat**.
start-time stop-time	Beginning and ending times for night service, in an HH:MM format using a 24-hour clock. If the stop time is a smaller value than the start time, the stop time occurs on the day following the start time. For example, mon 19:00 07:00 means "from Monday at 7 p.m. until Tuesday at 7 a.m." The value 24:00 is not valid. If 00:00 is entered as a stop time, it is changed to 23:59. If 00:00 is entered for both start time and stop time, night service is in effect for the entire 24-hour period on the specified day.

Defaults

No time period based on day of the week is defined for night service.

Command Modes

Telephony-service configuration

Command History

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

Usage Guidelines

After you define night-service periods using this command and the **night-service date** command, use the **night-service bell (ephone-dn)** command to specify the extensions that will ring on other phones and the **night-service bell (ephone)** command to specify the phones on which the extensions will ring during the designated night-service periods.

Examples

The following example defines a night-service time period from Monday at 7 p.m. to Tuesday at 9 a.m.:

```
Router(config)# telephony-service
Router(config-telephony)# night-service day mon 19:00 09:00
```

Related Commands

Command	Description
night-service bell (ephone)	Marks an IP phone to receive night-service bell notification when incoming calls are received on ephone-dns that are marked for night service during night-service time periods.
night-service bell (ephone-dn)	Marks an ephone-dn to send night-service bell notification to designated IP phones during night-service time periods.
night-service code	Defines a code to disable or reenable night service on IP phones.
night-service date	Defines a recurring time period associated with a month and day during which night service is active.
telephony-service	Enters telephony-service configuration mode.

night-service everyday

To define a recurring time period during which night service is active every day, use the **night-service everyday** command in telephony-service configuration mode. To delete the defined time period, use the **no** form of this command.

night-service everyday *start-time stop-time*

no night-service everyday

Syntax Description

start-time stop-time	Beginning and ending times for night service, in an HH:MM format using a 24-hour clock. If the stop time is a smaller value than the start time, the stop time occurs on the day following the start time. For example, mon 19:00 07:00 means "from Monday at 7 p.m. until Tuesday at 7 a.m." The value 24:00 is not valid. If 00:00 is entered as a stop time, it is changed to 23:59. If 00:00 is entered for both start time and stop time, night service is in effect for the entire 24-hour period on the specified day.

Command Default

No recurring night-service time period is defined for every day.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

After you define recurring night-service time periods, use the **night-service bell** (ephone-dn) command to specify the extensions that will ring on other phones and the **night-service bell** (ephone) command to specify the phones on which the extensions will ring during the designated night-service periods.

Examples

The following example defines a night-service time period to be in effect every day from 7 p.m. to 8 a.m.:

```
Router(config)# telephony-service
Router(config-telephony)# night-service everyday 19:00 08:00
```

Related Commands

Command	Description
night-service bell (ephone)	Marks an IP phone to receive night-service bell notification when incoming calls are received on ephone-dns that are marked for night service during night-service time periods.
night-service bell (ephone-dn)	Marks an ephone-dn to send night-service bell notification to designated IP phones during night-service time periods.
night-service code	Defines a code to disable or reenable night service on IP phones.
night-service date	Defines a recurring time period associated with a month and day during which night service is active.
night-service day	Defines a recurring time period associated with a day of the week during which night service is active.
night-service weekday	Defines a recurring night-service time period to be in effect only on weekdays.
night-service weekend	Defines a recurring night-service time period to be in effect only on weekends.
telephony-service	Enters telephony-service configuration mode.

night-service weekday

To define a recurring night-service time period to be in effect on all weekdays, use the **night-service weekday** command in telephony-service configuration mode. To delete the defined time period, use the **no** form of this command.

night-service weekday *start-time stop-time*

no night-service weekday

Syntax Description

start-time stop-time	Beginning and ending times for night service, in an HH:MM format using a 24-hour clock. If the stop time is a smaller value than the start time, the stop time occurs on the day following the start time. For example, mon 19:00 07:00 means "from Monday at 7 p.m. until Tuesday at 7 a.m." The value 24:00 is not valid. If 00:00 is entered as a stop time, it is changed to 23:59. If 00:00 is entered for both start time and stop time, night service is in effect for the entire 24-hour period on the specified day.

Command Default

No recurring night-service time period is defined for weekdays.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Weekdays are defined as Monday, Tuesday, Wednesday, Thursday, and Friday.

After you define night-service periods, use the **night-service bell** (ephone-dn) command to specify the extensions that will ring on other phones and the **night-service bell** (ephone) command to specify the phones on which the extensions will ring during the designated night-service periods.

Examples

The following example defines a night-service time period every weekday from 5 p.m. to 9 a.m.:

```
Router(config)# telephony-service
Router(config-telephony)# night-service weekday 17:00 09:00
```

Related Commands

Command	Description
night-service bell (ephone)	Marks an IP phone to receive night-service bell notification when incoming calls are received on ephone-dns that are marked for night service during night-service time periods.
night-service bell (ephone-dn)	Marks an ephone-dn to send night-service bell notification to designated IP phones during night-service time periods.
night-service code	Defines a code to disable or reenable night service on IP phones.
night-service date	Defines a recurring time period associated with a month and day during which night service is active.
night-service day	Defines a recurring time period associated with a day of the week during which night service is active.
night-service everyday	Defines a recurring night-service time period to be in effect everyday.
night-service weekend	Defines a recurring night-service time period to be in effect only on weekends.
telephony-service	Enters telephony-service configuration mode.

night-service weekend

To define a recurring night-service time period to be active on weekends, use the **night-service weekend** command in telephony-service configuration mode. To delete the defined time period, use the **no** form of this command.

night-service weekend *start-time stop-time*

no night-service weekend

Syntax Description

start-time stop-time	Beginning and ending times for night service, in an HH:MM format using a 24-hour clock. If the stop time is a smaller value than the start time, the stop time occurs on the day following the start time. For example, mon 19:00 07:00 means "from Monday at 7 p.m. until Tuesday at 7 a.m." The value 24:00 is not valid. If 00:00 is entered as a stop time, it is changed to 23:59. If 00:00 is entered for both start time and stop time, night service is in effect for the entire 24-hour period on the specified day.

Command Default

No recurring night-service time period is defined for weekends.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Weekend is defined as Saturday and Sunday.

After you define night-service periods, use the **night-service bell** (ephone-dn) command to specify the extensions that will ring on other phones and the **night-service bell** (ephone) command to specify the phones on which the extensions will ring during the designated night-service periods.

Examples

The following example defines a night-service time period for all day Saturdays and Sundays:

```
Router(config)# telephony-service
Router(config-telephony)# night-service weekend 00:00 00:00
```

Related Commands

Command	Description
night-service bell (ephone)	Marks an IP phone to receive night-service bell notification when incoming calls are received on ephone-dns that are marked for night service during night-service time periods.
night-service bell (ephone-dn)	Marks an ephone-dn to send night-service bell notification to designated IP phones during night-service time periods.
night-service code	Defines a code to disable or reenable night service on IP phones.
night-service date	Defines a recurring time period associated with a month and day during which night service is active.
night-service day	Defines a recurring time period associated with a day of the week during which night service is active.
night-service everyday	Defines a recurring night-service time period to be in effect everyday.
night-service weekday	Defines a recurring night-service time period to be in effect only on weekdays.
telephony-service	Enters telephony-service configuration mode.

no-reg

To specify that the pilot number for a Cisco CallManager Express (Cisco CME) peer ephone hunt group not register with an H.323 gatekeeper, use the **no-reg** command in ephone-hunt configuration mode. To return to the default of the pilot number registering with an H.323 gatekeeper, use the **no** form of this command.

no-reg [**both** | **pilot**]

no no-reg [**both** | **pilot**]

Syntax Description

both	(Optional) Both the primary and secondary pilot numbers are not registered.
pilot	(Optional) Only the primary pilot number is not registered.

Defaults

If this command is not used, the pilot number registers with the H.323 gatekeeper. If this command is used but neither the **both** keyword nor the **pilot** keyword is used, only the secondary number is not registered.

Command Modes

Ephone-hunt configuration

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.2(15)ZJ	3.0	This command was introduced.
12.3(4)T	3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.3(7)T	3.1	The **both** and **pilot** keywords were introduced.

Usage Guidelines

This command is valid only for Cisco CME peer ephone hunt groups.

Examples

The following example defines peer ephone hunt group 2 with a primary and secondary pilot number, and specifies that the secondary pilot number should not register with the H.323 gatekeeper:

```
Router(config)# ephone-hunt 2 peer
Router(config-ephone-hunt)# pilot 2222 secondary 4444
Router(config-ephone-hunt)# no-reg
```

Related Commands

Command	Description
ephone-hunt	Defines an ephone hunt group and enters ephone-hunt configuration mode.
final	Defines the last ephone-dn in an ephone hunt group.

Command	Description
hops	Defines the number of times that a call is redirected to the next ephone-dn in a peer ephone-hunt-group list before proceeding to the final ephone-dn.
list	Defines the ephone-dns that participate in an ephone hunt group.
max-redirect	Changes the current number of allowable redirects in a Cisco CME system.
pilot	Defines the ephone-dn that is dialed to reach an ephone hunt group.
preference (ephone-hunt)	Sets preference order for the ephone-dn associated with an ephone-hunt-group pilot number.
timeout (ephone-hunt)	Sets the number of seconds after which a call that is not answered is redirected to the next number in the hunt-group list.

no-reg (voice register dn)

To specify that a voice DN for a SIP phone line in a Cisco CallManager Express (Cisco CME) system not register with an external proxy server, use the **no-reg** command in voice register dn configuration mode. To return to the default, use the **no** form of this command.

no-reg

no no-reg

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled

Command Modes

Voice register dn configuration

Command History

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

Usage Guidelines

Use this command to specify that a particular voice DN not register with the external proxy server. Configure the no-reg command per line. The default is to register all SIP lines in the Cisco CME system.

Examples

The following example shows how to configure bulk registration for registering a block of phone numbers starting with 408555 with an external registrar and specify that directory number 1, number 4085550100 not register with the external registrar:

```
Router(config)# voice register global
Router(voice-register-global)# mode cme
Router(voice-register-global)# bulk 408555….
Router(voice-register-global)# exit
Router(config)# voice register dn 1
Router(config-register-dn)# number 408550100
Router(config-register-dn)# no-reg
```

Related Commands

Command	Description
number (voice register dn)	Associates a telephone or extension number with a SIP phone in a Cisco CME system.
voice register dn	Enters voice register dn configuration mode to define an extension for a SIP phone line.

notify redirect (dial-peer)

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

notify redirect {**ip2ip** | **ip2pots**}

no notify redirect

Syntax Description

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

Command Default

Disabled

Command Modes

Dial-peer configuration

Command History

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

Usage Guidelines

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

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

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

Examples

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

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

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

Related Commands

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

Related Commands

notify redirect (voice-service)

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

notify redirect {**ip2ip** | **ip2pots**}

no notify redirect

Syntax Description

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

Command Default

Disabled

Command Modes

Voice-service configuration

Command History

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

Usage Guidelines

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

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

Examples

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

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

Related Commands

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

ntp-server

To specify the IP address of the Network Time Protocol (NTP) server used by SIP phones in a Cisco Unified CME system, use the **ntp-server** command in voice register global configuration mode. To remove the NTP server, use the **no** form of this command.

ntp-server *ip-address* [**mode** {**anycast** | **directedbroadcast** | **multicast** | **unicast**}]

no ntp-server

Syntax Description

ip-address	IP address of the NTP server.
mode	(Optional) Enables the broadcast mode for the server.
anycast	Enables anycast mode.
directedbroadcast	Enables directed broadcast mode.
multicast	Enables multicast mode.
unicast	Enables unicast mode.

Command Default

An NTP server is not used.

Command Modes

Voice register global configuration

Command History

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

Usage Guidelines

This command synchronizes all SIP phones to the specified NTP server.

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

Examples

The following example shows the mode for the NTP server set to multicast:

```
Router(config)# voice register global
Router(config-register-global)# ntp-server 10.10.10.1 mode multicast
```

Related Commands

Command	Description
create profile	Generates the configuration profile files required for SIP phones.
restart (voice register)	Performs a fast reset of one or all SIP phones associated with a Cisco Unified CME router.

number (ephone-dn)

To associate a telephone or extension number with an ephone-dn in a Cisco CallManager Express (Cisco CME) system, use the **number** command in ephone-dn configuration mode. To disassociate a number from an ephone-dn, use the **no** form of this command.

number *number* [**secondary** *number*] [**no-reg** [**both** | **primary**]]

no number

Syntax Description

number	String of up to 16 characters that represents an E.164 telephone number. Normally the string is composed of digits, but the string may contain alphabetic characters when the number is dialed only by the router, as with an intercom number. One or more periods (.) can be used as wildcard characters. For details, see "Usage Guidelines."
secondary	(Optional) Associates the number that follows as an additional number for this ephone-dn.
no-reg	(Optional) The E.164 numbers in the dial peer do not register with the gatekeeper. If you do not specify an option (**both** or **primary**) after the **no-reg** keyword, only the secondary number is not registered.
both	(Optional) Both primary and secondary numbers are not registered.
primary	(Optional) Primary number is not registered.

Defaults

No primary or secondary phone number is associated with the ephone-dn.

Command Modes

Ephone-dn configuration

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.1(5)YD	1.0	This command was introduced on the following platforms: Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(2)XT	2.0	This command was implemented on the Cisco 1750 and Cisco 1751.
12.2(8)T	2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	2.0	This command was implemented on the Cisco 2600XM and Cisco 2691.
12.2(11)T	2.01	This command was implemented on the Cisco 1760.
12.2(15)ZJ	3.0	The ability to use alphabetic characters as part of the number string was introduced.
12.3(4)T	3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Usage Guidelines

This command defines a valid number for an ephone-dn (extension) that is to be assigned to an IP phone. The **secondary** keyword allows you to associate a second telephone number with an ephone-dn so that it can be called by dialing either the main or secondary phone number. The secondary number may contain wildcards; for example, 50.. (the number 50 followed by periods, which stand for wildcards).

The **no-reg** keyword causes an E.164 number in the dial peer not to register with the gatekeeper. If you do not specify **both** or **primary** after the **no-reg** keyword, only the secondary number does not register.

A number normally contains only numeric characters, which allow it to be dialed from any telephone keypad. However, in certain cases, such as intercom numbers, which are normally dialed only by the router, you can insert alphabetic characters into the number to prevent phone users from dialing it and using the intercom function without authorization.

A number can also contain one or more periods (.) as wildcard characters that will match any dialed number in that position. For example, 51.. rings when 5100 is dialed, when 5101 is dialed, and so forth.

After you use the **number** command, assign the ephone-dn to an ephone using the **button** command. Following the use of the **button** command, the **restart** command must be used to initiate a quick reboot of the phone to which this number is assigned.

Examples

The following example sets 5001 as the primary extension number for a Cisco IP phone and 0 as the secondary number. This configuration allows the telephone number 5001 to act as a regular extension number and also to act as the operator line such that callers who dial 0 are routed to the phone line with extension number 5001.

```
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 5001 secondary 0
```

The following example sets 5001 as the primary extension number for a Cisco IP phone and "500." (the number 500 followed by a period) as the secondary number. This configuration allows any calls to extension numbers from the range 5000 to 5009 to be routed to extension 5001 if the actual extension number dialed cannot be found. For example, IP phones may be active in the system with lines that correspond to 5001, 5002, 5004, 5005, and 5009. A call to 5003 would be unable to locate a phone with extension 5003, so the call would be routed to extension 5001.

```
Router(config-ephone-dn)# number 5001 secondary 500.
```

The following example defines a pair of intercom ephone-dns that are programmed to call each other. The intercom numbers contain alphabetic characters to prevent anyone from dialing them from another phone. Ephone-dn 19 is assigned the number A5511 and is programmed to dial A5522, which belongs to ephone-dn 20. Ephone-dn 20 is programmed to dial A5511. No one else can dial these numbers.

```
Router(config)# ephone-dn 19
Router(config-ephone-dn)# number A5511
Router(config-ephone-dn)# name Intercom
Router(config-ephone-dn)# intercom A5522
Router(config-ephone-dn)# exit
Router(config)# ephone-dn 20
Router(config-ephone-dn)# number A5522
Router(config-ephone-dn)# name Intercom
Router(config-ephone-dn)# intercom A5511
```

Related Commands

Command	Description
button	Associates ephone-dns with individual buttons on Cisco IP phones and specifies ring behavior per button.
ephone-dn	Enters ephone-dn configuration mode.
intercom	Creates an intercom by programming a pair of extensions (ephone-dns) to automatically call and answer each other.
name	Configures a username associated with a directory number.
preference	Sets preference for the attached dial peer for a directory number.
restart (ephone)	Performs a fast reboot of a single phone associated with a Cisco CME router.
restart (telephony-service)	Performs a fast reboot of one or all phones associated with a Cisco CME router.

number (voice register dn)

To associate a telephone or extension number with a SIP phone in a Cisco CallManager Express (Cisco CME) system, use the **number** command in voice register dn configuration mode. To disassociate a number, use the **no** form of this command.

number *number*

no number

Syntax Description

number	String of up to 16 characters that represents an E.164 telephone number. Normally the string is composed of digits, but the string may contain alphabetic characters when the number is dialed only by the router, as with an intercom number.

Defaults

No default behavior or values

Command Modes

Voice register dn configuration

Command History

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

Usage Guidelines

This command defines a valid number for an extension that is to be assigned to a SIP phone. Use this command before using the other commands in voice register dn configuration mode.

A number normally contains only numeric characters which allows users to dial the number from any telephone keypad. However, in certain cases, such as the numbers for intercom extensions, you want to use numbers that can only be dialed internally from the Cisco CallManager Express router and not from telephone keypads.

The **number** command allows you to assign alphabetic characters to the number so that the extension can be dialed by the router for intercom calls but not by unauthorized individuals from other phones.

After you use the **number** command, use the **reset** command to initiate a quick reboot of the phone to which this number is assigned.

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

Examples

The following example shows how to set 5001 as the extension number for directory number 1 on a SIP phone.

```
Router(config)# voice register dn 1
Router(config-register-dn)# number 5001
```

Related Commands

Command	Description
reset (voice register global)	Performs a complete reboot of all SIP phones associated with a Cisco CME router.
reset (voice register pool)	Performs a complete reboot of a single SIP phone associated with a Cisco CME router.
voice register dn	Enters voice register dn configuration mode to define an extension for a SIP phone line.

number (voice register pool)

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

number *tag* {*number-pattern* [**preference** *value*] [**huntstop**] | **dn** *dn-tag*}

no number *tag*

Syntax Description

tag	Identifies the telephone number when there are multiple **number** commands. Range is 1 to 10.
number-pattern	Phone numbers (including wild cards and patterns) that are permitted by the registrar to handle the Register message from the Cisco SIP IP phone.
preference *value*	(Optional) Defines the number list preference order. Range is 0 to 10. The highest preference is 0. There is no default.
huntstop	(Optional) Stops hunting if the dial peer is busy.
dn *dn-tag*	Identifies the directory number tag for this phone number as defined by the **voice register dn** command. Range is 1 to 150.

Command Default

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

Command Modes

Voice register pool configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco SIP SRST 3.0	This command was introduced.
12.3(4)T	Cisco SIP SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.4(4)T	Cisco CME 3.4 Cisco SIP SRST 3.4	This command was added to Cisco CME and the **dn** keyword was added.
12.4(11)XJ	Cisco Unified CME 4.1 Cisco Unified SRST 4.1	The *number-pattern* argument, **preference** keyword, and **huntstop** keyword were removed from Cisco Unified CME.
12.4(15)T	Cisco Unified CME 4.1 Cisco Unified SRST 4.1	The modifications to this command were integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

The **number** command indicates the phone numbers that are permitted by the registrar to handle the Register message from the SIP phone. The keywords and arguments of this command allow for more explicit setting of user preferences regarding what number patterns should match the voice register pool.

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

Examples

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

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

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

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

Related Commands

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

number (voice user-profile and voice logout-profile)

To create line definitions in a voice-user profile or voice-logout profile to be downloaded to a Cisco Unified IP phone that is enabled for extension mobility, use the **number** command in voice user-profile or voice logout-profile configuration mode. To remove line definition from a profile, use the no form of this command.

number *number*[**,**...*number*] **type** *type*

no number *number*[**,**...*number*] **type** *type*

Syntax Description

number	String of up to 16 characters that represents an E.164 telephone number to be associated with and displayed next to a line button on an IP phone. This directory number must be already configured by using the **number** command in ephone-dn or voice register dn configuration mode.
[**,**...*number*]	(Optional) For overlay lines only, with or without call waiting. Directory number that should roll over to this line. This directory number must be already configured by using the **number** command in ephone-dn or voice register dn configuration mode.
type	Characteristics to be associated with this line button.
type	Word that describes characteristics to be associated with the line button being configured. Valid entries are as follows: • beep-ring • feature-ring • monitor-ring • silent-ring • overlay • cw-overlay

Command Default

No line definition is created.

Command Modes

Voice logout-profile configuration (config-logout-profile)
Voice user-profile configuration (config-user-profile)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XW1	Cisco Unified CME 4.2	This command was introduced.

Usage Guidelines

Use this command in voice user-profile configuration mode to create a line button definition in a user profile to be downloaded to the IP phone when the user is logged into an IP phone that is enabled for extension mobility.

Use this command in voice logout-profile configuration mode to create a line button definition in a default profile to be downloaded to an IP phone when no user is logged into an IP phone that is enabled for extension mobility.

For button appearance, extension mobility will associate line definitions in the voice-logout profile or voice-user profile to phone buttons in a sequential manner. If the profile contains more directory and speed-dial numbers than there are buttons on the physical phone to which the profile is downloaded, the remaining numbers in the profile are ignored.

On Cisco Unified IP phones, line definitions are assigned to available extension buttons before speed-dial definitions and sequentially, starting with the lowest directory number first.

After creating or modifying a profile, use the **reset** (voice logout-profile or voice user-profile) command to reset all phones associated with the profile being configured to propagate the changes.

Type **?** to list valid options for the **type** keyword. The following options are valid at the time that this document was written:

- **beep-ring**

  Beep but no ring. Audible ring is suppressed for incoming calls but call-waiting beeps are allowed. Visible cues are the same as those for a normal ring.

- **feature-ring**

  Differentiates incoming calls on a special line from incoming calls on other lines on the phone. The feature-ring cadence is a triple pulse, as opposed to a single-pulse ring for normal internal calls and a double-pulse ring for normal external calls.

- **monitor-ring**

  A line button that is configured for monitor mode on one phone provides visual line status for a line that also appears on another phone. When monitor mode is set for a button with a shared line, the line status indicates that the shared line is either idle or in use. The line and line button are available in monitor mode for visual status only. Calls cannot be made or received using a line button that has been set in monitor mode. Incoming calls on a line button that is in monitor mode do not ring and do not display caller ID or call-waiting caller ID. Monitor mode is intended to be used only in the context of shared lines so that one user, such as a receptionist, can visually monitor the in-use status of several users' phone extensions (for example, as a busy-lamp field).

  The line button for a monitored line can be used as a direct-station-select for a call transfer when the monitored line is in an idle state. In this case, the phone user who transfers a call from a normal line can press the Transfer button and then press the line button of the monitored line, causing the call to be transferred to the phone number of the monitored line.

- **silent-ring**

  You can configure silent ring on any type of phone. However, you typically set silent ring only on buttons of a phone with multiple lines, such as a Cisco Unified IP Phone 7940 or Cisco Unified IP Phone 7960 and 7960G. The only visible cue is a flashing **((<** icon in the phone display.

  If you configure a button to have a silent ring, you will not hear a call-waiting beep or call-waiting ring regardless of whether the ephone-dn associated with the button is configured to generate a call-waiting beep or call-waiting ring.

Note In Cisco IOS Release 12.4(4)XC and later releases, the silent ringing behavior is overridden during active night-service periods. Silent ringing does not apply during designated night-service periods when the s keyword is used.

- **overlay**

Overlay lines are directory numbers that share a single line button on a multibutton phone. When more than one incoming call arrives on lines that are set on a single button, the line (ephone-dn) that is the left most in the **number** command list is the primary line and is given the highest priority. If this call is answered by another phone or if the caller hangs up, the phone selects the next line in its overlay set to present as the ringing call. The caller ID display updates to show the caller ID for the currently presented call.

Directory numbers that are part of an overlay set can be single-line directory numbers or dual-line directory numbers, but the set must contain either all single-line or all dual-line directory numbers, and not a mixture of the two.

The primary directory number on each phone in a shared-line overlay set should be an ephone-dn that is unique to the phone to guarantee that the phone will have a line available for outgoing calls, and to ensure that the phone user can obtain dial-tone even when there are no idle lines available in the rest of the shared-line overlay set. Use a unique directory number in this manner to provide for a unique calling party identity on outbound calls made by the phone so that the called user can see which specific phone is calling.

The name of the first directory number in the overlay set is not displayed because it is the default directory number for calls to the phone, and the name or number is permanently displayed next to the phone's button. For example, if there are ten numbers in an overlay set, only the last nine numbers are displayed when calls are made to them.

- **cw-overlay**

  The configuration for the overlaid lines with call waiting and without call waiting is the same.

  Directory numbers can accept call interruptions, such as call waiting, by default. For call waiting to work, the default must be active. To ensure that this is the case, remove the no call-waiting beep accept command from the configurations of directory numbers for which you want to use call waiting.

  Directory numbers that are part of an cw-overlay set can be single-line directory numbers or dual-line directory numbers, but the set must contain either all single-line or all dual-line directory numbers, and not a mixture of the two.

  The Cisco Unified IP Phone 7931G cannot support overlays that contain directory numbers that are configured for dual-line mode.

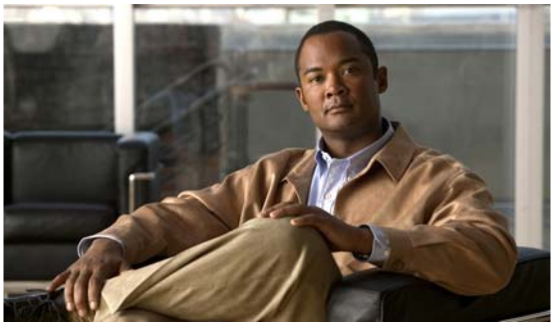
Examples

The following example shows the configuration for a voice-user profile to be downloaded when the a phone user logs into a Cisco Unified IP phone that is enabled for extension mobility. Which lines and speed-dial buttons in this profile are configured on an IP phone after the user logs in depends on phone type. For example, if the user logs into a Cisco Unified IP Phone 7970, all buttons are configured according to voice-user profile1. However, if the phone user logs into a Cisco Unified IP Phone 7960, all six lines are mapped to phone buttons and the speed dial is ignored because there is no button available for speed dial.

```
pin 12345
user me password pass123
number 2001 type silent-ring
number 2002 type beep-ring
number 2003 type feature-ring
number 2004 type monitor-ring
number 2005,2006 type overlay
number 2007,2008 type cw-overly
speed-dial 1 3001
speed-dial 2 3002 blf
```

Related Commands

Command	Description
logout-profile	Enables Cisco Unified IP phone for extension mobility and assigns a logout profile to this phone.
reset (voice logout-profile and voice user-profile)	Performs a complete reboot of all IP phones to which a particular logout-profile or user-profile is downloaded.



Cisco Unified CME Commands: P

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This chapter contains commands to configure and maintain Cisco Unified Communications Manager Express (formally known as Cisco Unified CallManager Express). The commands are presented in alphabetical order. Some commands required for configuring Cisco Unified Communications Manager Express (Cisco Unified CME) may be found in other Cisco IOS command references. Use the command reference master index or search online to find these commands.

paging

To define an extension (ephone-dn) as a paging extension that can be called to broadcast an audio page to a set of Cisco IP phones, use the **paging** command in ephone-dn configuration mode. To disable this feature, use the **no** form of this command.

paging [**ip** *multicast-address* **port** *udp-port-number*]

no paging [**ip**]

Syntax Description

ip *multicast-address*	(Optional) Uses an IP multicast address to multicast voice packets for audio paging; for example, 239.0.1.1. Note that IP phones do not support multicast at 224.x.x.x addresses. Default is that multicast is not used and IP phones are paged individually using IP unicast transmission (up to ten phones).
port *udp-port-number*	(Optional) Uses this UDP port for the multicast. Range is from 2000 to 65535. Default is 2000.

Defaults

No paging number is established.

Command Modes

Ephone-dn configuration

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.2(2)XT	2.0	This command was introduced on the following platforms: Cisco 1750, Cisco 1751, Cisco 2600 series, Cisco 3600 series, Cisco IAD2420 series.
12.2(8)T	2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	2.0	This command was implemented on the Cisco 2600XM and Cisco 2691.
12.2(11)T	2.01	This command was implemented on the Cisco 1760.

Usage Guidelines

To configure a set of phones to receive an audio page, follow these steps:

1. Use the **paging** command in ephone-dn configuration mode to define a number that people can dial to send a page. The following example defines a paging-dn tag (21) and extension number (34455) to dial to send a page.

```
ephone-dn 21
 paging
 number 34455
```

2. Use the **paging-dn** command in ephone configuration mode to assign the same paging-dn tag that you defined in Step 1 to all the phones that you want to receive the page. This set of phones is called a "paging set." You can have more than one paging set in a Cisco CME system. The following example assigns the paging-dn tag from Step 1 (21) to two phones (3 and 4) so that they will receive audio pages.

```
ephone 3
 paging-dn 21

ephone 4
 paging-dn 21
```

The **paging** command configures an ephone-dn as an extension that people can dial to broadcast audio pages to a specified set of idle Cisco IP phones. The extension associated with this command does not appear on any ephone; it is a "dummy" extension. The dn-tag associated with this extension becomes the paging-dn tag for this paging set.

When a person dials the number assigned to the dummy extension and speaks into the phone, the audio stream is sent as a page to the paging set (the set of all phones that have been configured with this paging-dn tag as an argument to the **paging-dn** command). Idle phones in the paging set automatically answer the paging call in one-way speakerphone mode. Paging sets can be joined into a single combined paging group with the **paging group** command.

The optional **ip** keyword and *multicast-address* argument define a paging multicast address for this paging set. If an IP multicast address is not configured, IP phones are paged individually using IP unicast transmission (to a maximum of ten IP phones). The recommended operation is with an IP multicast address. When multiple paging-dn tags are configured using the **paging** command, each paging-dn tag should use a unique IP multicast address.

Note IP phones do not support multicast at 224.x.x.x addresses.

Each ephone-dn and paging-dn tag that is used for paging can support a maximum of ten distinct targets (IP addresses and interfaces). A multicast address counts as a single target for each physical interface in use (regardless of the number of phones connected via the interface). Paging using a single IP multicast address that requires output on three different Ethernet interfaces represents use of three counts out of the maximum ten. Each unicast target counts as a single target, such that paging that does not use multicast at all is limited to paging ten phones. For example, ten IP phones paged through multicast on Fast Ethernet interface 0/1.1 plus five IP phones paged through multicast on Fast Ethernet interface 0/1.2 are counted as two targets.

For simultaneous paging to more than one paging ephone-dn, Cisco recommends that you use different IP multicast addresses (not just different port numbers) for paging configuration.

Examples

The following example creates a paging extension number that uses IP multicast paging:

```
Router(config)# ephone-dn 20
Router(config-ephone-dn) number 2000
Router(config-ephone-dn) paging ip 239.0.1.1 port 2000
```

A more complete configuration example follows, in which paging sets 20 and 21 are created. Pages to extension 2000 are multicast to ephones 1 and 2. Pages to extension 2001 are multicast to ephones 3 and 4.

```
ephone-dn 1
 number 2345
```

```
ephone-dn 2
 number 2346

ephone-dn 3
 number 2347

ephone-dn 4
 number 2348

ephone-dn 20
 number 2000
 paging ip 239.0.1.20 port 2000

ephone-dn 21
 number 2001
 paging ip 239.0.1.21 port 2000

ephone 1
 button 1:1
 paging-dn 20

ephone 2
 button 1:2
 paging-dn 20

ephone 3
 button 1:3
 paging-dn 21

ephone 4
 button 1:4
 paging-dn 21
```

Related Commands

Command	Description
ephone-dn	Enters ephone-dn configuration mode.
paging-dn	Assigns audio paging reception capability to a Cisco IP phone.
paging group	Combines two or more paging sets into a combined paging group.

paging group

To create a combined paging group from two or more previously established paging sets, use the **paging group** command in ephone-dn configuration mode. To remove a paging group, use the **no** form of this command.

paging group *paging-dn-tag*, *paging-dn-tag...*

no paging group

Syntax Description

paging-dn-tag	Comma-separated list of paging-dn-tags (unique sequence numbers associated with paging ephone-dns) that have previously been associated with the paging extension of a paging set using the **paging-dn** command. You can include up to ten paging-dn-tags separated by commas; for example, 4, 6, 7, 8.

Defaults

Paging is disabled on all Cisco IP phones.

Command Modes

Ephone-dn configuration

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.2(2)XT	2.0	This command was introduced on the following platforms: Cisco 1750, Cisco 1751, Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(8)T	2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	2.0	This command was implemented on the Cisco 2600XM and Cisco 2691.
12.2(11)T	2.01	This command was implemented on the Cisco 1760.

Usage Guidelines

Use this command to combine previously defined sets of phones associated with individual paging extensions (ephone-dns) into a combined group to enable a single page to be sent to large numbers of phones at once. To remove a paging group, use the **no** form of the command. All paging-dn tags included in the list must have already been defined as paging-dns using the **paging** command.

The use of paging groups allows phones to participate in a small local paging set (for example, paging to four phones in a company's shipping and receiving department) but also supports company-wide paging when needed (for example, by combining the paging sets for shipping and receiving with the paging sets for accounting, customer support, and sales into a single paging group).

Examples

In the following example, paging sets 20 and 21 are defined and then combined into paging group 22. Paging set 20 has a paging extension of 2000. When someone dials extension 2000 to deliver a page, the page is sent to Cisco IP phones (ephones) 1 and 2. Paging set 21 has a paging extension of 2001. When someone dials extension 2001 to deliver a page, the page is sent to ephones 3 and 4. Paging group 22 combines sets 20 and 21, and when someone dials its paging extension, 2002, the page is sent to all the phones in both sets and to ephone 5, which is directly subscribed to the combined paging group.

```
ephone-dn 20
 number 2000
 paging ip 239.0.1.20 port 2000

ephone-dn 21
 number 2001
 paging ip 239.0.1.21 port 2000

ephone-dn 22
 number 2002
 paging ip 239.0.2.22 port 2000
 paging group 20,21

ephone 1
 button 1:1
 paging-dn 20

ephone 2
 button 1:2
 paging-dn 20

ephone 3
 button 1:3
 paging-dn 21

ephone 4
 button 1:4
 paging-dn 21

ephone 5
 button 1:5
 paging-dn 22
```

Related Commands

Command	Description
ephone-dn	Enters ephone-dn configuration mode.
paging	Creates a paging extension (ephone-dn) that can be called in order to broadcast an audio page to a group of Cisco IP phones.
paging-dn	Assigns a paging extension (paging-dn) to a Cisco IP phone.

paging-dn

To create a paging extension (paging-dn) to receive audio pages on a Cisco Unified IP phone in a Cisco Unified CME system, use the **paging-dn** command in ephone or ephone-template configuration mode. To disable this feature, use the **no** form of this command.

paging-dn *paging-dn-tag* {**multicast** | **unicast**}

no paging-dn

Syntax Description

paging-dn-tag	Dn-tag of an ephone-dn that was designated as a paging ephone-dn with the **paging** command.
multicast	Uses multicast if available. By default, audio paging is transmitted to the Cisco Unified IP phone using multicast.
unicast	Forces unicast paging for this phone. This keyword indicates that the Cisco Unified IP phone is not capable of receiving audio paging through multicast and requests that all pages to this phone be sent through unicast. The maximum number of phones supported through unicast is ten.

Command Default

Paging is disabled on all Cisco Unified IP phones.

Command Modes

Ephone configuration
Ephone-template configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(2)XT	Cisco ITS 2.0	This command was introduced on the following platforms: Cisco 1750, Cisco 1751, Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	Cisco ITS 2.0	This command was implemented on the Cisco 2600XM and Cisco 2691.
12.2(11)T	Cisco ITS 2.01	This command was implemented on the Cisco 1760.
12.4(4)XC	Cisco Unified CME 4.0	This command was made available in ephone-template configuration mode.
12.4(9)T	Cisco Unified CME 4.0	This command in ephone-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

To configure a set of phones to receive an audio page, follow these steps:

1. Use the **paging** command in ephone-dn configuration mode to define a number that people can dial to send a page. The following example defines a paging-dn tag (21) and extension number (34455) to dial to send a page.

```
ephone-dn 21
 paging
 number 34455
```

2. Use the **paging-dn** command in ephone configuration mode to assign the same paging-dn tag that you defined in Step 1 to all the phones that you want to receive the page. This set of phones is called a "paging set." You can have more than one paging set in a Cisco Unified CME system. The following example assigns the paging-dn tag from Step 1 (21) to two phones (3 and 4) so that they will receive audio pages.

```
ephone 3
 paging-dn 21

ephone 4
 paging-dn 21
```

This command creates a paging extension (paging-dn) associated with an IP phone. Each phone can support only one paging-dn extension. This extension does not occupy a phone button and is therefore not configured on the phone with the **button** command. The paging-dn allows the phone to automatically answer audio pages in one-way speakerphone mode. There is no press-to-answer option as there is with an intercom extension.

The *paging-dn-tag* argument in this command takes the value of the dn-tag of an extension (ephone-dn) that has been made a paging ephone-dn using the **paging** command. This is the extension that callers dial to deliver an audio page. All of the phones that are going to receive the same audio pages are configured with the same *paging-dn-tag*. These phones form a paging set.

An IP phone can belong to only one paging set, but any number of phones can belong to the same paging set using multicast. There can be any number of paging sets in a Cisco Unified CME system, and paging sets can be joined to create a combined paging group using the **paging group** command. For example, you may create separate paging sets for each department (sales, support, shipping) and combine them into a single combined paging group (all departments). Only single-level grouping is supported (no support for groups of groups).

Normal phone calls that are received while an audio page is in progress interrupt the page.

The paging mechanism supports audio distribution using IP multicast, replicated unicast, and a mixture of both (so that multicast is used where possible, and unicast is used with specific phones that cannot be reached through multicast).

Note For unicast paging to all phones, omit the IP multicast address in the ephone-dn configuration. For unicast paging to a specific phone using an ephone-dn configured for multicast, add the **unicast** keyword as part of the **paging-dn** command in ephone configuration mode.

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

Examples

The following example creates paging number 5001 on ephone-dn 22 and adds ephone 4 as a member of the paging set. Multicast is set for the paging-dn. Note that IP phones do not support multicast at 224.x.x.x addresses.

```
ephone-dn 1
 number 5123

ephone-dn 22
 name Paging Shipping
 number 5001
 paging ip 239.1.1.10 port 2000

ephone 4
 mac-address 0030.94c3.8724
 button 1:1
 paging-dn 22 multicast
```

Related Commands

Command	Description
ephone-dn	Enters ephone-dn configuration mode.
number	Configures a valid number for the Cisco Unified IP phone.
paging	Creates a paging extension (ephone-dn) that can be called in order to broadcast an audio page to a group of Cisco Unified IP phones.
paging group	Combines two or more paging sets into a combined paging group.

param aa-hunt

To declare a Cisco Unified CME B-ACD menu number and associate it with the pilot number of an ephone hunt group, use the **param aa-hunt command in** application-parameter configuration mode. To remove the menu number and the ephone hunt group pilot number, use the **no** form of this command.

param aa-hunt*menu-number pilot-number*

no param aa-hunt*menu-number pilot-number*

Syntax Description

menu-number	Number that callers must dial to reach the pilot number of an ephone hunt group. The range is from 1 to 10. The default is 1.
pilot-number	Ephone hunt group pilot number.

Defaults

Menu number 1 is configured, but it is not associated with a pilot number**.**

Command Modes

Application-parameter configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco CME 3.3	This command was introduced to replace the **call application voice aa-hunt** command.

Usage Guidelines

This command is used with Cisco Unified CME Basic Automatic Call Distribution and Auto-Attendant (B-ACD) service. It is configured under the **service** command for the call-queue script.

Up to ten aa-hunt menu options, or hunt groups, are allowed per call-queue service. You can use any of the allowable numbers in any order.

This command associates a menu option with the pilot number of an ephone hunt group. When a caller presses the digit of a menu option that has been associated with an ephone hunt group using this command, the call is routed to the pilot number of the hunt group.

Menu options for B-ACD services can be set up in many ways. For more information, see the *Cisco Unified CallManager Express B-ACD and Tcl Call-Handling Applications* document for your release.

The highest aa-hunt number that you establish using this command also automatically maps to zero (0) and can therefore be used to represent operator services to your callers. In the following example, callers can dial either 8 or 0 to reach aa-hunt8, the hunt group with the pilot number 8888.

```
application
 service queue flash:
  param aa-hunt1 1111
  param aa-hunt3 3333
  param aa-hunt8 8888
```

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

Examples

The following example configures a call-queue service called queue to associate three menu numbers with three pilot numbers of three ephone hunt groups:

- Pilot number 1111 for ephone hunt group 1 (sales)
- Pilot number 2222 for ephone hunt group 2 (customer service)
- Pilot number 3333 for ephone hunt group 3 (operator)

If a caller presses 2 for customer service, the call is transferred to 2222 and then is sent to the next available ephone-dn from the group of ephone-dns assigned to ephone hunt group 1: 2001, 2002, 2003, 2004, 2005, and 2006. The sequencing of ephone-dns within a hunt group is handled by the ephone hunt group itself, not by the B-ACD service. (Note that the configuration for ephone hunt groups used with Cisco Unified CME B-ACD services do not use the **final** command.)

```
ephone-hunt 1 peer
 pilot 1111
 list 1001, 1002, 1003, 1004, 1005, 1006, 1007, 1008, 1009, 1010

ephone-hunt 2 peer
 pilot 2222
 list 2001, 2002, 2003, 2004, 2005, 2006

ephone-hunt 3 peer
 pilot 3333
 list 3001, 3002, 3003, 3004

application
 service queue flash:
  param aa-hunt1 1111
  param aa-hunt2 2222
  param aa-hunt3 3333
.
.
.
```

Related Commands

Command	Description
application	Enters application configuration mode.
service	Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.

param aa-pilot

To assign a pilot number to a Cisco Unified CME B-ACD automated attendant (AA) service, use the **param aa-pilot command in** application-parameter configuration mode. To remove the AA pilot number, use the **no** form of this command.

param aa-pilot *aa-pilot-number*

no param aa-pilot *aa-pilot-number*

Syntax Description

aa-pilot-number	Telephone number that callers dial in order to reach this AA service.

Defaults

Cisco Unified CME B-ACD menu number 1 is configured, but it has no pilot number**.**

Command Modes

Application-parameter configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(14)T	Cisco CME 3.3	This command was introduced to replace the **call application voice aa-pilot** command.

Usage Guidelines

This command is used with the Cisco Unified CallManager Express (Cisco Unified CME) Basic Automatic Call Distribution and Auto-Attendant (B-ACD) service. Each AA has one AA pilot number, although there may be more than one AA used with a B-ACD service.

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

For more information, see the *Cisco Unified CallManager Express B-ACD and Tcl Call-Handling Applications* document for your release.

Examples

The following example sets up a B-ACD with two AAs, both in drop-through mode. The first AA is called acdaa and it has an AA pilot number of (800) 555-0121. The second AA is aa-bcd and has an AA pilot number of (800) 555-0123. Both AAs use the call-queue service named callq. Incoming POTS dial peers are established for both AA pilot numbers.

```
dial-peer voice 1010 pots
 service acdaa
 port 1/1/0
 incoming called-number 8005550121

dial-peer voice 1020 pots
 service aa-bcd
 port 1/1/1
 incoming called-number 8005550123
.
.
.
application
```

```
 service callq tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd.tcl
  param queue-manager-debugs 1
  param aa-hunt1 5071
  param aa-hunt2 5072
  param number-of-hunt-grps 2
  param queue-len 10
!
 service acdaa tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
  paramspace english location tftp://192.168.254.254/user1/prompts/
  paramspace english index 0
  paramspace english language en
  param aa-pilot 8005550121
  param service-name callq
  param max-time-vm-retry 2
  param voice-mail 5007
  param call-retry-timer 20
  param number-of-hunt-grps 1
  param drop-through-prompt _bacd_welcome.au
  param drop-through-option 2
  param second-greeting-time 45
  param handoff-string acdaa
  param max-time-call-retry 360
!
 service aa-bcd tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
  paramspace english location tftp://192.168.254.254/user1/prompts/
  paramspace english index 0
  paramspace english language en
  param aa-pilot 8005550123
  param service-name callq
  param second-greeting-time 60
  param max-time-call-retry 180
  param max-time-vm-retry 2
  param voice-mail 5007
  param call-retry-timer 5
  param handoff-string aa-bcd
  param drop-through-option 1
  param number-of-hunt-grps 1
```

Related Commands

Command	Description
application	Enters application configuration mode.
service	Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.

param call-retry-timer

To specify the time interval before each attempt to retry to connect a call to an ephone hunt group used with a Cisco CME B-ACD service, use the **param call-retry-timer command in** application-parameter configuration mode. To return to the default, use the **no** form of this command.

param call-retry-timer *seconds*

no param call-retry-timer *seconds*

Syntax Description

seconds	Time that a call must wait before attempting or reattempting to transfer a call to an ephone hunt group pilot number, in seconds. Range is from 5 to 30 seconds.

Defaults

15 seconds

Command Modes

Application-parameter configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(14)T	Cisco CME 3.3	This command was introduced to replace the **call application voice call-retry-timer** command.

Usage Guidelines

This command is used with Cisco Unified CME Basic Automatic Call Distribution and Auto-Attendant (B-ACD) service. This command is configured under the **service** command for an AA service. A Cisco Unified CME B-ACD service can have more than one AA, and each AA can specify a different interval for retries to connect to ephone hunt group phones.

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

For more information, see the *Cisco Unified CallManager Express B-ACD and Tcl Call-Handling Applications* document for your release.

Examples

The following example sets up a B-ACD with two AAs. The first AA is called acdaa and it has an AA pilot number of (800) 555-0121. The second AA is aa-bcd and has an AA pilot number of (800) 555-0123. Both AAs use the call-queue service named callq. The first AA has a call-retry timer set to 10 seconds, and the second AA has a call-retry timer set to 5 seconds.

```
dial-peer voice 1010 pots
 service acdaa
 port 1/1/0
 incoming called-number 8005550121

dial-peer voice 1020 pots
 service aa-bcd
 port 1/1/1
 incoming called-number 8005550123
```

```
.
.
.
application
 service callq tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd.tcl
  param queue-manager-debugs 1
  param aa-hunt1 5071
  param aa-hunt2 5072
  param number-of-hunt-grps 2
  param queue-len 10
!
 service acdaa tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
  paramspace english location tftp://192.168.254.254/user1/prompts/
  paramspace english index 0
  paramspace english language en
  param aa-pilot 8005550121
  param service-name callq
  param max-time-vm-retry 2
  param voice-mail 5007
  param call-retry-timer 10
  param number-of-hunt-grps 1
  param drop-through-prompt _bacd_welcome.au
  param drop-through-option 2
  param second-greeting-time 45
  param handoff-string acdaa
  param max-time-call-retry 60
!
 service aa-bcd tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
  paramspace english location tftp://192.168.254.254/user1/prompts/
  paramspace english index 0
  paramspace english language en
  param aa-pilot 8005550123
  param service-name callq
  param second-greeting-time 60
  param max-time-call-retry 180
  param max-time-vm-retry 2
  param voice-mail 5007
  param call-retry-timer 5
  param handoff-string aa-bcd
  param drop-through-option 1
  param number-of-hunt-grps 1
```

Related Commands

Command	Description
application	Enters application configuration mode.
service	Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.

param co-did-max

To set the upper boundary of the range of valid digits coming from the PSTN Central Office (CO) for use with the Direct Inward Dial (DID) Digit Translation Service, use the **param co-did-max** command in application-parameter configuration mode. To disable this option, use the **no** form of this command.

param co-did-max *max-co-value*

no param co-did-max *max-co-value*

Syntax Description

max-co-value	Maximum value of digits coming from the CO. The digit string can be any length, but the string length must be the same in the **param co-did-min**, **param co-did-max**, **param store-did-min**, and **param store-did-max** commands.

Defaults

No maximum value is defined for the range of digits coming from the CO.

Command Modes

Application-parameter configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced to replace the **call application voice co-did-max** command.
12.4(9)T	Cisco Unified CME 4.0	This command replaced the **call application voice co-did-max** command and was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command defines the upper limit of the range of digits accepted from the CO when it is used with the Cisco Unified CallManager Express (Cisco Unified CME) DID Digit Translation Service. This service provides number translation for DID calls when the range of DID digits provided by the PSTN Central Office (CO) does not match the digits in the Cisco Unified CME extension numbers.

The Tcl script that provides the service accepts PSTN DID numbers of any length and maps them to the internal extension numbers that are assigned by a system administrator. Where necessary, a prefix is appended to the DID digits to create a valid extension number. The script uses the parameters that you input to determine the valid range of digits to be accepted from the CO, the valid range of digits in the local dial plan, and the prefix to append, if necessary. The script also handles DID calls that map to invalid extension numbers: a prompt is played and the calls are disconnected.

Examples

The following example configures DID Digit Translation Service on the Cisco Unified CME router. It sets a lower boundary of 00 and an upper boundary of 79 for the valid range of digits coming from the CO.

```
application
 service didapp tftp://192.168.254.254/scripts/did/app-THD-DID-2.0.0.1.tcl
```

```
paramspace english index 1
paramspace english language en
paramspace english location tftp://192.168.254.254/apps/dir25/
param secondary-prefix 4
param did-prefix 5
param co-did-min 00
param co-did-max 79
param store-did-min 00
param store-did-max 79
```

Related Commands

Command	Description
application	Enters application configuration mode.
service	Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.
param co-did-min	Sets the lower boundary of the range of valid digits coming from the PSTN Central Office (CO) that is used with the DID Digit Translation Service.
param store-did-max	Sets the upper boundary of the range of digits that is valid in the Cisco Unified CME numbering plan that is used with the DID Digit Translation Service.
param store-did-min	Sets the lower boundary of the range of digits that is valid in the Cisco Unified CME numbering plan that is used with the DID Digit Translation Service.

param co-did-min

To set the lower boundary of the range of valid digits coming from the PSTN Central Office (CO) that is used with the Direct Inward Dial (DID) Digit Translation Service, use the **param co-did-min** command in application-parameter configuration mode. To disable this option, use the **no** form of this command.

param co-did-min *min-co-value*

no param co-did-min *min-co-value*

Syntax Description

min-co-value	Minimum value of digits coming from the CO. The digit string can be any length, but the string length must be the same in the **param co-did-max**, **param co-did-max**, **param store-did-min**, and **param store-did-max** commands.

Defaults

No minimum value is defined for the range of digits coming from the CO.

Command Modes

Application-parameter configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced to replace the **call application voice co-did-min** command.
12.4(9)T	Cisco Unified CME 4.0	This command replaced the **call application voice co-did-min** command and was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command defines the upper limit of the range of digits accepted from the CO when it is used with the Cisco Unified CallManager Express (Cisco Unified CME) DID Digit Translation Service. This service provides number translation for DID calls when the range of DID digits provided by the PSTN Central Office (CO) does not match the digits in the Cisco Unified CME extension numbers.

The Tcl script that provides the service accepts PSTN DID numbers of any length and maps them to the internal extension numbers that are assigned by a system administrator. Where necessary, a prefix is appended to the DID digits to create a valid extension number. The script uses the parameters that you input to determine the valid range of digits to be accepted from the CO, the valid range of digits in the local dial plan, and the prefix to append, if necessary. The script also handles DID calls that map to invalid extension numbers: a prompt is played and the calls are disconnected.

Examples

The following example configures DID Digit Translation Service on the Cisco Unified CME router. It sets a lower boundary of 00 and an upper boundary of 79 for the valid range of digits coming from the CO.

```
application
 service didapp tftp://192.168.254.254/scripts/did/app-THD-DID-2.0.0.1.tcl
  paramspace english index 1
  paramspace english language en
  paramspace english location tftp://192.168.254.254/apps/dir25/
  param secondary-prefix 4
  param did-prefix 5
  param co-did-min 00
  param co-did-max 79
  param store-did-min 00
  param store-did-max 79
```

Related Commands

Command	Description
application	Enters application configuration mode.
service	Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.
param co-did-max	Sets the upper boundary of the range of valid digits coming from the PSTN Central Office (CO) that is used with the DID Digit Translation Service.
param store-did-max	Sets the upper boundary of the range of digits that is valid in the Cisco Unified CME numbering plan that is used with the DID Digit Translation Service.
param store-did-min	Sets the lower boundary of the range of digits that is valid in the Cisco Unified CME numbering plan that is used with the DID Digit Translation Service.

param dial-by-extension-option

To assign a menu number to an Cisco CME B-ACD option by which callers can directly dial known extension numbers, use the **param dial-by-extension-option** command in application-parameter configuration mode. To disable this option, use the **no** form of this command.

param dial-by-extension-option *menu-number*

no param dial-by-extension-option *menu-number*

Syntax Description

menu-number	Menu option number to be associated with the dial-by-extension option. Range is from 1 to 9. There is no default.

Defaults

Dial-by-extension option is not assigned.

Command Modes

Application-parameter configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(14)T	Cisco CME 3.3	This command was introduced to replace the **call application voice dial-by-extension-option** command.

Usage Guidelines

This command is used with the Cisco Unified CME Basic Automatic Call Distribution and Auto-Attendant (B-ACD) service. This command is configured under the **service** command for an AA service.

This command allows you to designate a menu option number for callers to press if they want to dial an extension number that they already know. This command also enables the playing of the en_bacd_enter_dest.au audio file after a caller presses the dial-by-extension menu number. The default announcement in this audio file is “Please enter the extension number you want to reach.”

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

For more information, see the *Cisco Unified CallManager Express B-ACD and Tcl Call-Handling Applications* document for your release.

Examples

The following example sets up a B-ACD with an AA called acd1, which has an AA pilot number of (800) 555-0121. The call-queue service used with this AA is named callq. Callers to (800) 555-0121 can press 1 to dial an extension number (**param dial-by-extension-option 1** under **service acd1**) or press 2 to be connected to the hunt group with the pilot number 5072 (**param aa-hunt2 5072** under **service callq**).

```
dial-peer voice 1010 pots
 service acd1
 port 1/1/0
 incoming called-number 8005550121
```

```
.
.
.
application
 service callq tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd.tcl
  param queue-manager-debugs 1
  param aa-hunt2 5072
  param number-of-hunt-grps 1
  param queue-len 10
!
 service acd1 tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
  paramspace english location tftp://192.168.254.254/user1/prompts/
  paramspace english index 0
  paramspace english language en
  param handoff-string acd1
  param service-name callq
  param aa-pilot 8005550121
  param number-of-hunt-grps 1
  param dial-by-extension-option 1
  param second-greeting-time 45
  param call-retry-timer 20
  param max-time-call-retry 360
  param max-time-vm-retry 2
  param voice-mail 5007
```

Related Commands

Command	Description
application	Enters application configuration mode.
service	Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.

param did-prefix

To set a prefix to add to digits coming from the PSTN Central Office (CO) to create valid extension numbers when using the Direct Inward Dial (DID) Digit Translation Service, use the **param did-prefix** command in application-parameter configuration mode. To disable this option, use the **no** form of this command.

param did-prefix *did-prefix*

no param did-prefix *did-prefix*

Syntax Description

did-prefix	Prefix to add. Range is from 0 to 99.

Defaults

No prefix is defined.

Command Modes

Application-parameter configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME4.0	This command was introduced to replace the **call application voice did-prefix** command.
12.4(9)T	Cisco Unified CME4.0	This command replaced the **call application voice did-prefix** command and was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used with the Cisco Unified CallManager Express (Cisco Unified CME) DID Digit Translation Service, which provides number translation for DID calls when the range of DID digits provided by the PSTN Central Office (CO) does not match the digits in the Cisco Unified CME extension numbers.

The Tcl script that provides the service accepts PSTN DID numbers of any length and maps them to the internal extension numbers that are assigned by a system administrator. Where necessary, a prefix is appended to the DID digits to create a valid extension number. The script uses the parameters that you input to determine the valid range of digits to be accepted from the CO, the valid range of digits in the local dial plan, and the prefix to append, if necessary. The script also handles DID calls that map to invalid extension numbers: a prompt is played and the calls are disconnected.

Examples

The following example configures DID Digit Translation Service on the Cisco Unified CME router. It specifies that a prefix of 5 should be applied to the digits coming from the CO in order to construct a valid extension number.

```
application
 service didapp tftp://192.168.254.254/scripts/did/app-THD-DID-2.0.0.1.tcl
  paramspace english index 1
  paramspace english language en
  paramspace english location tftp://192.168.254.254/apps/dir25/
```

```
param secondary-prefix 4
param did-prefix 5
param co-did-min 00
param co-did-max 79
param store-did-min 00
param store-did-max 79
```

Related Commands

Command	Description
application	Enters application configuration mode.
service	Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.

param drop-through-option

To assign the drop-through option to a Cisco Unified CME B-ACD auto-attendant (AA) application, use the **param drop-through option** command in application-parameter configuration mode. To disable this option, use the **no** form of this command.

param drop-through-option *menu-number*

no param drop-through-option *menu-number*

Syntax Description

menu-number	Menu option number (aa-hunt number) to be associated with the drop-through option.

Defaults

Drop-through option is not assigned.

Command Modes

Application-parameter configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(14)T	Cisco CME 3.3	This command was introduced to replace the **call application voice drop-through-option** command.

Usage Guidelines

This command is used with the Cisco Unified CME Basic Automatic Call Distribution and Auto-Attendant (B-ACD) service. This command is configured under the **service** command for an AA service.

When an AA is configured for drop-through mode, the AA sends incoming calls directly to the call queue associated with the menu number specified in this command. Once in the queue, a caller hears ringback if an agent is available or music on hold (MOH) if all agents are busy. If a greeting prompt for drop-through mode is configured using the **param drop-through-prompt** command, a caller hears the prompt before being sent to the queue as described.

The menu option number is an aa-hunt number that is associated with an ephone hunt group using the **param aa-hunt** command.

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

For more information, see the *Cisco Unified CallManager Express B-ACD and Tcl Call-Handling Applications* document for your release.

Examples

The following example sets up a B-ACD with two AAs, both in drop-through mode. The first AA is called acdaa and it has an AA pilot number of (800) 555-0121. The second AA is aa-bcd and has an AA pilot number of (800) 555-0123. Both AAs use the call-queue service named callq. Callers to

(800) 555-0121 drop directly through to the hunt group with the pilot number 5072 after hearing the greeting prompt in the audio file named en_dto_welcome.au. Callers to (800) 555-0123 drop directly through to the hunt group with the pilot number 5071 without hearing any greeting.

```
dial-peer voice 1010 pots
 service acdaa
 port 1/1/0
 incoming called-number 8005550121

dial-peer voice 1020 pots
 service aa-bcd
 port 1/1/1
 incoming called-number 8005550123
.
.
.
application
 service callq tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd.tcl
  param queue-manager-debugs 1
  param aa-hunt1 5071
  param aa-hunt2 5072
  param number-of-hunt-grps 2
  param queue-len 10
!
 service acdaa tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
  paramspace english location tftp://192.168.254.254/user1/prompts/
  paramspace english index 0
  paramspace english language en
  param aa-pilot 8005550121
  param service-name callq
  param max-time-vm-retry 2
  param voice-mail 5007
  param call-retry-timer 20
  param number-of-hunt-grps 1
  param drop-through-prompt _bacd_dto_welcome.au
  param drop-through-option 2
  param second-greeting-time 45
  param handoff-string acdaa
  param max-time-call-retry 360
!
 service aa-bcd tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
  paramspace english location tftp://192.168.254.254/user1/prompts/
  paramspace english index 0
  paramspace english language en
  param aa-pilot 8005550123
  param service-name callq
  param second-greeting-time 60
  param max-time-call-retry 180
  param max-time-vm-retry 2
  param voice-mail 5007
  param call-retry-timer 5
  param handoff-string aa-bcd
  param drop-through-option 1
  param number-of-hunt-grps 1
```

Related Commands

Command	Description
application	Enters application configuration mode.
service	Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.

param drop-through-prompt

To associate an audio prompt file with the drop-through option for a Cisco Unified CME B-ACD automated attendant (AA) application, use the **param drop-through-prompt** command in application-parameter configuration mode. To disable the prompt, use the **no** form of this command.

param drop-through-prompt *audio-filename*

no param drop-through-prompt *audio-filename*

Syntax Description

audio-filename	Identifying part of the filename of the prompt to be played when calls for the drop-through option are answered.

Defaults

No prompt is designated for the drop-through option.

Command Modes

Application-parameter configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(14)T	Cisco CME 3.3	This command was introduced to replace the **call application voice drop-through-prompt** command.

Usage Guidelines

This command is used with the Cisco Unified CME Basic Automatic Call Distribution and Auto-Attendant (B-ACD) service. This command is configured under the **service** command for an AA service.

When an AA is configured for drop-through mode, the AA sends incoming calls directly to the call queue associated with the menu number specified in this command. Once in the queue, a caller hears ringback if an agent is available or music on hold (MOH) if all agents are busy. If an greeting prompt for drop-through mode is configured, a caller hears the prompt before being sent to the queue as described.

The menu option number is an aa-hunt number that is associated with an ephone hunt group using the **param aa-hunt** command.

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

For more information, see the *Cisco Unified CallManager Express B-ACD and Tcl Call-Handling Applications* document for your release.

Examples

The following example sets up a B-ACD with two AAs, both in drop-through mode. The first AA is called acdaa and it has an AA pilot number of (800) 555-0121. The second AA is aa-bcd and has an AA pilot number of (800) 555-0123. Both AAs use the call-queue service named callq. Callers to (800) 555-0121 drop directly through to the hunt group with the pilot number 5072 after hearing the greeting prompt in the audio file named en_dto_welcome.au. (The prefix en is specified in the

paramspace language command and is automatically added to the filename provided in the **param drop-through-prompt** command.) Callers to (800) 555-0123 drop directly through to the hunt group with the pilot number 5071 without hearing any greeting.

```
dial-peer voice 1010 pots
 service acdaa
 port 1/1/0
 incoming called-number 8005550121

dial-peer voice 1020 pots
 service aa-bcd
 port 1/1/1
 incoming called-number 8005550123
.
.
.
application
 service callq tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd.tcl
  param queue-manager-debugs 1
  param aa-hunt1 5071
  param aa-hunt2 5072
  param number-of-hunt-grps 2
  param queue-len 10
!
 service acdaa tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
  paramspace english location tftp://192.168.254.254/user1/prompts/
  paramspace english index 0
  paramspace english language en
  param aa-pilot 8005550121
  param service-name callq
  param max-time-vm-retry 2
  param voice-mail 5007
  param call-retry-timer 20
  param number-of-hunt-grps 1
  param drop-through-prompt _bacd_dto_welcome.au
  param drop-through-option 2
  param second-greeting-time 45
  param handoff-string acdaa
  param max-time-call-retry 360
!
 service aa-bcd tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
  paramspace english location tftp://192.168.254.254/user1/prompts/
  paramspace english index 0
  paramspace english language en
  param aa-pilot 8005550123
  param service-name callq
  param second-greeting-time 60
  param max-time-call-retry 180
  param max-time-vm-retry 2
  param voice-mail 5007
  param call-retry-timer 5
  param handoff-string aa-bcd
  param drop-through-option 1
  param number-of-hunt-grps 1
```

Related Commands

Command	Description
application	Enters application configuration mode.
service	Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.

param ea-password

To create a password for accessing the extension assigner application, use the **param ea-password** command in application service configuration mode.

param ea-password *password*

Syntax Description

password	Numeric string to be used as password for the extension assigner application. Password string must be 2 to 10 characters long and can contain numbers 0 to 9.

Command Default

No password is created.

Command Modes

Application service configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC4	Cisco Unified CME 4.0(3)	This command was introduced.
12.4(11)XJ	Cisco Unified CME 4.1	This command was introduced.
12.4(15)T	Cisco Unified CME 4.1	This command was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

This command creates a password for using the extension assigner application.

If this command is not configured, you cannot use the extension assigner application.

Note There is no **no** form of this command. To change or remove the password for the extension assigner application, remove the service using the **no** form of the **service** command in application configuration mode.

Examples

The following example shows that a password (1234) is configured for the extension assigner application:

```
application
 service EA flash:ea/app-cme-ea-2.0.0.0.tcl
  paramspace english index 0
  paramspace english language en
  param ea-password 1234
  paramspace english location flash:ea/
  paramspace english prefix en
```

Related Commands

Command	Description
application	Enters application configuration mode.
service	Loads and configures a specific, standalone application on a dial peer.

param handoff-string

To specify the name of a Cisco Unified CME B-ACD auto-attendant (AA) to be passed to the call-queue script, use the **param handoff-string** command in application-parameter configuration mode. To disable the handoff string, use the no form of this command.

param handoff-string *aa-service-name*

no param drop-through-prompt *aa-service-name*

Syntax Description

aa-service-name	Service name that was assigned to the AA script with the **service** command.

Defaults

No string is designated to be passed to the call-queue service.

Command Modes

Application-parameter configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(14)T	Cisco CME 3.3	This command was introduced to replace the **call application voice handoff-string** command.

Usage Guidelines

This command is used with the Cisco Unified CME Basic Automatic Call Distribution and Auto-Attendant (B-ACD) service. This command is configured under the **service** command for an AA service.

The handoff string is used only when the call-queue script starts for the first time or restarts after a failure.

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

For more information, see the *Cisco Unified CallManager Express B-ACD and Tcl Call-Handling Applications* document for your release.

Examples

The following example sets parameters for an AA application called aa and a call-queue application called queue. The direct-dial number to reach the AA service is (800) 555-0100. Callers to this number drop through to the ephone hunt group that has a pilot number of 5071 after hearing the initial prompt from the file en_dt_prompt.au. The AA name, aa is passed to the call-queue service in the **param handoff-string** command.

```
dial-peer voice 1000 pots
 service aa
 port 1/1/0
 incoming called-number 8005550100

ephone-hunt 10 sequential
 pilot 5071
 list 5011, 5012, 5013, 5014, 5015
```

```
!
application
 service callq tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd.tcl
  param queue-manager-debugs 1
  param aa-hunt1 5071
  param number-of-hunt-grps 1
  param queue-len 10
!
 service aa tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
  paramspace english location tftp://192.168.254.254/user1/prompts/
  paramspace english index 0
  paramspace english language en
  param aa-pilot 8005550100
  param number-of-hunt-groups 1
  param service-name callq
  param handoff-string aa
  param second-greeting-time 60
  param drop-through-option 1
  param drop-through-prompt _dt_prompt.au
  param call-retry-timer 15
  param max-time-call-retry 700
  param voice-mail 5000
  param max-time-vm-retry 2
```

Related Commands

Command	Description
application	Enters application configuration mode.
service	Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.

param max-extension-length

To specify the maximum number of digits callers can dial when they choose the dial-by-extension option from the Cisco Unified CME B-ACD service, use the **param max-extension-length** command in application-parameter configuration mode. To return to the default, use the **no** form of this command.

param max-extension-length *number*

no param max-extension-length *number*

Syntax Description

number	Number of digits. The lower limit is 0; there is no upper limit. The default is 5.

Defaults

The default number of digits callers can dial using the dial-by-extension option is 5.

Command Modes

Application-parameter configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(14)T	Cisco CME3.3	This command was introduced to replace the **call application voice max-extension-length** command.

Usage Guidelines

This command is used with the Cisco Unified CME Basic Automatic Call Distribution and Auto-Attendant (B-ACD) service. This command is configured under the **service** command for an AA service.

Use this command to restrict the number of digits that callers can dial when using the dial-by-extension option.

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

For more information, see the *Cisco Unified CallManager Express B-ACD and Tcl Call-Handling Applications* document for your release.

Examples

The following example sets parameters for an AA application called aa and a call-queue application called queue. The direct-dial number to reach the AA service is (800) 555-0100. Callers to this number can press 1 to be connected to the ephone hunt group with the pilot number 5071 or can press 2 to dial an extension number of 4 or fewer digits.

```
dial-peer voice 1000 pots
 service aa
 port 1/1/0
 incoming called-number 8005550100

ephone-hunt 10 sequential
 pilot 5071
```

```
 list 5011, 5012, 5013, 5014, 5015
!
application
 service callq tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd.tcl
  param queue-manager-debugs 1
  param aa-hunt1 5071
  param number-of-hunt-grps 1
  param queue-len 10
!
 service aa tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
  paramspace english location tftp://192.168.254.254/user1/prompts/
  paramspace english index 0
  paramspace english language en
  param aa-pilot 8005550100
  param welcome-prompt _aa_welcome.au
  param number-of-hunt-groups 1
  param dial-by-extension-option 2
  param max-extension-length 4
  param service-name callq
  param handoff-string aa
  param second-greeting-time 60
  param call-retry-timer 15
  param max-time-call-retry 700
  param voice-mail 5000
  param max-time-vm-retry 2
```

Related Commands

Command	Description
application	Enters application configuration mode.
service	Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.

param max-time-call-retry

To specify the maximum length of time for which calls to a Cisco Unified CME B-ACD service can stay in a call queue, use the **param max-time-call-retry command in** application-parameter configuration mode. To return to the default, use the **no** form of this command.

param max-time-call-retry *seconds*

no param max-time-call-retry *seconds*

Syntax Description

seconds	Maximum length of time that the call-queue service can keep redialing an ephone hunt group pilot number, in seconds. The range is from 0 to 3600. The default is 600.

Defaults

A call in a B-ACD call queue continues to try to connect to a hunt group for 600 seconds.

Command Modes

Application-parameter configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(14)T	Cisco CME 3.3	This command was introduced to replace the **call application voice max-time-call-retry** command.

Usage Guidelines

This command is used with the Cisco Unified CME Basic Automatic Call Distribution and Auto-Attendant (B-ACD) service. This command is configured under the **service** command for an AA service.

A call to a Cisco Unified CME B-ACD service is put into a call queue if the hunt group that the call tried to reach has no phones available to take the call because they are all busy. While the call is in the queue, a second greeting is played at intervals specified by the **param second-greeting-time** command. From the queue, the call makes retries to connect at intervals specified by the **param call-retry-timer** command until the maximum amount of time to be spent in the queue expires. The maximum amount of time is set by the **param max-time-call-retry** command. After the maximum amount of time expires, the call is routed to the alternate destination specified in the **param voice-mail** command. If the alternate destination number is busy, the call makes the number of retries to connect specified in the **param max-time-vm-retry** command. If the call is unable to connect to the alternate destination after the number of retries that has been specified, it is disconnected.

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

For more information, see the *Cisco Unified CallManager Express B-ACD and Tcl Call-Handling Applications* document for your release.

Examples

The following example sets parameters for an AA application called aa and a call-queue application called queue. The direct-dial number to reach the AA service is (800) 555-0100. Callers to this number can press 1 to be connected to the ephone hunt group with the pilot number 5071 or can press 2 to dial an extension number of 4 or fewer digits.

If a caller presses 2 and all the phones in ephone-hunt group 10 are busy, the call is put into a queue for hunt group 10. Every 60 seconds, the caller hears the second greeting, which is “Please continue to hold. An agent will be with you shortly.” Every 15 seconds, the call-queue service tries again to connect the call to the hunt group. If no phones become available before 700 seconds expire, the call is routed to extension 5000. If that extension is busy, the call-queue service retries it 2 times more. If the call still cannot be connected, it is disconnected.

```
dial-peer voice 1000 pots
 service aa
 port 1/1/0
 incoming called-number 8005550100

ephone-hunt 10 sequential
 pilot 5071
 list 5011, 5012, 5013, 5014, 5015
!
application
 service callq tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd.tcl
  param queue-manager-debugs 1
  param aa-hunt1 5071
  param number-of-hunt-grps 1
  param queue-len 10
!
 service aa tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
  paramspace english location tftp://192.168.254.254/user1/prompts/
  paramspace english index 0
  paramspace english language en
  param aa-pilot 8005550100
  param welcome-prompt _aa_welcome.au
  param number-of-hunt-groups 1
  param dial-by-extension-option 2
  param max-extension-length 4
  param service-name callq
  param handoff-string aa
  param second-greeting-time 60
  param call-retry-timer 15
  param max-time-call-retry 700
  param voice-mail 5000
  param max-time-vm-retry 2
```

Related Commands

Command	Description
application	Enters application configuration mode.
service	Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.

param max-time-vm-retry

To specify the maximum number of times that calls in a Cisco Unified CME B-ACD call queue can attempt to connect to the alternate destination number, use the **param max-time-vm-retry command in** application-parameter configuration mode. To return to the default, use the **no** form of this command

param max-time-vm-retry *number*

no param max-time-vm-retry *number*

Syntax Description

number	Number of times that the alternate destination number is redialed by the call-queue service. Range is from 1 to 3. Default is 1.

Defaults

A call in a B-ACD call queue tries to connect to an alternate destination number 1 time.

Command Modes

Application-parameter configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(14)T	Cisco CME 3.3	This command was introduced to replace the **call application voice max-time-vm-retry** command.

Usage Guidelines

This command is used with the Cisco Unified CME Basic Automatic Call Distribution and Auto-Attendant (B-ACD) service. This command is configured under the **service** command for an AA service.

A call to a Cisco Unified CME B-ACD service is put into a call queue if the hunt group that the call tried to reach has no phones available to take the call because they are all busy. While the call is in the queue, a second greeting is played at intervals specified by the **param second-greeting-time** command. From the queue, the call makes retries to connect at intervals specified by the **param call-retry-timer** command until the maximum amount of time to be spent in the queue expires. The maximum amount of time is set by the **param max-time-call-retry** command. After the maximum amount of time expires, the call is routed to the alternate destination specified in the **param voice-mail** command. If the alternate destination number is busy, the call makes the number of retries to connect specified in the **param max-time-vm-retry** command. If the call is unable to connect to the alternate destination after the number of retries that has been specified, it is disconnected.

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

For more information, see the *Cisco Unified CallManager Express B-ACD and Tcl Call-Handling Applications* document for your release.

Examples

The following example sets parameters for an AA application called aa and a call-queue application called queue. The direct-dial number to reach the AA service is (800) 555-0100. Callers to this number can press 1 to be connected to the ephone hunt group with the pilot number 5071 or can press 2 to dial an extension number of 4 or fewer digits.

If a caller presses 2 and all the phones in ephone-hunt group 10 are busy, the call is put into a queue for hunt group 10. Every 60 seconds, the caller hears the second greeting, which is “Please continue to hold. An agent will be with you shortly.” Every 15 seconds, the call-queue service tries again to connect the call to the hunt group. If no phones become available before 700 seconds expire, the call is routed to extension 5000. If that extension is busy, the call-queue service retries it 2 times more. If the call still cannot be connected, it is now disconnected.

```
dial-peer voice 1000 pots
 service aa
 port 1/1/0
 incoming called-number 8005550100

ephone-hunt 10 sequential
 pilot 5071
 list 5011, 5012, 5013, 5014, 5015
!
application
 service callq tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd.tcl
  param queue-manager-debugs 1
  param aa-hunt1 5071
  param number-of-hunt-grps 1
  param queue-len 10
!
 service aa tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
  paramspace english location tftp://192.168.254.254/user1/prompts/
  paramspace english index 0
  paramspace english language en
  param aa-pilot 8005550100
  param welcome-prompt _aa_welcome.au
  param number-of-hunt-groups 1
  param dial-by-extension-option 2
  param max-extension-length 4
  param service-name callq
  param handoff-string aa
  param second-greeting-time 60
  param call-retry-timer 15
  param max-time-call-retry 700
  param voice-mail 5000
  param max-time-vm-retry 2
```

Related Commands

Command	Description
application	Enters application configuration mode.
service	Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.

param number-of-hunt-grps

To specify the number of hunt groups used with a Cisco Unified CME B-ACD call-queue or AA service, use the **param number-of-hunt-grps** command in application-parameter configuration mode. To return to the default, use the **no** form of this command.

param number-of-hunt-grps *number*

no param number-of-hunt-grps *number*

Syntax Description

number	Number of ephone hunt groups used by the service. Range is 1 to 10 for the call-queue service and 1 to 3 for an automated attendant (AA) service.

Defaults

This parameter is not set.

Command Modes

Application-parameter configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(14)T	Cisco CME 3.3	This command was introduced to replace the **call application voice number-of-hunt-grps** command.

Usage Guidelines

This command is used with the Cisco Unified CME Basic Automatic Call Distribution and Auto-Attendant (B-ACD) service. This command is configured both under the **service** command for the call-queue service and under the **service** command for an AA service.

The number of hunt groups specified for the call-queue service is the total of the number of hunt groups used with all the AAs with which it is associated. For example, if a B-ACD has three AAs, each with two hunt groups, the number of hunt groups for each AA is two and the number of hunt groups for the call-queue service is six.

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

For more information, see the *Cisco Unified CallManager Express B-ACD and Tcl Call-Handling Applications* document for your release.

Examples

A call-queue service called CQ is set up to work with two AA services. CQ lists 4 as the number of hunt groups it uses. AA1 is associated with 3 hunt groups, and its callers hear the following prompt: “Press 1 for sales, press 2 for service, press 0 for operator.” AA2 uses drop-through mode. Its callers do not hear a prompt and are directly connected to the single hunt group that is associated with it.

```
application
 service CQ tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd.tcl
  param queue-manager-debugs 1
  param aa-hunt1 1001
  param aa-hunt2 2001
```

```
 param aa-hunt3 3001
 param aa-hunt4 4001
 param number-of-hunt-grps 4
 param queue-len 10

service AA1 tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
 paramspace english location tftp://192.168.254.254/user1/prompts/
 paramspace english index 0
 paramspace english language en
 param aa-pilot 8005550111
 param number-of-hunt-groups 3
 param service-name CQ
 param welcome-prompt _bacd_welcome.au
 param handoff-string AA1

service AA2 tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
 paramspace english location tftp://192.168.254.254/user1/prompts/
 paramspace english index 0
 paramspace english language en
 param aa-pilot 8005550122
 param number-of-hunt-groups 1
 param service-name CQ
 param drop-through-option 4
 param handoff-string AA2
```

Related Commands

Command	Description
application	Enters application configuration mode.
service	Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.

param queue-len

To specify the number of calls that can be held in each call queue in a Cisco Unified CME B-ACD service, use the **param queue-len** command in application-parameter configuration mode. To return to the default, use the **no** form of this command.

param queue-len *number*

no param queue-len *number*

Syntax Description

number	Number of calls that can be held in a call queue. Range is 1 to 30. Default is 10.

Defaults

The default queue length is 10.

Command Modes

Application-parameter configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(14)T	Cisco CME 3.3	This command was introduced to replace the **call application voice queue-len** command.

Usage Guidelines

This command is used with the Cisco Unified CME Basic Automatic Call Distribution and Auto-Attendant (B-ACD) service. This command is configured under the **service** command for a call-queue service.

This command specifies the maximum number of calls that can be held in a call queue for a hunt group used with B-ACD when all of the hunt group member phones are busy.

Note that having calls in queue keeps PSTN ports occupied for a longer time, and you may want to plan for more ports if you have longer queues. The maximum number of calls allowed in the queues of ephone hunt groups must be based on the number of ports or trunks available. For example, if you have 20 foreign exchange office (FXO) ports and two ephone hunt groups, you can configure a maximum of ten calls per ephone hunt-group queue using the **param queue-len 10** command. You can use the same configuration if you have a single T1 trunk, which supports 23 channels.

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

For more information, see the *Cisco Unified CallManager Express B-ACD and Tcl Call-Handling Applications* document for your release.

Examples

A call-queue service called CQ is set up to work with two AA services. CQ lists four as the number of hunt groups it uses. AA1 is associated with three hunt groups, and its callers hear the following prompt: "Press 1 for sales, press 2 for service, press 0 for operator." AA2 uses drop-through mode. Its callers do not hear a prompt and are directly connected to the single hunt group that is associated with it. Up to 12 calls can be held in the call queue for each hunt group if all the phones in the hunt group are busy.

```
application
 service CQ tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd.tcl
  param queue-manager-debugs 1
  param aa-hunt1 1001
  param aa-hunt2 2001
  param aa-hunt3 3001
  param aa-hunt4 4001
  param number-of-hunt-grps 4
  param queue-len 12

 service AA1 tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
  paramspace english location tftp://192.168.254.254/user1/prompts/
  paramspace english index 0
  paramspace english language en
  param aa-pilot 8005550111
  param number-of-hunt-groups 3
  param service-name CQ
  param welcome-prompt _bacd_welcome.au
  param handoff-string AA1

 service AA2 tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
  paramspace english location tftp://192.168.254.254/user1/prompts/
  paramspace english index 0
  paramspace english language en
  param aa-pilot 8005550122
  param number-of-hunt-groups 1
  param service-name CQ
  param drop-through-option 4
  param handoff-string AA2
```

Related Commands

Command	Description
application	Enters application configuration mode.
service	Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.

param que ue-manager-debugs

To enable the collection of call-queue debug information in a Cisco Unified CME B-ACD service, use the **param queue-manager-debugs** command in application-parameter configuration mode. To remove the setting, use the **no** form of this command with the keyword that was previously used.

param queue-manager-debugs [**0** | **1**]

no param queue-manager-debugs [**0** | **1**]

Syntax Description

0	Disables collection of call-queue debug information.
1	Enables collection of call-queue debug information

Command Default

Collection of debug information is disabled.

Command Modes

Application-parameter configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(14)T	Cisco CME 3.3	This command was introduced to replace the **call application voice queue-manager-debugs** command.

Usage Guidelines

This command is used with the Cisco Unified CME Basic Automatic Call Distribution and Auto-Attendant (B-ACD) service. This command is configured under the **service** command for the call-queue service.

This command enables the collection of data regarding call queue activity. It is used in conjunction with the **debug voip ivr script** command. Both commands must be enabled at the same time.

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

For more information, see the *Cisco Unified CallManager Express B-ACD and Tcl Call-Handling Applications* document for your release.

Examples

A call-queue service called CQ is set up to work with two AA services. CQ lists four as the number of hunt groups it uses. AA1 is associated with three hunt groups, and its callers hear the following prompt: "Press 1 for sales, press 2 for service, press 0 for operator." AA2 uses drop-through mode. Its callers do not hear a prompt and are directly connected to the single hunt group that is associated with it. Up to ten calls can be held in the call queue for each hunt group if all the phones in the hunt group are busy with other calls. Call-queue debugging is enabled.

```
application
 service CQ tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd.tcl
  param queue-manager-debugs 1
  param aa-hunt1 1001
  param aa-hunt2 2001
```

```
 param aa-hunt3 3001
 param aa-hunt4 4001
 param number-of-hunt-grps 4
 param queue-len 10

service AA1 tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
 paramspace english location tftp://192.168.254.254/user1/prompts/
 paramspace english index 0
 paramspace english language en
 param aa-pilot 8005550111
 param number-of-hunt-groups 3
 param service-name CQ
 param welcome-prompt _bacd_welcome.au
 param handoff-string AA1

service AA2 tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
 paramspace english location tftp://192.168.254.254/user1/prompts/
 paramspace english index 0
 paramspace english language en
 param aa-pilot 8005550122
 param number-of-hunt-groups 1
 param service-name CQ
 param drop-through-option 4
 param handoff-string AA2
```

Related Commands

Command	Description
application	Enters application configuration mode.
service	Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.

param secondary-prefix

To set a prefix to add to digits coming from the PSTN Central Office (CO) to route calls from a secondary Cisco Unified CME router to a primary Cisco Unified CME router when using the Direct Inward Dial (DID) Digit Translation Service, use the **param secondary-prefix** command in application-parameter configuration mode. To disable this option, use the **no** form of this command.

param secondary-prefix *secondary-prefix*

no param secondary-prefix *secondary-prefix*

Syntax Description

secondary-prefix	Prefix to add to digits in order to route calls to the primary Cisco Unified CME router. Range is from 0 to 99.

Defaults

No prefix is defined.

Command Modes

Application-parameter configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced to replace the **call application voice secondary-prefix** command.
12.4(9)T	Cisco Unified CME 4.0	This command replaced the **call application voice secondary-prefix** command and was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used with the Cisco Unified CallManager Express (Cisco Unified CME) DID Digit Translation Service, which provides number translation for DID calls when the range of DID digits provided by the PSTN Central Office (CO) does not match the digits in the Cisco Unified CME extension numbers.

The Tcl script that provides the service accepts PSTN DID numbers of any length and maps them to the internal extension numbers that are assigned by a system administrator. Where necessary, a prefix is appended to the DID digits to create a valid extension number. The script uses the parameters that you input to determine the valid range of digits to be accepted from the CO, the valid range of digits in the local dial plan, and the prefix to append, if necessary. The script also handles DID calls that map to invalid extension numbers: a prompt is played and the calls are disconnected.

When calls are received by a secondary Cisco Unified CME router, they are routed to the primary router by configuring an H.323 VoIP dial peer and matching the destination pattern for that dial peer. The DID prefix that was configured for use with the DID script is appended to the incoming DID digits first. The secondary prefix is appended next. For example, if the incoming DID digits are 25, the DID prefix is 3, and the secondary prefix is 7, the transformed number will be 7325. The transformed number matches a VoIP dial peer, which uses the forward-digits command to send only the three relevant digits, the extension number, to the primary router.

See the *Cisco Unified CallManager Express B-ACD and Tcl Call-Handling Applications* document for your release.

Examples

The following example configures a Basic DID application on the Cisco Unified CME router. It sets a prefix of 5 to apply to the digits coming from the CO in order to construct a valid extension number. Then the secondary prefix (4) is appended. If the incoming DID digits are 25, the DID prefix is 5, and the secondary prefix is 4, then the transformed number is 4525. The transformed number matches VoIP dial peer 1000. The VoIP dial peer sends calls to the primary Cisco Unified CME router using the IP address that is entered in the session target command. The dial peer uses the forward-digits command to send the extension number, 525, to the primary Cisco Unified CME router.

```
dial-peer voice 1000 voip
 destination-pattern 45..
 session target ipv4:10.1.1.1
 dtmf-relay h245-alphanumeric
 codec g711ulaw
 forward-digits 3

application
 service didapp tftp://192.168.254.254/scripts/did/app-THD-DID-2.0.0.1.tcl
  paramspace english index 1
  paramspace english language en
  paramspace english location tftp://192.168.254.254/apps/dir25/
  param secondary-prefix 4
  param did-prefix 5
  param co-did-min 00
  param co-did-max 79
  param store-did-min 00
  param store-did-max 79
```

Related Commands

Command	Description
application	Enters application configuration mode.
service	Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.

param second-greeting-time

To set the length of the intervals between playouts of the second greeting to calls waiting in hunt group call queues that are part of a Cisco Unified CME B-ACD service, use the **param second-greeting-time command in** application-parameter configuration mode. To return to the default, use the **no** form of this command

param second-greeting-time *seconds*

no param max-time-vm-retry *seconds*

Syntax Description

seconds	Length of time intervals between playouts of the second greeting to calls in a B-ACD call queue, in seconds. Range is from 30 to 120. Default is 60.

Defaults

The second greeting is played out every 60 seconds to calls in B-ACD call queues.

Command Modes

Application-parameter configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(14)T	Cisco CME 3.3	This command was introduced to replace the **call application voice second-greeting-time** command.

Usage Guidelines

This command is used with the Cisco Unified CME Basic Automatic Call Distribution and Auto-Attendant (B-ACD) service. This command is configured under the **service** command for an AA service.

A call to a Cisco Unified CME B-ACD service is put into a call queue if the hunt group that the call tried to reach has no phones available to take the call because they are all busy. While the call is in the queue, a second greeting is played at intervals specified by the **param second-greeting-time** command. From the queue, the call retries to connect to the hunt group at intervals specified by the **param call-retry-timer** command until the maximum amount of time to be spent in the queue expires. The maximum amount of time is set by the **param max-time-call-retry** command. After the maximum amount of time expires, the call is routed to the alternate destination specified in the **param voice-mail** command. If the alternate destination number is busy, the call makes the number of retries to connect specified in the **param max-time-vm-retry** command. If the call is unable to connect to the alternate destination after the number of retries that has been specified, it is disconnected.

The second greeting is stored in the audio file named en_bacd_allagentsbusy.au. You can rerecord over the default message that is provided in the file, but you cannot change the name of the file.

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

For more information, see the *Cisco Unified CallManager Express B-ACD and Tcl Call-Handling Applications* document for your release.

Examples

The following example sets parameters for an AA application called aa and a call-queue application called queue. The direct-dial number to reach the AA service is (800) 555-0100. Callers to this number can press 1 to be connected to the ephone hunt group with the pilot number 5071 or can press 2 to dial an extension number of 4 or fewer digits.

If a caller presses 2 and all the phones in ephone-hunt group 10 are busy, the call is put into a queue for hunt group 10. Every 60 seconds, the caller hears the second greeting, which is "Please continue to hold. An agent will be with you shortly." Every 15 seconds, the call-queue service tries again to connect the call to the hunt group. If no phones become available before 700 seconds expire, the call is routed to extension 5000. If that extension is busy, the call-queue service retries it 2 times more. If the call still cannot be connected, it is now disconnected.

```
dial-peer voice 1000 pots
 service aa
 port 1/1/0
 incoming called-number 8005550100

ephone-hunt 10 sequential
 pilot 5071
 list 5011, 5012, 5013, 5014, 5015
!
application
 service callq tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd.tcl
  param queue-manager-debugs 1
  param aa-hunt1 5071
  param number-of-hunt-grps 1
  param queue-len 10
!
 service aa tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
  paramspace english location tftp://192.168.254.254/user1/prompts/
  paramspace english index 0
  paramspace english language en
  param aa-pilot 8005550100
  param welcome-prompt _aa_welcome.au
  param number-of-hunt-groups 1
  param dial-by-extension-option 2
  param max-extension-length 4
  param service-name callq
  param handoff-string aa
  param second-greeting-time 60
  param call-retry-timer 15
  param max-time-call-retry 700
  param voice-mail 5000
  param max-time-vm-retry 2
```

Related Commands

Command	Description
application	Enters application configuration mode.
service	Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.

param service-name

To specify a Cisco Unified CME B-ACD call-queue service to use with an automated attendant (AA) service, use the **param service-name** command in application-parameter configuration mode. To return to the default, use the **no** form of this command.

param service-name *queue-service-name*

no param service-name *queue-service-name*

Syntax Description

queue-service-name	Name that was assigned to the B-ACD call-queue service with the **service** command.

Defaults

No call-queue service is specified.

Command Modes

Application-parameter configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(14)T	Cisco CME 3.3	This command was introduced to replace the **call application voice service-name** command.

Usage Guidelines

This command is used with the Cisco Unified CME Basic Automatic Call Distribution and Auto-Attendant (B-ACD) service. This command is configured under the **service** command for an AA service.

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

For more information, the *Cisco Unified CallManager Express B-ACD and Tcl Call-Handling Applications* document for your release.

Examples

A call-queue service called CQ is set up to work with two AA services. CQ lists four as the number of hunt groups it uses. AA1 is associated with three hunt groups, and its callers hear the following prompt: “Press 1 for sales, press 2 for service, press 0 for operator.” AA2 uses drop-through mode. Its callers do not hear a prompt and are directly connected to the single hunt group that is associated with it. Up to ten calls can be held in the call queue for each hunt group if all the phones in the hunt group are busy with other calls. Call-queue debugging is enabled.

```
application
 service CQ tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd.tcl
  param queue-manager-debugs 1
  param aa-hunt1 1001
  param aa-hunt2 2001
  param aa-hunt3 3001
  param aa-hunt4 4001
  param number-of-hunt-grps 4
  param queue-len 10
```

```
service AA1 tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
 paramspace english location tftp://192.168.254.254/user1/prompts/
 paramspace english index 0
 paramspace english language en
 param aa-pilot 8005550111
 param number-of-hunt-groups 3
 param service-name CQ
 param welcome-prompt _bacd_welcome.au
 param handoff-string AA1

service AA2 tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
 paramspace english location tftp://192.168.254.254/user1/prompts/
 paramspace english index 0
 paramspace english language en
 param aa-pilot 8005550122
 param number-of-hunt-groups 1
 param service-name CQ
 param drop-through-option 4
 param handoff-string AA2
```

Related Commands

Command	Description
application	Enters application configuration mode.
service	Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.

param store-did-max

To set the upper boundary of the range of digits that is valid in the Cisco Unified CME numbering plan used with the Direct Inward Dial (DID) Digit Translation Service, use the **param store-did-max** command in global configuration mode. To disable this option, use the **no** form of this command.

param store-did-max *max-store-value*

no param store-did-max *max-store-value*

Syntax Description

max-store-value	Maximum value of digits in the Cisco Unified CME dial plan. The digit string can be any length, but the string length must be the same in the **param co-did-max**, **param co-did-min**, **param store-did-max**, and **param store-did-min** commands.

Defaults

No maximum value is defined for the range of digits in the dial plan.

Command Modes

Application-parameter configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced to replace the **call application voice store-did-max** command.
12.4(4)XC	Cisco Unified CME 4.0	This command replaced the **call application voice store-did-max** command and was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command defines the upper limit of the range of digits in the site dial plan for the Cisco Unified CallManager Express (Cisco Unified CME) Direct Inward Dial Digit Translation Service, which provides number translation for DID calls when the DID digits provided by the PSTN Central Office (CO) do not match the digits in the Cisco Unified CME extension numbers.

The Tcl script that provides the service accepts PSTN DID numbers of any length and maps them to the internal extension numbers that are assigned by a system administrator. Where necessary, a prefix is appended to the DID digits to create a valid extension number. The script uses the parameters that you input to determine the valid range of digits to be accepted from the CO, the valid range of digits in the local dial plan, and the prefix to append, if necessary. The script also handles DID calls that map to invalid extension numbers. A prompt is played and the calls are disconnected.

For more information, see the *Cisco Unified CallManager Express B-ACD and Tcl Call-Handling Applications* document for your release.

Examples

The following example configures Direct Inward Dial Digit Translation Service on the Cisco Unified CME router. It sets a lower boundary of 00 and an upper boundary of 79 for the range of digits in the Cisco Unified CME extension dial plan.

```
application
 service didapp tftp://192.168.254.254/scripts/did/app-THD-DID-2.0.0.1.tcl
  paramspace english index 1
  paramspace english language en
  paramspace english location tftp://192.168.254.254/apps/dir25/
  param secondary-prefix 4
  param did-prefix 5
  param co-did-min 00
  param co-did-max 79
  param store-did-min 00
  param store-did-max 79
```

Related Commands

Command	Description
application	Enters application configuration mode.
service	Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.
param co-did-max	Sets the upper boundary of the range of valid digits coming from the PSTN Central Office (CO) that is used with the Direct Inward Dial Digit Translation Service.
param co-did-min	Sets the lower boundary of the range of valid digits coming from the PSTN Central Office (CO) that is used with the Direct Inward Dial Digit Translation Service.
param store-did-min	Sets the lower boundary of the range of digits that is valid in the Cisco Unified CME numbering plan that is used with the Direct Inward Dial Digit Translation Service.

param store-did-min

To set the lower boundary of the range of digits that is valid in the Cisco Unified CME numbering plan used with the Direct Inward Dial (DID) Digit Translation Service, use the **param store-did-min** command in application-parameter configuration mode. To disable this option, use the **no** form of this command.

param store-did-min *min-store-value*

no param store-did-min *min-store-value*

Syntax Description

min-store-value	Minimum value of digits in the Cisco Unified CME dial plan. The digit string can be any length, but the string length must be the same in the **param co-did-max**, **param co-did-min**, **param store-did-max**, and **param store-did-min** commands.

Defaults

No minimum value is defined for the range of digits in the dial plan.

Command Modes

Application-parameter configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced to replace the **call application voice store-did-min** command.
12.4(9)T	Cisco Unified CME 4.0	This command replaced the **call application voice store-did-min** command and was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command defines the lower limit of the range of digits in the site dial plan when it is used with the Cisco Unified CallManager Express (Cisco Unified CME) DID Digit Translation Service. This service provides number translation for DID calls when the range of DID digits provided by the PSTN Central Office (CO) does not match the digits in the Cisco Unified CME extension numbers.

The Tcl script that provides the service accepts PSTN DID numbers of any length and maps them to the internal extension numbers that are assigned by a system administrator. Where necessary, a prefix is appended to the DID digits to create a valid extension number. The script uses the parameters that you input to determine the valid range of digits to be accepted from the CO, the valid range of digits in the local dial plan, and the prefix to append, if necessary. The script also handles DID calls that map to invalid extension numbers: a prompt is played and the calls are disconnected.

Examples

The following example configures DID Digit Translation Service on the Cisco Unified CME router. It sets a lower boundary of 00 and an upper boundary of 79 for the range of digits in the Cisco Unified CME extension dial plan.

```
application
 service didapp tftp://192.168.254.254/scripts/did/app-THD-DID-2.0.0.1.tcl
  paramspace english index 1
  paramspace english language en
  paramspace english location tftp://192.168.254.254/apps/dir25/
  param secondary-prefix 4
  param did-prefix 5
  param co-did-min 00
  param co-did-max 79
  param store-did-min 00
  param store-did-max 79
```

Related Commands

Command	Description
application	Enters application configuration mode.
service	Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.
param co-did-max	Sets the upper boundary of the range of valid digits coming from the PSTN Central Office (CO) that is used with the DID Digit Translation Service.
param co-did-min	Sets the lower boundary of the range of valid digits coming from the PSTN Central Office (CO) that is used with the DID Digit Translation Service.
param store-did-max	Sets the upper boundary of the range of digits that is valid in the Cisco Unified CME numbering plan that is used with the DID Digit Translation Service.

param voice-mail

To set an alternate destination number to which to route calls that cannot be connected to a hunt group that is part of a Cisco Unified CME B-ACD service, use the **param voice-mail command in** application-parameter configuration mode. To return to the default, use the **no** form of this command

param voice-mail *number*

no param voice-mail *number*

Syntax Description

number	Extension number to which to route calls. The number must be associated with a dial peer that is reachable by the Cisco Unified CME system.

Defaults

No alternate destination number is set.

Command Modes

Application-parameter configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(14)T	Cisco CME 3.3	This command was introduced to replace the **call application voice voice-mail** command.

Usage Guidelines

This command is used with the Cisco Unified CME Basic Automatic Call Distribution and Auto-Attendant (B-ACD) service. This command is configured under the **service** command for an AA service.

Calls are diverted to an alternate destination only when one of the following criteria is met:

- The hunt group to which the call has been transferred is unavailable because all members are logged out.
- The call-queue maximum retry timer has expired.

The alternate destination can be any number at which you can assure call coverage, such as a voice-mail number, a permanently staffed number, or a number that rings an overhead night bell. Once a call is diverted to an alternate destination, it is no longer controlled by the B-ACD service. This parameter is set with the **param voice-mail** command.

If you send calls to a voice-mail system as an alternate destination, be sure to set up the voice-mail system as specified in the documentation for the system.

If you specify a number for an alternate destination, the number must be associated with a dial peer that is reachable by the Cisco Unified CME system.

For any configuration changes to take effect, you must reload the Cisco Unified CME B-ACD scripts.

For more information about B-ACD, see the *Cisco Unified CallManager Express B-ACD and Tcl Call-Handling Applications* document for your release.

Examples

The following example sets parameters for an AA application called aa and a call-queue application called queue. The direct-dial number to reach the AA service is (800) 555-0100. Callers to this number can press 1 to be connected to the ephone hunt group with the pilot number 5071 or can press 2 to dial an extension number of 4 or fewer digits.

If a caller presses 2 and all the phones in ephone-hunt group 10 are busy, the call is put into a queue for hunt group 10. Every 60 seconds, the caller hears the second greeting, which is "Please continue to hold. An agent will be with you shortly." Every 15 seconds, the call-queue service tries again to connect the call to the hunt group. If no phones become available before 700 seconds expire, the call is routed to extension 5000, which is the alternate destination. If that extension is busy, the call-queue service retries it 2 times more. If the call still cannot be connected, it is disconnected.

```
dial-peer voice 1000 pots
 service aa
 port 1/1/0
 incoming called-number 8005550100

ephone-hunt 10 sequential
 pilot 5071
 list 5011, 5012, 5013, 5014, 5015
!
application
 service callq tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd.tcl
  param queue-manager-debugs 1
  param aa-hunt1 5071
  param number-of-hunt-grps 1
  param queue-len 10
!
 service aa tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
  paramspace english location tftp://192.168.254.254/user1/prompts/
  paramspace english index 0
  paramspace english language en
  param aa-pilot 8005550100
  param welcome-prompt _aa_welcome.au
  param number-of-hunt-groups 1
  param dial-by-extension-option 2
  param max-extension-length 4
  param service-name callq
  param handoff-string aa
  param second-greeting-time 60
  param call-retry-timer 15
  param max-time-call-retry 700
  param voice-mail 5000
  param max-time-vm-retry 2
```

Related Commands

Command	Description
application	Enters application configuration mode.
service	Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.

param welcome-prompt

To specify an audio file containing a prompt to be played as a welcome for callers to an automated attendant (AA) that is part of a Cisco Unified CME B-ACD service, use the **param welcome-prompt command in** application-parameter configuration mode. To return to the default, use the **no** form of this command

param welcome-prompt *audio-filename*

no param welcome-prompt *audio-filename*

Syntax Description

audio-filename	Identifier part of name of the audio file that contains the welcome greeting to be played when callers first reach the Cisco Unified CME B-ACD service. This name does not include the language prefix and it must begin with an underscore. Default is _bacd_welcome.au.

Defaults

The audio file named en_bacd_welcome.au is used as a welcome prompt.

Command Modes

Application-parameter configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(14)T	Cisco CME 3.3	This command was introduced to replace the **call application voice voice-mail** command.

Usage Guidelines

This command is used with the Cisco Unified CME Basic Automatic Call Distribution and Auto-Attendant (B-ACD) service. This command is configured under the **service** command for an AA service.

Each AA service that is used with the Cisco Unified CME B-ACD service needs a welcome greeting to tell callers the destination they have reached and, sometimes, the options that they have. The en_bacd_welcome. au audio file is used by default. It announces "Thank you for calling," and includes a two-second pause after the message. The filename of the welcome prompt audio file has two parts: a two-letter prefix that denotes a language code specified in the **paramspace language** command, and the identifying part that indicates the purpose of the file. In the default welcome prompt audio file, the prefix is en and the identifying part is _bacd_welcome.au. Note that the identifying part starts with an underscore.

If your Cisco Unified CME B-ACD service uses a single AA application, you can record a custom welcome greeting in the audio file named en_welcome_prompt.au and record instructions about menu choices in the audio file named en_bacd_options_menu.au.

If your Cisco Unified CME B-ACD service uses multiple AA applications, you will need separate greetings and menu options for each AA. Use the following guidelines:

- Record a separate welcome prompt for each AA application, using a different name for the audio file for each welcome prompt. For example, en_welcome_aa1.au and en_welcome_aa2.au. The welcome prompts that you record in these files should include both the greeting and the instructions about menu options.
- Record silence in the audio file en_bacd_options_menu.au. A minimum of one second of silence must be recorded. Note that you cannot change the identifier part of the name of this audio file.

For any Cisco Unified CME B-ACD configuration changes to take effect, you must reload the scripts.

For more information, see the *Cisco Unified CallManager Express B-ACD and Tcl Call-Handling Applications* document for your release.

Examples

The following example sets parameters for two AA applications, called aa1 and aa2, and a call-queue application called queue. The direct-dial numbers to reach the AA services are (800) 555-0100 for aa1 and (800) 555-0110 for aa2. Callers to aa1 can press 1 to be connected to the ephone hunt group with the pilot number 5071 or can press 2 to dial an extension number of 4 or fewer digits. Callers to aa2 can press 2 to dial an extension number of 4 or fewer digits or press 3 to be connected to the ephone hunt group with the pilot number 5073. Both AAs share an operator hunt group, which is menu option 4.

The welcome prompt for aa1 is “Thank you for calling the Sales department. Press 1 to place an order. Press 2 if you know the extension of the party you want, or press 0 to speak to an operator.” The filename of the audio file that contains this welcome prompt is en_aa1_welcome.au.

The welcome prompt for aa2 is “Thank you for calling the Service department. Press 2 if you know the extension of the party you want. Press 3 to speak to a service technician or press 0 to speak to an operator.” The filename of the audio file that contains this welcome prompt is en_aa2_welcome.au.

```
dial-peer voice 1000 pots
 service aa1
 port 1/1/0
 incoming called-number 8005550100

dial-peer voice 1100 pots
 service aa2
 port 1/1/1
 incoming called-number 8005550110

ephone-hunt 10 sequential
 pilot 5071
 list 5011, 5012, 5013, 5014, 5015

ephone-hunt 11 sequential
 pilot 5073
 list 5021, 5022, 5023, 5024, 5025
!
application
 service callq tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd.tcl
  param queue-manager-debugs 1
  param aa-hunt1 5071
  param aa-hunt3 5073
  param aa-hunt4 6000
  param number-of-hunt-grps 3
  param queue-len 10
!
 service aa1 tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
  paramspace english location tftp://192.168.254.254/user1/prompts/
  paramspace english index 0
  paramspace english language en
```

```
 param aa-pilot 8005550100
 param welcome-prompt _aa1_welcome.au
 param number-of-hunt-groups 2
 param dial-by-extension-option 2
 param max-extension-length 4
 param service-name callq
 param handoff-string aa1
 param second-greeting-time 60
 param call-retry-timer 15
 param max-time-call-retry 700
 param voice-mail 5000
 param max-time-vm-retry 2

service aa2 tftp://192.168.254.254/user1/CallQ/B-ACD/app-b-acd-aa.tcl
 paramspace english location tftp://192.168.254.254/user1/prompts/
 paramspace english index 0
 paramspace english language en
 param aa-pilot 8005550110
 param welcome-prompt _aa2_welcome.au
 param number-of-hunt-groups 2
 param dial-by-extension-option 2
 param max-extension-length 4
 param service-name callq
 param handoff-string aa2
 param second-greeting-time 60
 param call-retry-timer 15
 param max-time-call-retry 700
 param voice-mail 5000
 param max-time-vm-retry 2
```

Related Commands

Command	Description
application	Enters application configuration mode.
service	Enters application-parameter configuration mode and specifies a name for the application and the location of the Tcl script to load for the application.

paramspace callsetup after-hours-exempt

To specify that an individual dial peer does not have any of its calls blocked by the Cisco router even though call blocking has been enabled, use the **paramspace callsetup after-hours-exempt** command in dial-peer voice configuration mode. To return to the default, use the no form of this command.

paramspace callsetup after-hours-exempt {true | false}

no paramspace callsetup after-hours-exempt

Syntax Description

true	Dial peer is exempt from call-blocking configuration.
false	Dial peer is subject to call-blocking configuration. This is default.

Defaults

All dial peers are subject to call-blocking configuration.

Command Modes

Dial-peer voice configuration

Command History

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

Usage Guidelines

This command is intended to allow H.323 and SIP trunk calls to utilize the voice gateway in spite of the the after-hours configuration in Cisco Unified CME or Cisco Unified SRST.

A Cisco voice gateway (session application) accesses the after-hours call-blocking configuration set by Cisco Unified CME or Cisco Unified SRST and blocks all SCCP, SIP, H.323, and POTS calls that go through the Cisco router regardless of whether the call is from a phone controlled by the Cisco router or from a phone controlled by some other call control application, such as Cisco Unified CallManager.

To disable the After Hours Call Blocking feature for incoming calls from phones other than those registered to a Cisco Unified CME or Cisco Unified SRST router, use this command to exempt an individual H.323, SIP, or POTS dial peer from the call blocking configuration.

To disable the After Hours Call Blocking feature for an individual IP phone registered in Cisco Unified CME or Cisco Unified SRST:

- In Cisco CME 3.4 and later, disable the After Hours Call Blocking feature for a directory number on a SIP phone by using the **after-hour exempt** command in voice register pool or voice register dn configuration mode.
- In Cisco CME 3.0 and later, disable the After Hours Call Blocking feature for an individual SCCP phone by using the **after-hour exempt** command in ephone or ephone-template configuration mode.
- In Cisco SIP SRST 3.4 and later, disable the After Hours Call Blocking feature for SIP phones in a voice register pool by using the **after-hour exempt** command in voice register pool configuration mode.

- In Cisco SRST, you cannot create an exemption for an individual phone from the call-blocking configuration.

Examples

The following example shows how to set the After Hours Call Blocking feature in Cisco Unified CME, and how to configure a particular dial peer (255) so that outgoing calls through this dial peer are exempt from this after-hours call blocking configuration:

```
Router(config)# telephony-service
Router(config-telephony)# after-hours block pattern 1 9011
Router(config-telephony)# exit
Router(config)# dial-peer voice 255 voip
Router(config-dial-peer)# paramspace callsetup after-hours-exempt true
```

Related Commands

Command	Description
after-hour exempt	Specifies that a SCCP phone does not have any of its outgoing calls blocked even though call blocking has been defined.
after-hour exempt (voice register dn)	Specifies that an individual SIP IP phone or a phone extension on a SIP IP phone does not have any of its outgoing calls blocked even though call blocking has been defined.
after-hour exempt (voice register pool)	Specifies that an individual SIP IP phone or phones in a voice register pool does not have any of its outgoing calls blocked even though call blocking has been defined.
after-hours block pattern	Defines a pattern of digits for blocking outgoing calls from IP phones.
after-hours date	Defines a recurring period based on date during which outgoing calls that match defined block patterns are blocked on IP phones.
after-hours day	Defines a recurring period based on day of the week during which outgoing calls that match defined block patterns are blocked on IP phones.

park-slot

To create a floating extension (ephone-dn) at which calls can be temporarily held (parked), use the **park-slot** command in ephone-dn configuration mode. To disable the extension, use the **no** form of this command.

park-slot [**reserved-for** *extension-number*] [**timeout** *seconds* **limit** *count*] [**notify** *extension-number* [**only**]] [**recall**] [**transfer** *extension-number*] [**alternate** *extension-number*] [**retry** *seconds* **limit** *count*]

no park-slot [**reserved-for** *extension-number*] [**timeout** *seconds* **limit** *count*] [**notify** *extension-number* [**only**]] [**recall**] [**transfer** *extension-number*] [**alternate** *extension-number*] [**retry** *seconds* **limit** *count*]

Syntax Description

reserved-for *extension-number*	(Optional) Indicates that this slot is a private park slot for the phone with the specified extension number as its primary line. All lines on that phone can use this park slot.
timeout *seconds*	(Optional) Sets the call-park reminder timeout in seconds. Range is from 0 to 65535. This reminder sends a 1-second ring to the Cisco Unified IP phone that parked the call and displays a message on the LCD panel of all phones in the Cisco Unified CME system, indicating that a call is on hold. By default, the reminder ring is sent only to the phone that parked the call.
limit *count*	(Optional) Sets a limit on the number of reminder or retry timeouts. Range is from 1 to 65535.
notify *extension-number*	(Optional) Sends a reminder ring to the specified extension in addition to the reminder ring that is sent to the phone that parked the call.
only	(Optional) Sends a reminder ring only to the extension specified with the **notify** keyword and does not send a reminder ring to the phone that parked the call. This option allows all reminder rings for parked calls to be sent to a receptionist's phone or an attendant's phone, for example.
recall	Optional) Returns the call to the phone that parked it after the timeout limits expire.
transfer *extension-number*	(Optional) Returns the call to the specified number after the timeout limits expire.
alternate *extension-number*	(Optional) Returns the call to a specified second target number if the recall or transfer target phone is in use on any of its extensions (ringing or in conversation).
retry *seconds*	(Optional) Sets the delay before another attempt to recall or transfer a parked call, in seconds. Range is from 0 to 65535. Number of attempts is set by the **limit** keyword.

Command Default

No call-park slot is defined.

Command Modes

Ephone-dn configuration

Command History

Release	Cisco Product	Modification
12.3(7)T	Cisco CME 3.1	This command was introduced.
12.4(4)XC	Cisco Unified CME 4.0	The **reserved-for**, **recall**, **transfer**, **alternate**, and **retry** keywords were added.
12.4(9)T	Cisco Unified CME 4.0	The **reserved-for**, **recall**, **transfer**, **alternate**, and **retry** keywords were integrated into the Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command creates a call-park slot that is a floating extension, or ephone-dn that is not bound to a physical phone, to which phone users can transfer calls and automatically place them on hold for later pickup from the transferring extension or from another extension. This action is known as “call park.”

At least one call-park slot must be defined using this command before the Park soft key is displayed on phones in a Cisco Unified CME system.

Phone users park calls using the Park soft key. The phone user who parked the call can retrieve the call using the PickUp soft key and an asterisk (*). Other phone users retrieve calls using the PickUp soft key and the extension number of the call-park slot. Calls can also be transferred to a call-park-slot extension number using the Transfer key; a transfer to a call-park slot is always a blind transfer. Calls can also be forwarded to a call-park-slot extension, and callers can directly dial call-park-slot extensions.

When a call that uses a G.711 codec is parked, the caller hears the music-on-hold (MOH) audio stream; otherwise, the caller hears tone on hold.

A reminder ring can be sent to the extension that parked the call by using the **timeout** keyword with the **park-slot** command. The **timeout** keyword and argument set the interval length during which the call-park reminder ring is timed out or inactive. If the **timeout** keyword is not used, no reminder ring is sent to the extension that parked the call. The number of time-out intervals and reminder rings are configured with the **limit** keyword and argument. For example, a limit of 3 timeout intervals sends 2 reminder rings (interval 1, ring 1, interval 2, ring 2, interval 3). The **timeout** and **limit** keywords and arguments also set the maximum time that calls stay parked. For example, a timeout interval of 10 seconds and a limit of 5 timeout intervals (**park-slot timeout 10 limit 5**) will park calls for approximately 50 seconds.

If the **timeout** keyword is not used with this command, no reminder ring is sent to the extension that parked the call. If the **timeout** keyword is used, a reminder ring is sent only to the extension that parked the call unless the **notify** keyword is also used to specify an additional extension number to receive a reminder ring. When an additional extension number is specified using the **notify** keyword, the phone user at that extension can retrieve a call from this slot by pressing the PickUp soft key and an asterisk (*).

Each call-park slot can hold one call at a time, so the number of simultaneous calls that can be parked is equal to the number of slots that have been created in the Cisco Unified CME system. Using the **reserved-for** keyword, you can also create a call-park slot that is dedicated for use by one extension so that extension always has a slot available at which to park a call. With nonreserved slots, multiple call-park slots can be created with the same extension number so that all the calls that are parked for a particular group can be parked at a known extension number. For example, at a hardware store, calls for the plumbing department can be parked at extension 101, calls for lighting can be parked at 102, and so forth. Then, anyone in the plumbing department can pick up calls from extension 101. When multiple calls are parked at the same extension number, they are picked up in the order in which they were parked; that is, the call that has been parked the longest is the first call picked up from that extension number.

IP phones park calls at their dedicated call-park slots using the Park soft key. IP phones can also transfer calls to dedicated call-park slots using the Transfer soft key and a standard or custom feature access code (FAC) for call park. Analog phones transfer calls to dedicated call-park slots using hookflash and a standard or custom FAC for call park. The standard FAC for call park is **6. Custom FACs are created using the **fac** command.

If no dedicated park slot is found anywhere in the Cisco Unified CME system for an ephone-dn attempting to park a call, the system uses the standard call-park procedure; that is, the system searches for a preferred park slot (one with an ephone-dn number that matches the last two digits of the ephone-dn attempting to park the call) and if none is found, uses any available call-park slot.

If a name has been specified for a call-park slot, that name will be displayed rather than an extension number on a recall or transfer of the call.

A parked call can have the following dispositions after its timeouts expire:

- Recall—If you specify that a call should be recalled to the parking phone after the timeout interval expires, the call is always returned to the phone's primary extension number, regardless of which extension on the phone did the parking.
- Transfer—If you specify a transfer target, the call is transferred to the specified number after the timeout intervals expire instead of returning to the primary number of the parking phone.
- Alternate—You can also specify an alternate target extension to which to transfer a parked call if the recall or transfer target is in use. *In use* is defined as either ringing or connected to a call. For example, a call is parked at the dedicated park slot for the phone with the primary extension of 2001. After the timeouts expire, the system attempts to recall the call to extension 2001, but that line is now connected to a different call. The system transfers the call to the alternate target that was specified when the park slot was defined.
- Disconnect—When a timeout limit is set and no other disposition has been specified, a call parked at a call-park slot is disconnected after the number of reminder timeouts has been reached.

Examples

The following example creates a call-park slot with the number 1001. After a call is parked at this number, the system provides 10 reminder rings at intervals of 30 seconds to the extension that parked the call.

```
ephone-dn 45
 number 1001
 park-slot timeout 30 limit 10
```

The following example creates a dedicated call-park slot, 2558, that is reserved for the phone that has the primary extension of 2977. Both extension 2977 and 2976 are on the same phone, so they both can use this slot, which is reserved only for the extensions on that phone. After three timeout intervals of 60 seconds, a parked call is recalled to extension 2977. If extension 2977 is busy, the call is rerouted to extension 3754.

```
ephone-dn 24
 number 2977

ephone-dn 25
 number 2976

ephone-dn 27
 number 3754

ephone-dn 30
 number 2558
 name Park 2977
 park-slot reserved-for 2977 timeout 60 limit 3 recall alternate 3754
```

```
ephone 44
 button 1:24 2:25

ephone 45
 button 1:27
```

Related Commands

Command	Description
ephone-dn	Enters ephone-dn configuration mode.
fac	Enables standard feature access codes (FACs) or creates custom FACs.
number	Associates a telephone or extension number with an extension (ephone-dn).

pattern (voice register dialplan)

To define a dial pattern for a SIP dial plan, use the **pattern** command in voice register dialplan configuration mode. To remove the pattern, use the **no** form of this command.

pattern *tag string* [**button** *button-number*] [**timeout** *seconds*] [**user** {**ip** | **phone**}]

no pattern *tag*

Syntax Description

tag	Number that identifies the dial pattern. Range: 1 to 24.
string	Dial pattern, such as the area code, prefix, and first one or two digits of the telephone number, plus wildcard characters or dots (.) for the remainder of the dialed digits.
button *button-number*	(Optional) Button to which the dial pattern applies.
timeout *seconds*	(Optional) Time, in seconds, that the system waits before dialing the number entered by the user. Range: 0 to 30. To have the number dialed immediately, specify 0. If this parameter is not used, the phone's default interdigit timeout value is used (10 seconds).
user	(Optional) Tag that automatically gets added to the dialed number. Do not use this keyword if Cisco Unified CME is the only SIP call agent.
ip	(Optional) Sets the value of the user tag to IP in the dialed number.
phone	(Optional) Sets the value of the user tag to phone in the dialed number.

Command Default

No pattern is defined.

Command Modes

Voice register dialplan configuration

Command History

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

Usage Guidelines

Use this command to define a pattern of dialed digits that are matched by the phone and passed to Cisco Unified CME to initiate a call. Dial strings that match the pattern trigger the sending of a SIP INVITE message to Cisco Unified CME. Patterns are matched sequentially in order of the *tag* number.

You must first use the **type** command to specify the type of phone that the dial plan is being defined for before you can enter a pattern. Enter this command for each dial pattern that is part of the dial plan definition. After you define a dial plan, assign it to a SIP phone by using the **dialplan** command.

The **button** keyword specifies the button to which the dial pattern applies. If the user is initiating a call on line button 1, only the dial patterns specified for button 1 apply. If this keyword is not configured, the dial pattern applies to all lines on the phone. This keyword is not supported on Cisco Unified IP Phones 7905 or 7912. The button number corresponds to the order of the buttons on the side of the screen, from top to bottom, with 1 being the top button.

The **pattern** command and **filename** command are mutually exclusive. You can use either the **pattern** command to define dial patterns manually for a dial plan, or the **filename** command to select a custom dial pattern file that is loaded in system flash.

Examples

The following example shows the dial patterns set for SIP dial plan 10:

```
Router(config)# voice register dialplan 10
Router(config-register-dialplan)# type 7905-7912
Router(config-register-dialplan)# pattern 52...
Router(config-register-dialplan)# pattern 91.......
```

Related Commands

Command	Description
dialplan	Assigns a dial plan to a SIP phone.
filename	Specifies a custom configuration file that contains dial patterns to use for the SIP dial plan.
show voice register dialplan	Displays all configuration information for a specific SIP dial plan.
type (voice register dialplan)	Defines a phone type for a SIP dial plan.

pattern direct

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

pattern direct *tag1* {**CDN** | **CGN** | **FDN**} [*tag2* {**CDN** | **CGN** | **FDN**}]
[*tag3* {**CDN** | **CGN** | **FDN**}] [*last-tag*]

no pattern direct

Syntax Description

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

Defaults

This feature is disabled.

Command Modes

Voice-mail integration configuration

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.2(2)XT	2.0	This command was introduced for Cisco IOS Telephony Services on the Cisco 1750, Cisco 1751, Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(8)T	2.0	This command was integrated for Cisco IOS Telephony Services into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	2.0	This command was implemented for Cisco IOS Telephony Services on the Cisco 2600XM and Cisco 2691.
12.2(11)T	2.01	This command was integrated for Cisco IOS Telephony Services into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760.

Usage Guidelines

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

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

Examples

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

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

Related Commands

Command	Description
pattern ext-to-ext busy	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an internal extension fails to connect to a busy extension and the call is forwarded to voice mail.
pattern ext-to-ext no-answer	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an internal extension attempts to connect to an extension that does not answer and the call is forwarded to voice mail.
pattern trunk-to-ext busy	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an external trunk call reaches a busy extension and the call is forwarded to voice mail.
pattern trunk-to-ext no-answer	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when an external trunk call reaches an unanswered extension and the call is forwarded to voice mail.
vm-integration	Enters voice-mail integration configuration mode and enables voice-mail integration with DTMF and an analog voice-mail systems.

pattern ext-to-ext busy

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

pattern ext-to-ext busy *tag1* {**CDN** | **CGN** | **FDN**} [*tag2* {**CDN** | **CGN** | **FDN**}] [*tag3* {**CDN** | **CGN** | **FDN**}] [*last-tag*]

no pattern ext-to-ext busy

Syntax Description

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

Defaults

This feature is disabled.

Command Modes

Voice-mail integration configuration

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.2(2)XT	2.0	This command was introduced for Cisco IOS Telephony Services on the Cisco 1750, Cisco 1751, Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(8)T	2.0	This command was integrated for Cisco IOS Telephony Services into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	2.0	This command was implemented for Cisco IOS Telephony Services on the Cisco 2600XM and Cisco 2691.
12.2(11)T	2.01	This command was integrated for Cisco IOS Telephony Services into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760.
12.2(13)T	2.02	This command was implemented in Cisco Survivable Remote Site Telephony, Version 2.02.

Usage Guidelines

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

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

Examples

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

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

Related Commands

Command	Description
pattern direct	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when the user presses the Messages button on the phone.
pattern ext-to-ext no-answer	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an internal extension fails to connect to an extension that does not answer and the call is forwarded to voice mail.
pattern trunk-to-ext busy	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an external trunk call reaches a busy extension and the call is forwarded to voice mail.
pattern trunk-to-ext no-answer	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when an external trunk call reaches an unanswered extension and the call is forwarded to voice mail.
vm-integration	Enters voice-mail integration configuration mode and enables voice-mail integration with DTMF and an analog voice-mail systems.

pattern ext-to-ext no-answer

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

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

no pattern ext-to-ext no-answer

Syntax Description

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

Defaults

This feature is disabled.

Command Modes

Voice-mail integration configuration

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.2(2)XT	2.0	This command was introduced for Cisco IOS Telephony Services on the Cisco 1750, Cisco 1751, Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(8)T	2.0	This command was integrated for Cisco IOS Telephony Services into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	2.0	This command was implemented for Cisco IOS Telephony Services on the Cisco 2600XM and Cisco 2691.
12.2(11)T	2.01	This command was integrated for Cisco IOS Telephony Services into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760.
12.2(13)T	2.02	This command was implemented in Cisco Survivable Remote Site Telephony, Version 2.02.

Usage Guidelines

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

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

Examples

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

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

Related Commands

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

pattern trunk-to-ext busy

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

pattern trunk-to-ext busy *tag1* {**CDN** | **CGN** | **FDN**} [*tag2* {**CDN** | **CGN** | **FDN**}] [*tag3* {**CDN** | **CGN** | **FDN**}] [*last-tag*]

no pattern trunk-to-ext busy

Syntax Description

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

Defaults

This feature is disabled by default.

Command Modes

Voice-mail integration configuration

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.2(2)XT	2.0	This command was introduced for Cisco IOS Telephony Services on the Cisco 1750, Cisco 1751, Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(8)T	2.0	This command was integrated for Cisco IOS Telephony Services into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	2.0	This command was implemented for Cisco IOS Telephony Services on the Cisco 2600XM and Cisco 2691.
12.2(11)T	2.01	This command was integrated for Cisco IOS Telephony Services into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760.
12.2(13)T	2.02	This command was implemented in Cisco Survivable Remote Site Telephony, Version 2.02.

Usage Guidelines

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

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

Examples

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

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

Related Commands

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

pattern trunk-to-ext no-answer

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

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

no pattern trunk-to-ext no-answer

Syntax Description

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

Defaults

This feature is disabled.

Command Modes

Voice-mail integration configuration

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.2(2)XT	2.0	This command was introduced for Cisco IOS Telephony Services on the Cisco 1750, Cisco 1751, Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(8)T	2.0	This command was integrated for Cisco IOS Telephony Services into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	2.0	This command was implemented for Cisco IOS Telephony Services on the Cisco 2600XM and Cisco 2691.
12.2(11)T	2.01	This command was integrated for Cisco IOS Telephony Services into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760.
12.2(13)T	2.02	This command was implemented in Cisco Survivable Remote Site Telephony, Version 2.02.

Usage Guidelines

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

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

Examples

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

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

Related Commands

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

phone-key-size

To specify the size of the RSA key pair that is generated on phones, use the **phone-key-size** command in CAPF-server configuration mode. To return the size to the default, use the **no** form of this command.

phone-key-size {**512** | **1024** | **2048**}

no phone-key-size

Syntax Description

512	512 bits
1024	1024 bits. This is the default key size.
2048	2048 bits

Command Default

RSA key pair size is 1024.

Command Modes

CAPF-server configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used with Cisco Unified CME phone authentication.

If you choose a higher key size than the default setting, the phones take longer to generate the entropy that is required to generate the keys. Key generation, which is set at low priority, allows the phone to function while the action occurs. Depending on the phone model, you may notice that key generation takes up to 30 or more minutes to complete.

Examples

The following example specifies a key size of 2048 bits.

```
Router(config)# capf-server
Router(config-capf-server)# source address 10.10.10.1
Router(config-capf-server)# trustpoint-label server25
Router(config-capf-server)# cert-oper upgrade all
Router(config-capf-server)# cert-enroll-trustpoint server12 password 0 x8oWiet
Router(config-capf-server)# auth-mode auth-string
Router(config-capf-server)# auth-string generate all
Router(config-capf-server)# port 3000
Router(config-capf-server)# keygen-retry 5
Router(config-capf-server)# keygen-timeout 45
Router(config-capf-server)# phone-key-size 2048
```

phone-redirect-limit (voice register global)

To set the number of 3XX responses an originating SIP phone in a Cisco CallManager Express (Cisco CME) system can accept for a single cal., use the **phone-redirect-limit** command in voice register global configuration mode. To revert to the default, use the **no** form of this command.

phone-redirect-limit *number*

no phone-redirect-limit

Syntax Description

number	Maximum number of 3XX responses accepted for a single call. Range is 5 to 20. Default is 5.

Defaults

5

Command Modes

Voice register global configuration

Command History

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

Usage Guidelines

Use this command to control how many subsequent 3XX responses an originating SIP phone can handle for a single call. The terminating side is any forwarding party which does not use B2BUA, but sends 3XX directly to the originating calling phone. When Cisco CME gets a 3XX from the terminating side, Cisco CME relays the 3XX to the originating SIP phone. The default number of 3XXs that the originating phone can accept is 5.

The following example shows how to set the maximum number of redirects to 6:

```
Router(config)# voice register global
Router(config-register-global)# phone-redirect-limit 6
```

Related Commands

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

pickup-group

To assign an extension (ephone-dn) to a Cisco Unified CME call-pickup group, use the **pickup-group** command in ephone-dn or ephone-dn-template configuration mode. To remove the extension from the group, use the **no** form of this command.

pickup-group *number*

no pickup-group

Syntax Description

number	Digit string representing a pickup group number. The string can contain a maximum of 32 digits.

Command Default

An extension does not belong to any pickup group.

Command Modes

Ephone-dn configuration
Ephone-dn-template configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.4(4)XC	Cisco Unified CME 4.0	This command was made available in ephone-dn-template configuration mode.
12.4(9)T	Cisco Unified CME 4.0	This command in ephone-dn-template configuration mode was integrated into Cisco IOS Release1 12.4(9)T.

Usage Guidelines

This command allows administrators to assign an individual ephone-dn to a call-pickup group. Phone users can pick up ringing calls within their own pickup group more easily than calls outside their group.

Each ephone-dn can be assigned to a maximum of one pickup group.

Pickup group numbers may be of varying length, but their leading digits must be unique. For example, you cannot define both pickup group 17 and pickup group 177 in the same Cisco Unified CME system, because a pickup in group 17 will always be triggered before the user can enter the final 7 for group 177. You can, however, define pickup groups 27 and 177 in the same Cisco Unified CME system.

There is no limit to the number of ephone-dns that can be assigned to a single pickup group, and there is no limit to the number of pickup groups that can be defined in a Cisco Unified CME system.

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

Examples

The following example assigns extension 3242 to pickup group 25:

```
Router(config)# ephone-dn 4
Router(config-ephone-dn)# number 3242
Router(config-ephone-dn)# pickup-group 25
```

The following example uses an ephone-dn-template to assign extension 3242 to pickup group 25:

```
Router(config)# ephone-dn-template 8
Router(config-ephone-dn-template)# pickup-group 25
Router(config-ephone-dn-template)# exit
Router(config)# ephone-dn 4
Router(config-ephone-dn)# number 3242
Router(config-ephone-dn)# ephone-dn-template 8
```

Related Commands

Command	Description
ephone-dn	Enters ephone-dn configuration mode.
ephone-dn-template	Enters ephone-dn-template configuration mode.

pilot

To define the ephone-dn that callers dial to reach a Cisco CallManager Express (Cisco CME) ephone hunt group, use the **pilot** command in ephone-hunt configuration mode. To remove the pilot number from the ephone hunt group, use the **no** form of this command.

pilot *number* [**secondary** *number*]

no pilot *number* [**secondary** *number*]

Syntax Description

number	String of up to 27 characters that represents an E.164 telephone number. Normally the string is composed of digits, but the string may contain alphabetic characters if the number is dialed only by the router, as with an intercom number, or is not intended to be dialed at all. Secondary numbers can contain wildcards in the string. For details, see "Usage Guidelines."
secondary	(Optional) Defines the number that follows as an additional pilot number for the ephone hunt group.

Defaults

No pilot number is defined.

Command Modes

Ephone-hunt configuration

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.2(15)ZJ	3.0	This command was introduced.
12.3(4)T	3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.3(7)T	3.1	The **secondary** *secondary-number* keyword-argument pair was introduced.

Usage Guidelines

This command defines a valid number for an ephone-dn (extension) that is to be assigned to an ephone hunt pilot group. The dial-plan pattern can be applied to the pilot number.

The **secondary** keyword allows you to associate a second telephone number with this ephone-dn so that the hunt group can be called by dialing either the main or secondary phone number. The secondary number may contain one or more wildcards instead of digits, even if the wildcard number overlaps the primary number. For example, 50.. (the number 50 followed by periods, which stand for wildcards) matches all four-digit extensions that start with 50. Wildcard characters cannot be used in the primary pilot number.

Alphabetic characters can be used to create a primary or secondary pilot number that cannot be dialed from a phone and is not part of the dial plan.

Examples

The following example sets the pilot number to 2345 for peer ephone hunt group number 5:

```
ephone-hunt 5 peer
 pilot 2345
 list 2346, 2347, 2348
 hops 3
 timeout 45
 final 6000
```

The following example sets the pilot number for ephone hunt group 3 to 2222 and the secondary pilot number to 4444:

```
ephone-hunt 3 sequential
 pilot 2222 secondary 4444
 list 2555, 2556, 2557
 final 6000
```

The following example uses wildcards in the secondary pilot number to create a hunt group that receives the calls made to all numbers that start with 555. The primary pilot number, A0, cannot be dialed.

```
ephone-hunt 1 longest-idle
 pilot A0 secondary 555....
 list 1000, 1001, 1002
 timeout 5
 hops 3
 final 1100
```

Related Commands

Command	Description
ephone-hunt	Enters ephone-hunt configuration mode to define a Cisco CME ephone hunt group.
final	Defines the last ephone-dn in an ephone hunt group.
hops	Defines the number of times that a call is redirected to the next ephone-dn in a peer ephone-hunt-group list before proceeding to the final ephone-dn.
list	Lists the ephone-dns that participate in an ephone hunt group.
max-redirect	Changes the current number of allowable redirects in a Cisco CME system.
no-reg (ephone-hunt)	Specifies that the pilot number of an ephone hunt group should not register with the H.323 gatekeeper.
preference (ephone-hunt)	Sets preference order for the ephone-dn associated with an ephone-hunt-group pilot number.
timeout (ephone-hunt)	Sets the number of seconds after which a call that is not answered is redirected to the next number in the ephone-hunt-group list.

pilot (voice hunt-group)

To define the voice dn that callers dial to reach a Cisco CallManager Express (Cisco CME) voice hunt group, use the **pilot** command in voice hunt-group configuration mode. To remove the pilot number from the voice hunt group, use the **no** form of this command.

pilot *number* [**secondary** *number*]

no pilot *number* [**secondary** *number*]

Syntax Description

number	String of up to 32 characters that represents an E.164 telephone number. Normally the string is composed of digits, but the string may contain alphabetic characters if the number is dialed only by the router, as with an intercom number, or is not intended to be dialed at all. Secondary numbers can contain wild cards in the string. For details, see "Usage Guidelines."
secondary	(Optional) Defines the number that follows as an additional pilot number for the voice hunt group.

Defaults

No pilot number is defined

Command Modes

Voice hunt-group configuration

Command History

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

Usage Guidelines

This command defines a valid number for an voice dn (extension) that is to be assigned to an voice hunt pilot group. The dial-plan pattern can be applied to the pilot number.

The **secondary** keyword allows you to associate a second telephone number with this voice dn so that the hunt group can be called by dialing either the main or secondary phone number. The secondary number may contain one or more wild cards instead of digits, even if the wildcard number overlaps the primary number. For example, 50.. (the number 50 followed by periods, which stand for wild card) matches all four-digit extensions that start with 50. Wildcard characters cannot be used in the primary pilot number.

Alphabetic characters can be used to create a primary or secondary pilot number that cannot be dialed from a phone and is not part of the dial plan.

Examples

The following example shows how to set the pilot number to 2345 for voice hunt group hunt group number 5:

```
voice-hunt 5 peer
 pilot 2345
 list 2346, 2347, 2348
 hops 3
```

```
 timeout 45
 final 6000
```

The following example shows how to set the pilot number for voice hunt group 3 to 2222 and the secondary pilot number to 4444:

```
voice hunt-group 3 sequential
 pilot 2222 secondary 4444
 final 6000
```

The following example shows how to use wild cards in the secondary pilot number to create a voice hunt group that receives the calls made to all numbers that start with 55501. The primary pilot number, A0, cannot be dialed.

```
voice hunt-group 1 longest-idle
 pilot A0 secondary 55501..
 list 1000, 1001, 1002
 timeout 5
 hops 3
 final 1100
```

Related Commands

Command	Description
final (voice hunt-group)	Defines the last extension in a voice hunt group.
hops (voice hunt-group)	Defines the number of times that a call is redirected to the next directory number in a peer voice hunt-group list before proceeding to the final directory number.
list (voice hunt-group)	Defines the directory numbers that participate in a hunt group.
voice hunt-group	Defines the type of hunt group.

pin

To set a personal identification number (PIN) for an IP phone in a Cisco CallManager Express (Cisco CME) system, use the **pin** command in ephone configuration mode. To remove a PIN, use the **no** form of this command.

pin *number*

no pin

Syntax Description

number	PIN that will be used to log in to a Cisco IP phone. This is a numeric string from four to eight digits in length.

Defaults

No PIN is set.

Command Modes

Ephone configuration

Command History

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

Usage Guidelines

The **pin** command allows individual phone users to override call-blocking patterns that are associated with defined time periods. Call-blocking patterns that are in effect at all times (7 days a week, 24 hours a day) cannot be overridden using a PIN.

Call blocking on IP phones is defined in the following way. First, one or more patterns of outgoing digits to be blocked are defined using the **after-hours block pattern** command. Next, one or more time periods during which calls to those patterns are to be blocked are defined using the **after-hours date** or **after-hours day** command or both. By default, all IP phones in a Cisco CME system are restricted if at least one pattern and at least one time period are defined. Individual phones can be completely exempted from call blocking using the **after-hour exempt** command. An individual with a PIN can override call blocking by entering the PIN after pressing the Login soft key to log in to a phone that has been configured for that PIN using the **pin** command.

The PIN functionality applies only to IP phones that have soft keys, such as the Cisco IP Phones 7940 and 7940G and the Cisco IP Phones 7960 and 7960G.

Examples

The following example sets a PIN for an IP phone:

```
Router(config)# ephone 1
Router(config-ephone)# pin 1000
```

Related Commands

Command	Description
after-hour exempt	Specifies that an IP phone does not have any of its outgoing calls blocked even though call blocking has been defined for a Cisco CME system.
after-hours block pattern	Defines a pattern of digits to be blocked for outgoing calls from IP phones.
after-hours date	Defines a recurring period based on month and day during which outgoing calls that match defined call-block patterns are blocked on IP phones.
after-hours day	Defines a recurring period based on day of the week during which outgoing calls that match defined call-block patterns are blocked on IP phones.
ephone	Enters the Ethernet phone (ephone) configuration mode.
login	Defines when IP phones in a Cisco CME system are logged out automatically.
show ephone login	Displays the login states of all phones.

pin (voice logout-profile and voice user-profile)

To configured a personal identification number (PIN) for accessing a particular IP phone that is enabled for extension mobility, use the **pin** command in voice logout-profile or voice user-profile configuration mode. To remove a PIN, use the no form of this command.

pin *number*

no pin *number*

Syntax Description

number	Four to eight digits numeric string for accessing Cisco Unified IP phone.

Command Default

No PIN is configured.

Command Modes

Voice logout-profile configuration (config-logout-profile)
Voice user-profile configuration (config-user-profile)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XW1	Cisco Unified CME 4.2	This command was introduced.

Usage Guidelines

Use this command in voice logout-profile configuration mode to create a PIN to be used by a phone user to disable the call blocking configuration for a Cisco Unified IP phone on which a logout profile is downloaded.

Use this command in voice user-profile configuration mode to create a PIN to be used by a phone user to disable the call blocking configuration for a Cisco Unified IP phone on which a user profile is downloaded.

PIN functionality applies only to IP phones that have soft keys, such as the Cisco Unified IP Phone 7940 and 7940G.

Examples

The following example shows the configuration for a user profile to be downloaded when the a phone user logs into a Cisco Unified IP phone that is enabled for extension mobility, including a PIN of 12345:

```
pin 12345
user me password pass123
number 2001 type silent-ring
number 2002 type beep-ring
number 2003 type feature-ring
number 2004 type monitor-ring
number 2005,2006 type overlay
number 2007,2008 type cw-overly
speed-dial 1 3001
speed-dial 2 3002 blf
```

port (CAPF-server)

To define the TCP port number on which the CAPF server listens for incoming socket connections, use the **port** command in CAPF-server configuration mode. To use the default, use the **no** form of this command.

port *tcp-port*

no port

Syntax Description

tcp-port	Port for secure communication. Range is from 2000 to 9999. Default is 3804.

Command Default

TCP port number 3804.

Command Modes

CAPF-server configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used with Cisco Unified CME phone authentication.

Examples

The following example specifies TCP port 3000 instead of the default port 3804:

```
Router(config)# capf-server
Router(config-capf-server)# source address 10.10.10.1
Router(config-capf-server)# trustpoint-label server25
Router(config-capf-server)# cert-oper upgrade all
Router(config-capf-server)# cert-enroll-trustpoint server12 password 0 x8oWiet
Router(config-capf-server)# auth-mode auth-string
Router(config-capf-server)# auth-string generate all
Router(config-capf-server)# port 3000
Router(config-capf-server)# keygen-retry 5
Router(config-capf-server)# keygen-timeout 45
Router(config-capf-server)# phone-key-size 2048
```

preference (ephone-dn)

To set dial-peer preference order for an extension (ephone-dn) associated with a Cisco IP phone, use the **preference** command in ephone-dn configuration mode. To reset the preference order to the default, use the **no** form of this command.

preference *preference-order* [**secondary** *secondary-order*]

no preference

Syntax Description

preference-order	Preference order for the primary number associated with an extension (ephone-dn). Refer to CLI help for a range of numeric options, where 0 is the highest preference. Default is 0.
secondary *secondary-order*	(Optional) Preference order for the secondary number associated with the ephone-dn. Refer to CLI help for a range of numeric options, where 0 is the highest preference. Default is 9.

Defaults

preference-order: 0 (highest preference)

secondary-order: 9 (lowest preference)

Command Modes

Ephone-dn configuration

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.1(5)YD	1.0	This command was introduced on the following platforms: Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(2)XT	2.0	This command was implemented on the Cisco 1750 and Cisco 1751.
12.2(8)T	2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	2.0	This command was implemented on the Cisco 2600XM and Cisco 2691.
12.2(11)T	2.01	This command was implemented on the Cisco 1760.
12.2(15)ZJ	3.0	The **secondary** *secondary-order* keyword-argument pair was introduced.
12.3(4)T	3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Usage Guidelines

When you create an ephone-dn for an IP phone in a Cisco CallManager Express (Cisco CME) system, you automatically create a virtual voice port and one to four virtual dial peers to be used by that ephone-dn. This command sets a preference value for the primary and secondary numbers that are associated with the ephone-dn that you are creating. The preference values are passed transparently into the dial peer or dial peers created by the ephone-dn. The preference values allow you to control the selection of a desired dial peer when multiple dial peers are matched on the same destination-pattern (target) number value. In this way, the **preference** command can be used to establish a hunt strategy for incoming calls.

The **huntstop** command can be used to prevent further hunting for a dial-peer match when an ephone-dn is busy or does not answer.

Examples

The following example sets a preference of 2 for the directory number 3000:

```
ephone-dn 1
 number 3000
 preference 2
```

In the following example, the number 1222 under ephone-dn 4 has a higher preference than the number 1222 under ephone-dn 5.

```
ephone-dn 4
 number 1222
 preference 0
!
!
ephone-dn 5
 number 1222
 preference 1
```

The following example shows an ephone-dn with two numbers. The primary number has a higher preference than the secondary number.

```
ephone-dn 6
 number 2233 secondary 2234
 preference 0 secondary 1
```

Related Commands

Command	Description
ephone-dn	Enters ephone-dn configuration mode.
huntstop	Discontinues call hunting behavior for an extension (ephone-dn) or an extension channel.

preference (ephone-hunt)

To set preference order for the ephone-dn associated with a Cisco CallManager Express (Cisco CME) ephone-hunt-group pilot number, use the **preference** command in ephone-hunt configuration mode. To delete this preference order, use the **no** form of this command.

preference *preference-order* [**secondary** *secondary-order*]

no preference *preference-order* [**secondary** *secondary-order*]

Syntax Description

preference-order	Preference order. Range is from 0 to 8, where 0 is the highest preference and 8 is the lowest preference. Default is 0.
secondary *secondary-order*	(Optional) Preference order for the secondary pilot number. Range is from 0 to 10, where 0 is the highest preference and 10 is the lowest preference. Default is 9.

Defaults

preference-order: 0 (highest preference)

secondary-order: 9 (lowest preference)

Command Modes

Ephone-hunt configuration

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.2(15)ZJ	3.0	This command was introduced.
12.3(4)T	3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.3(7)T	3.1	The **secondary** *secondary-order* keyword-argument pair was introduced.

Usage Guidelines

This command sets a preference value that is used for matching dial peers in a Cisco IP phone virtual dial-peer group. The preference value is associated with a pilot number for a Cisco CME ephone hunt group. The preference value is passed transparently into the dial peer created by the pilot number. Setting the preference enables the desired dial peer to be selected when multiple dial peers within a hunt group are matched for a dial string.

Examples

The following example sets the preference for the pilot number of hunt group 23 to 1:

```
Router(config)# ephone-hunt 23 sequential
Router(config-ephone-hunt)# pilot 2355
Router(config-ephone-hunt)# preference 1
```

Related Commands

Command	Description
ephone-hunt	Defines an ephone hunt group and enters ephone-hunt configuration mode.
final	Defines the last ephone-dn in an ephone hunt group.
hops	Defines the number of times that a call is redirected to the next ephone-dn in a peer ephone-hunt-group list before proceeding to the final ephone-dn.
list	Lists the ephone-dns that participate in an ephone hunt group.
max-redirect	Changes the current number of allowable redirects in an Cisco CME system.
no-reg (ephone-hunt)	Specifies that the pilot number of an ephone hunt group not register with the H.323 gatekeeper.
pilot	Defines the ephone-dn that callers dial to reach an ephone hunt group.
timeout (ephone-hunt)	Sets the number of seconds after which a call that is not answered is redirected to the next number in the ephone-hunt-group list.

preference (voice hunt-group)

To set preference order for the voice dial peer associated with a Cisco CallManager Express (Cisco CME) voice hunt-group pilot number, use the **preference** command in voice hunt-group configuration mode. To delete this preference order, use the **no** form of this command.

preference *preference-order* [**secondary** *secondary-order*]

no preference *preference-order* [**secondary** *secondary-order*]

Syntax Description

preference-order	Preference order for the extension or telephone number associated with a dial peer. Range is 0 to 8. Default is 0.
secondary *secondary-order*	(Optional) Preference order for the secondary pilot number. Range is from 0 to 10, where 0 is the highest preference and 10 is the lowest preference. Default is 9.

Defaults

0 (highest preference)

Command Modes

Voice hunt-group configuration

Command History

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

Usage Guidelines

This command sets a preference value that is used for matching dial peers in a Cisco IP phone virtual dial-peer group. The preference value is associated with a pilot number for a Cisco CME voice hunt group. The preference value is passed transparently into the dial peer created by the pilot number. Setting the preference enables the desired dial peer to be selected when multiple dial peers within a hunt group are matched for a dial string.

Note It is recommended that the parallel hunt-group pilot number be unique in the system. Parallel hunt groups may not work if there are more than one partial or exact dial-peer match. For example, this happens if the pilot number is "8000" and there is another dial peer that matches "8…". If multiple matches cannot be avoided, give call parallel hunt group the highest priority to run by assigning a lower preference to the other dial peers. Note that 10 is the lowest preference value. By default, dial peers created by parallel hunt groups have a preference of 0.

Examples

The following is an example of a parallel voice hunt group. The pilot number is 6000 and the preference assigned to the pilot number is 1:

```
voice hunt-group 2 parallel
 pilot 6000
 preference 1
 list 3000, 3010, 3020
```

```
final 9999
timeout 10
```

Related Commands

Command	Description
pilot (voice hunt-group)	Defines the voice dn that callers dial to reach a Cisco CallManager Express (Cisco CME) voice hunt group.
voice hunt-group	Defines the type of hunt group.

preference (voice register dn)

To set the dial-peer preference order for VoIP dial peer to be created for a directory number on a SIP phone, use the **preference** command in voice register dn configuration mode. To reset the preference order to the default, use the **no** form of this command.

preference *preference-order*

no preference

Syntax Description

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

Defaults

0 (highest preference)

Command Modes

Voice register dn configuration

Command History

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

Usage Guidelines

When you create a directory number for a SIP phone in a Cisco CallManager Express (Cisco CME) or Cisco SIP SRST environment, you automatically create a virtual voice port and one to four virtual dial peers to be used by that directory number. This command sets a preference value for the extension or telephone number that is associated with the directory number hat you are creating. The preference value is passed transparently to dial peers created by the directory number. The preference value allows you to control the selection of a desired dial peer when multiple dial peers are matched on the same destination pattern (extension or telephone number). In this way, the **preference** command can be used to establish a hunt strategy for incoming calls.

The **huntstop** command can be used to prevent further hunting for a dial-peer match when a number is busy or does not answer.

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

Examples

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

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

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

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

Related Commands

Command	Description
huntstop (voice register dn)	Discontinues call hunting behavior for an extension (directory number) or an extension channel.
voice register dn	Enters voice register dn configuration mode to define an extension for a SIP phone line.

preference (voice register pool)

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

preference *preference-order*

no preference

Syntax Description

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

Command Default

0 (highest preference order)

Command Modes

Voice register pool configuration

Command History

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

Usage Guidelines

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

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

Examples

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

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

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

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

Related Commands

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

presence

To enable presence service and enter presence configuration mode, use the **presence** command in global configuration mode. To disable presence service, use the **no** form of this command.

presence

no presence

Syntax Description

This command has no arguments or keywords.

Command Default

Presence service is disabled.

Command Modes

Global configuration

Command History

Release	Modification
12.4(11)XJ	This command was introduced.
12.4(15)T	This command was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

This command enables the router to perform the following presence functions:

- Process presence requests from internal lines to internal lines. Notify internal subscribers of any status change.
- Process incoming presence requests from a SIP trunk for internal lines. Notify external subscribers of any status change.
- Send presence requests to external presentities on behalf of internal lines. Relay status responses to internal lines.

Examples

The following example shows how to enable presence and enter presence configuration mode to set the maximum subscriptions to 150:

```
Router(config)# presence
Router(config-presence)# max-subscription 150
```

Related Commands

Command	Description
allow watch	Allows a directory number on a phone registered to Cisco Unified CME to be watched in a presence service.
debug presence	Displays debugging information about the presence service.
max-subscription	Sets the maximum number of concurrent watch sessions that are allowed.
presence enable	Allows the router to accept incoming presence requests.

Command	Description
server	Specifies the IP address of a presence server for sending presence requests from internal watchers to external presence entities.
show presence global	Displays configuration information about the presence service.
show presence subscription	Displays information about active presence subscriptions.

presence call-list

To enable Busy Lamp Field (BLF) monitoring for call lists and directories on phones registered to the Cisco Unified CME router, use the **presence call-list** command in ephone, presence, or voice register pool configuration mode. To disable BLF indicators for call lists, use the **no** form of this command.

presence call-list

no presence call-list

Syntax Description

This command has no arguments or keywords.

Command Default

BLF monitoring for call lists is disabled.

Command Modes

Ephone configuration
Presence configuration
Voice register pool configuration

Command History

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

Usage Guidelines

This command enables a phone to monitor the line status of directory numbers listed in a directory or call list, such as a missed calls, placed calls, or received calls list. Using this command in presence mode enables the BLF call-list feature for all phones. To enable the feature for an individual SCCP phone, use this command in ephone configuration mode. To enable the feature for an individual SIP phone, use this command in voice register pool configuration mode.

If this command is disabled globally and enabled in voice register pool or ephone configuration mode, the feature is enabled for that voice register pool or ephone.

If this command is enabled globally, the feature is enabled for all voice register pools and ephones regardless of whether it is enabled or disabled on a specific voice register pool or ephone.

To display a BLF status indicator, the directory number associated with a telephone number or extension must have presence enabled with the **allow watch** command.

For information on the BLF status indicators that display on specific types of phones, see the Cisco Unified IP Phone documentation for your phone model.

Examples

The following example shows the BLF call-list feature enabled for ephone 1. The line status of a directory number that appears in a call list or directory is displayed on phone 1 if the directory number has presence enabled.

```
Router(config)# ephone 1
Router(config-ephone)# presence call-list
```

Related Commands

Command	Description
allow watch	Allows a directory number on a phone registered to Cisco Unified CME to be watched in a presence service.
blf-speed-dial	Enables BLF monitoring for a speed-dial number on a phone registered to Cisco Unified CME.
presence	Enables presence service and enters presence configuration mode.
show presence global	Displays configuration information about the presence service.

presence enable

To allow incoming presence requests, use the **presence enable** command in SIP user-agent configuration mode. To block incoming requests, use the **no** form of this command.

presence enable

no presence enable

Syntax Description

This command has no arguments or keywords.

Command Default

Incoming presence requests are blocked.

Command Modes

SIP user-agent configuration

Command History

Release	Modification
12.4(11)XJ	This command was introduced.
12.4(15)T	This command was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

This command allows the router to accept incoming presence requests (SUBSCRIBE messages) from internal watchers and SIP trunks. It does not impact outgoing presence requests.

Examples

The following example shows how to allow incoming presence requests:

```
Router(config)# sip-ua
Router(config-sip-ua)# presence enable
```

Related Commands

Command	Description
allow subscribe	Allows internal watchers to monitor external presence entities (directory numbers).
allow watch	Allows a directory number on a phone registered to Cisco Unified CME to be watched in a presence service.
max-subscription	Sets the maximum number of concurrent watch sessions that are allowed.
show presence global	Displays configuration information about the presence service.
show presence subscription	Displays information about active presence subscriptions.
watcher all	Allows external watchers to monitor internal presence entities (directory numbers).

present-call

To present ephone-hunt-group calls only to member phones that are idle or onhook, use the **present-call** command in ephone-hunt configuration mode. To return to the default, use the **no** form of this command.

present-call {**idle-phone** | **onhook-phone**}

no present-call {**idle-phone** | **onhook-phone**}

Syntax Description

idle-phone	Presents calls from the ephone-hunt group only if all lines are idle on the phone on which the hunt-group line appears. This option does not consider monitored lines that have been configured on the phone using the **button m** command.
onhook-phone	Presents calls from the ephone-hunt group only if the phone on which the number appears is in the onhook state. When this keyword is configured, calls in the ringing or hold state that are unrelated to the hunt group do not prevent the presentation of calls from the ephone-hunt group.

Command Default

Ephone hunt group calls are presented to lines (ephone-dns) that are not in use, regardless of the state of other lines on the same ephone.

Command Modes

Ephone-hunt configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

If you do not use this command, an ephone hunt group presents calls to an ephone whenever the phone line (ephone-dn) that corresponds to a number in an ephone-hunt list is available. The status of other phone lines on the phone is not considered.

The **present-call** command adds additional controls that allow you to take into account the activity on all lines of a phone that has an ephone-dn that is assigned to an ephone hunt group. The **present-call** command allows you to specify that hunt groups should present calls to these phones only when they are on hook or are not busy with an active call. This keeps hunt group calls from possibly going unanswered because a phone is occupied with a call on a line other than the line assigned to the hunt group.

Examples

The following example sets up a peer hunt group with three ephone-dns to answer calls. Incoming calls are sent only to ephone-dns on phones that are on-hook.

```
ephone-hunt 17 peer
pilot 3000
list 3011, 3021, 3031
```

```
hops 3
final 7600
present-call onhook-phone
```

Related Commands

Command	Description
ephone-hunt	Defines an ephone hunt group and enters ephone-hunt configuration mode.

provision-tag

To create a provision tag for identifying an ephone configuration for the extension assigner application, use the **provision-tag** command in ephone configuration mode. To remove the provision tag, use the **no** form of this command.

provision-tag *tag*

no provision-tag *tag*

Syntax Description

tag	Unique number that identifies this provision tag. Range: 1 to 2147483647.

Command Default

No provision tag is created.

Command Modes

Ephone configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC4	Cisco Unified CME 4.0(3)	This command was introduced.
12.4(11)XJ	Cisco Unified CME 4.1	This command was introduced.
12.4(15)T	Cisco Unified CME 4.1	This command was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

This command creates a provision tag.

A provision tag enables you to use some number other than an ephone tag, such as a jack number or an extension number, to identify an ephone configuration. The provision tag can be used with the extension assigner application to assign the corresponding ephone configuration to an IP phone.

This command is ignored unless you also use the **extension-assigner tag-type** command with the **provision-tag** keyword.

Examples

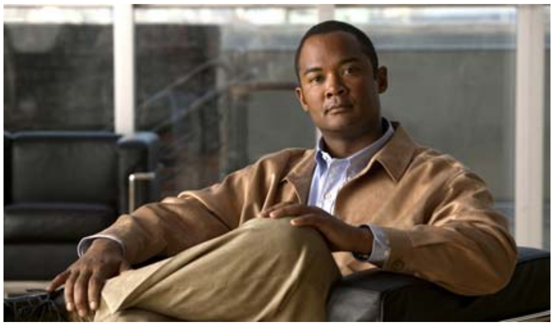
The following example shows that provision tag 1001 is configured for ephone 1 and provision tag 1002 is configured for ephone 2:

```
Telephony-service
    extension-assigner tag-type provision-tag
    auto assign 101-102
    auto-reg-ephone

Ephone-dn 101
    number 1001

Ephone-dn 102
    number 1002
```

```
Ephone 1
    provision-tag 1001
    mac-address 02EA.EAEA.0001
    button 1:101

Ephone 2
    provision-tag 1002
    mac-address 02EA.EAEA.0002
    button 1:102
```

Related Commands

Command	Description
extension-assigner tag-type	Specifies which type of tag is used by the extension assigner application to identify an ephone configuration.



Cisco Unified CME Commands: R

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This chapter contains commands to configure and maintain Cisco Unified Communications Manager Express (formally known as Cisco Unified CallManager Express). The commands are presented in alphabetical order. Some commands required for configuring Cisco Unified Communications Manager Express (Cisco Unified CME) may be found in other Cisco IOS command

ences. Use the command reference master index or search online to find these commands.

refer-ood enable

To enable out-of-dialog refer (OOD-R) processing, use the **refer-ood enable** command in SIP user-agent configuration mode. To disable OOD-R, use the **no** form of this command.

refer-ood enable [*request-limit*]

no refer-ood enable

Syntax Description

request-limit	(Optional) Maximum number of concurrent incoming OOD-R requests that the router can process. Range: 1 to 500. Default: 500.

Command Default

OOD-R processing is disabled.

Command Modes

SIP user-agent configuration

Command History

Release	Modification
12.4(11)XJ	This command was introduced.
12.4(15)T	This command was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

Out of dialog Refer allows applications to establish calls using the SIP gateway or Cisco Unified CME. The application sets up the call and the user does not dial out from their own phone.

Examples

The following example shows how to enable OOD-R:

```
Router(config)# sip-ua
Router(config-sip-ua)# refer-ood enable
```

Related Commands

Command	Description
authenticate (voice global)	Defines the authenticate mode for SIP phones in a Cisco Unified CME or Cisco Unified SRST system.
credential load	Reloads a credential file into flash memory.
debug voip application	Displays all application debug messages.

refer target dial-peer

To populate the Refer To portion of a SIP Refer message with the address from the dial peer for the directory number being configured, use the **refer target dial-peer** command in voice register dn configuration mode. To return to the default, use the **no** form of this command.

refer target dial-peer

no refer target

Syntax Description

This command has no arguments or keywords.

Command Default

Call is transferred to the destination as specified in the SIP Refer message.

Command Modes

Voice register dn configuration (config-register-dn)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XW2	Cisco Unified CME 4.2	This command was introduced.

Usage Guidelines

Use this command in voice register dn configuration mode to specify that the destination address for this directory number be the dial peer. If this command is not configured, Cisco IOS software will transfer the call to the destination in the SIP Refer message and if that destination address is Cisco Unified CME, call SIP will send out and route back to CME before sending to the directory number, creating two extra call legs.

The following partial output from the **show working-configuration** command shows the configuration for three directory numbers. This configuration will populate the Refer To portion of the SIP Refer message with the address from the dial peer for each of the directory numbers.

```
voice register dn  1
 session-server 1
 number 8999
 allow watch
 refer target dial-peer
!
voice register dn  2
 session-server 1
 number 8001
 allow watch
 refer target dial-peer
!
voice register dn  3
 session-server 1
 number 8101
 allow watch
 refer target dial-peer
```

regenerate (ctl-client)

To create a new CTLFile.tlv file after making changes to the CTL client configuration, use the **regenerate** command in CTL-client configuration mode. The **no** form of this command has no effect in the configuration.

regenerate

no regenerate

Syntax Description

This command has no arguments or keywords.

Command Default

A new CTLFile.tlv file is not created until this command is used.

Command Modes

CTL-client configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used with Cisco Unified CME phone authentication.

Examples

The following example gives the instruction to regenerate the CTL file with the current information.

```
Router(config)# ctl-client
Router(config-ctl-client)# server capf 10.2.2.2 trustpoint capftrust
Router(config-ctl-client)# server cme 10.2.2.3 trustpoint cmetp
Router(config-ctl-client)# server tftp 10.2.2.4 trustpoint tftptp
Router(config-ctl-client)# sast1 trustpoint sast1tp
Router(config-ctl-client)# sast2 trustpoint sast2tp
Router(config-ctl-client)# regenerate
```

register-id

To create an ID for explicitly identifying an external feature server during Register requests, use the **register-id** command in voice register session-server configuration mode. To remove a ID, use the **no** form of this command.

register-id *name*

no register-id *name*

Syntax Description

name	String of up to 30 alphanumeric characters.

Command Default

No identifier is created.

Command Modes

Voice register session-server configuration (config-register-fs)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XW1	Cisco Unified CME 4.2	This command was introduced.

Usage Guidelines

Use this command to create an ID for identifying a route point during Register requests.
Cisco Unified CME challenges and authenticates the initial keepalive Register request and issues a systemwide unique Cisco-referenceID to be included in the response to the Register request from this route point.

Examples

The following partial output shows the configuration of a session manager for an external feature server, including the register ID of CSR1:

```
router# show running-configuration
!
!
voice register session-server 1
 register-id CSR1
 keepalive 360
```

Related Commands

Command	Description
keepalive	Duration for registration after which the registration expires unless the feature server reregisters before the registration expiry.

registrar server (SIP)

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

registrar server [expires [max *sec***] [min** *sec***]]**

no registrar server

Syntax Description

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

Command Default

Disabled

Command Modes

SIP configuration

Command History

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

Usage Guidelines

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

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

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

Examples

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

```
voice service voip
 allow-connections sip-to-sip
```

```
sip
 registrar server expires max 1200 min 300
```

Related Commands

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

reset (ephone)

To perform a complete reboot of a single phone associated with a Cisco CallManager Express (Cisco CME) router, use the **reset** command in ephone configuration mode.

reset

Syntax Description

This command has no arguments or keywords.

Defaults

No default behavior or values

Command Modes

Ephone configuration

Command History

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

Usage Guidelines

After you update information for one or more phones associated with a Cisco CME router, the phone or phones must be rebooted. There are two commands to reboot the phones: **reset** and **restart.** The **reset** command performs a "hard" reboot similar to a power-off-power-on sequence. It reboots the phone and contacts the Dynamic Host Configuration Protocol (DHCP) server and the TFTP server to update from their information as well. The **restart** command performs a "soft" reboot by simply rebooting the phone without contacting the DHCP and TFTP servers. The **reset** command takes significantly longer to process than the **restart** command when you are updating multiple phones, but it must be used after updating phone firmware, user locale, network locale, or URL parameters. For simple button, line, or speed-dial changes, you can use the **restart** command.

Use the **reset (ephone)** command to perform a complete reboot of an IP phone when you are in ephone configuration mode. This command has the same effect as a **reset (telephony-service)** command that is used to reset a single phone.

This command has a **no** form, but the **no** form has no effect.

Examples

The following example resets the Cisco IP phone with a phone-tag of 1:

```
Router(config)# ephone 1
Router(config-ephone)# reset
```

Related Commands

Command	Description
ephone	Enters ephone configuration mode.
reset (telephony-service)	Performs a complete reboot of one or all phones associated with a Cisco CME router.
restart (ephone)	Performs a fast reboot of a single phone associated with a Cisco CME router.
restart (telephony-service)	Performs a fast reboot of one or all phones associated with a Cisco CME router.

reset (telephony-service)

To perform a complete reboot of one or all phones associated with a Cisco CallManager Express (Cisco CME) router, use the **reset** command in telephony-service configuration mode. To interrupt and cancel a sequential reset cycle, use the **no** form of the command with the **sequence-all** keyword.

reset {**all** [*time-interval*] | **cancel** | *mac-address* | **sequence-all**}

no reset {**all** [*time-interval*] | **cancel** | *mac-address* | **sequence-all**}

Syntax Description

all	Resets all Cisco IP phones served by the Cisco CME router. The router pauses for 15 seconds between the reset starts for each successive phone unless the *time-interval* argument is used to change that value.
time-interval	(Optional) Time interval, in seconds, between each phone reset. Range is from 0 to 60. Default is 15.
cancel	Interrupts a sequential reset cycle that was started with a **reset sequence-all** command.
mac-address	MAC address of a particular Cisco IP phone.
sequence-all	Resets all phones in strict one-at-a-time order by waiting for one phone to reregister before starting the reset for the next phone. The sequencing of resets prevents possible conflicts between phones trying to access TFTP services simultaneously. There is a reset timeout of 4 minutes, after which the router stops waiting for the currently registering phone to complete registration and starts to reset the next phone.

Defaults

time-interval: 15

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.1(5)YD	1.0	This command was introduced on the Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(2)XT	2.0	This command was implemented on the Cisco 1750 and Cisco 1751.
12.2(8)T	2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	2.0	This command was implemented on the Cisco 2600XM and Cisco 2691.
12.2(11)T	2.01	This command was implemented on the Cisco 1760.
12.2(11)YT	2.1	The *time-interval* range maximum was increased from 15 to 60 and the default was changed from 0 to 15.

Cisco IOS Release	Cisco CME Version	Modification
12.2(11)YT1	2.1	The **cancel** and **sequence-all** keywords were introduced.
12.2(15)T	2.1	This command was integrated into Cisco IOS Release 12.2(15)T.

Usage Guidelines

After you update information for one or more phones associated with a Cisco CME router, the phone or phones must be rebooted using either the **reset** command or the **restart** command**.** The **reset** command performs a "hard" reboot similar to a power-off-power-on sequence and contacts the Dynamic Host Configuration Protocol (DHCP) server and the TFTP server for updated information as well. The **restart** command performs a "soft" reboot by simply rebooting the phone without contacting the DHCP and TFTP servers. The **reset** command takes significantly longer to process than the **restart** command when you are updating multiple phones, but it must be used after you make changes to phone firmware, user locale, network locale, or URL parameters. For simple button, line, or speed-dial changes, you can use the **restart** command.

When you use the **reset** command, the default time interval of 15 seconds is recommended so that phone reset operations are staggered in order to avoid all phones attempting to access router system resources at the same time. A shorter interval may be used on systems with only a small number of phones or for cases where a simple reset of the phones is desired that does not result in the phones downloading updates to the phone firmware (using the router's TFTP service).

When you use the **reset sequence-all** command, the router waits for one phone to complete its reset and reregister before starting to reset the next phone. The delay provided by this command prevents multiple phones from attempting to access the TFTP server simultaneously and therefore failing to reset properly. Each reset operation can take several minutes when you use this command. There is a reset timeout of 4 minutes, after which the router stops waiting for the currently registering phone to complete registration and starts to reset the next phone.

If the router configuration is changed so that the eXtensible Markup Language (XML) configuration files for the phones are modified (changes are made to user locale, network locale, or phone firmware), then whenever you use the **reset all** or **restart all** command, the router automatically executes the **reset sequence-all** command instead. The **reset sequence-all** command resets phones one at a time in order to prevent multiple phones from trying to contact the TFTP server simultaneously. This one-at-a-time sequencing can take a long time if there are many phones. To avoid this automatic behavior, use the **reset all** *time-interval* command or the **restart all** *time-interval* command with an explicit argument that is not equal to the default 15-second time interval; for example, set a time interval of 14 seconds. If a **reset sequence-all** command has been started in error, use the **reset cancel** command to interrupt and cancel the sequence of resets.

The **restart** command allows the system to perform quick phone resets in which only the button template, line information, and speed-dial information is updated. See the documentation for the **restart** command for more information.

The **no** form of the command has an effect only when used with the **all** or **sequence-all** keyword, when it interrupts and cancels the sequential resetting of phones.

Examples

The following example resets all IP phones served by the Cisco CME router:

```
Router(config)# telephony-service
Router(config-telephony)# reset all
```

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

```
Router(config)# telephony-service
```

```
Router(config-telephony)# reset CFBA.321B.96FA
```

The following example resets all IP phones in sequential, nonoverlapping order:

```
Router(config)# telephony-service
Router(config-telephony)# reset sequence-all
```

Related Commands

Command	Description
reset (ephone)	Performs a complete reboot of a single phone associated with a Cisco CME router.
restart (ephone)	Performs a fast reboot of a single phone associated with a Cisco CME router.
restart (telephony-service)	Performs a fast reboot of one or all phones associated with a Cisco CME router.
telephony-service	Enters telephony-service configuration mode.

reset (voice logout-profile and voice user-profile)

To perform a complete reboot of all IP phones to which a particular extension-mobility profile is downloaded, use the **reset** command in voice logout-profile or voice user-profile configuration mode.

reset

Syntax Description

This command has no arguments or keywords.

Command Default

No reset is performed.

Command Modes

Voice logout-profile configuration (voice-logout-profile)
Voice user-profile configuration (voice-user-profile)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XW1	Cisco Unified CME 4.2	This command was introduced.

Usage Guidelines

Use this command to perform a "hard" reboot similar to a power-off-power-on sequence, which includes downloading updated information from the Dynamic Host Configuration Protocol (DHCP) server and the TFTP server.

Configure this command in voice logout-profile configuration mode after after creating or modifying a logout profile and assigning the profile to a Cisco Unified IP phone to enable that phone for extension mobility.

Configure this command in voice user-profile configuration mode after creating or modifying an individual user's profile for extension mobility.

Examples

The following example shows how to add a speed-dial definition to a logout profile and then reset all IP phones to which the logout profile is downloaded to propagate the modification:

```
router# configure terminal
router(config)# voice logout-profile 12
router (config-logout-profile)# speed-dial index number label label blf
router (config-logout-profile)# reset
router (config-logout-profile)# exit
```

Related Commands

Command	Description
logout-profile	Enables Cisco Unified IP phone for extension mobility and assigns a logout profile to this phone.

reset (voice register global)

To perform a complete reboot of all SIP phones associated with a Cisco CallManager Express (Cisco CME) router, use the **reset** command in voice register global configuration mode.

reset

Syntax Description

This command has no arguments or keywords.

Defaults

No default behavior or values

Command Modes

Voice register global configuration

Command History

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

Usage Guidelines

After you update information for one or more SIP phones associated with a Cisco CME router, reboot the phones by using the **reset** command. The **reset** command performs a "hard" reboot similar to a power-off-power-on sequence and contacts the Dynamic Host Configuration Protocol (DHCP) server and the TFTP server for updated information as well. Configure the **reset** command after you make changes to phone firmware, user locale, network locale, or URL parameters.

The time interval between each phone reset is 15 seconds, thereby avoiding an attempt by all phones to access router system resources at the same time.

This command has a **no** form, but the **no** form has no effect.

Examples

The following example shows how to reset all SIP phones served by the Cisco CME router:

```
Router(config)# voice register global
Router(config-register-global)# reset
```

Related Commands

Command	Description
reset (voice register pool)	Performs a complete reboot of a single SIP phone associated with a Cisco CME router.
voice register global	Enters voice register global configuration mode in order to set global parameters for all supported Cisco SIP phones in a Cisco CME or Cisco SIP SRST environment.
voice register pool	Enters voice register pool configuration mode for SIP phones.

reset (voice register pool)

To perform a complete reboot of a specific SIP phone associated with a Cisco CallManager Express (Cisco CME) router, use the **reset** command in voice register pool configuration mode. To interrupt a reset cycle, use the **no** form of this command.

reset

no reset

Syntax Description

This command has no arguments or keywords.

Defaults

No default behavior or values

Command Modes

Voice register pool configuration

Command History

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

Usage Guidelines

After you update information for one or more phones associated with a Cisco CME router, the phones must be rebooted by using the **reset** command. The **reset** command performs a "hard" reboot similar to a power-off-power-on sequence and contacts the Dynamic Host Configuration Protocol (DHCP) server and the TFTP server for updated information as well. Configure the **reset** command after you make changes to phone firmware, user locale, network locale, or URL parameters.

Use this command to perform a complete reboot of an individual SIP phone when you are in voice register pool configuration mode. To reset all SIP phones, use the **reset** (voice register global) command.

This command has a **no** form, but the **no** form has no effect.

Examples

The following example shows how to reset SIP phone 1 served by the Cisco CME router:

```
Router(config)# voice register pool 1
Router(config-register-pool)# reset
```

Related Commands

Command	Description
reset (voice register global)	Performs a complete reboot of all SIP phones associated with a Cisco CME router.
voice register global	Enters voice register global configuration mode in order to set global parameters for all supported Cisco SIP phones in a Cisco CME or Cisco SIP SRST environment.
voice register pool	Enters voice register pool configuration mode for SIP phones.

restart (ephone)

To perform a fast reboot of an IP phone associated with a Cisco CallManager Express (Cisco CME) router, use the **restart** command in ephone configuration mode. To cancel the reboot, use the **no** form of this command.

restart

no restart

Syntax Description

This command has no arguments or keywords.

Defaults

No default behavior or values

Command Modes

Ephone configuration

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.2(11)YT1	2.1	This command was introduced.
12.2(15)T	2.1	This command was integrated into Cisco IOS Release 12.2(15)T.

Usage Guidelines

This command causes the system to perform a fast phone reboot in which only the button template, lines, and speed-dial numbers are updated on the phone. For updates related to phone firmware, user locale, network locale, or URL parameters, use the **reset** command. The **restart** command is much faster than the **reset** command because the phone does not need to access the DHCP or TFTP server.

To restart all phones in a Cisco CME system for quick changes to buttons, lines, and speed-dial numbers, use the **restart** command in telephony-service configuration mode.

This command has a **no** form, but the **no** form has no effect.

Examples

The following example restarts the phone with phone-tag 1:

```
Router(config)# ephone 1
Router(config-ephone)# restart
```

Related Commands

Command	Description
reset (ephone)	Performs a complete reboot of a Cisco IP phone associated with a Cisco CME router.

Command	Description
reset (telephony-service)	Performs a complete reboot of one or all phones associated with a Cisco CME router.
restart (telephony-service)	Performs a fast reboot of one or all phones associated with a Cisco CME router.

restart (telephony-service)

To perform a fast reboot of one or all phones associated with a Cisco CallManager Express (Cisco CME) router, use the **restart** command in telephony-service configuration mode. To cancel the reboot, use the **no** form of this command.

restart {**all** [*time-interval*] | *mac-address*}

no restart {**all** [*time-interval*] | *mac-address*}

Syntax Description

all	Restarts all phones associated with the Cisco CME router.
time-interval	(Optional) Time between each phone restart, in seconds. Range is from 0 to 60. Default is 15.
mac-address	MAC address of the phone to be restarted.

Defaults

time-interval: 15

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.2(11)YT1	2.1	This command was introduced.
12.2(15)T	2.1	This command was integrated into Cisco IOS Release 12.2(15)T.

Usage Guidelines

This command causes the system to perform a fast phone reset in which only the button template, lines, and speed-dial numbers are updated on the phone. For updates related to phone firmware, user locale, network locale, or URL parameters, use the **reset** command.

Use the **restart** command to reboot IP phones after quick changes to buttons, lines, and speed-dial numbers. This command is much faster than the **reset** command because the phone does not access the DHCP or TFTP server.

To restart a single phone, use the **restart** command with the *mac-address* argument or use the **restart** command in ephone configuration mode.

If the router configuration is changed so that the eXtensible Markup Language (XML) configuration files for the phones are modified (changes are made to user locale, network locale, or phone firmware), then whenever you use the **reset all** or **restart all** command, the router automatically executes the **reset sequence-all** command instead. The **reset sequence-all** command resets phones one at a time in order to prevent multiple phones trying to contact the TFTP server simultaneously. This one-at-a-time sequencing can take a long time if there are many phones. To avoid this automatic behavior, use the **reset all** *time-interval* command or the **restart all** *time-interval* command with an explicit argument that is not equal to the default 15-second time interval; for example, set a time interval of 14 seconds. If a **reset sequence-all** command has been started in error, use the **reset cancel** command to interrupt and cancel the sequence of resets.

The **no** form of the command has an effect only when used with the **all** keyword, when it interrupts and cancels the sequential restarting of phones.

Examples

The following example performs a quick restart of all phones in a Cisco CME system:

```
Router(config)# telephony-service
Router(config-telephony)# restart all
```

Related Commands

Command	Description
reset (ephone)	Performs a complete reboot of a Cisco IP phone associated with a Cisco CME router.
reset (telephony-service)	Performs a complete reboot of one or all phones associated with a Cisco CME router.
restart (ephone)	Performs a fast reboot of a single phone associated with a Cisco CME router.
telephony-service	Enters telephony-service configuration mode.

restart (voice register)

To perform a fast reset of one or all SIP phones associated with a Cisco Unified CME router, use the **restart** command in voice register global or voice register pool configuration mode. To cancel the reboot, use the **no** form of this command.

restart

no restart

Syntax Description

This command has no arguments or keywords.

Command Default

SIP phones are not restarted.

Command Modes

Voice register global configuration
Voice register pool configuration

Command History

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

Usage Guidelines

This command causes the system to perform a fast phone reset in which only the button template, lines, and speed-dial numbers are updated on the phone. For updates related to phone firmware, user locale, network locale, or URL parameters, use the **reset** command.

Use this command to reboot SIP phones after quick changes to buttons, lines, and speed-dial numbers. This command is much faster than the **reset** command because the phone does not access the DHCP or TFTP server.

To restart a single SIP phone, use the **restart** command in voice register pool configuration mode. To restart all SIP phones in a Cisco Unified CME system, use the **restart** command in voice register global configuration mode.

This command has a **no** form, however the **no** form has no effect.

Note This command requires firmware load 8-0-2-14 or later versions; it is not supported in older SIP phone loads. To support this command on SIP phones using older firmware, you must upgrade all your phone firmware.

Examples

The following example performs a quick restart of all SIP phones in a Cisco Unified CME system:

```
Router(config)# voice register global
Router(config-register-global)# restart
```

The following example performs a quick restart of SIP phone 10:

```
Router(config)# voice register pool 10
Router(config-register-pool)# restart
```

Related Commands

Command	Description
reset (voice register pool)	Performs a complete reboot of a single SIP phone associated with a Cisco Unified CME router.
reset (voice register global)	Performs a complete reboot of all SIP phones associated with a Cisco Unified CME router.

ring (ephone-dn)

o set the ring pattern for all incoming calls to an ephone-dn, use the **ring** command in ephone-dn configuration mode. To return to the standard ring pattern, use the **no** form of this command.

ring {**external** | **feature** | **internal**} [**primary** | **secondary**]

no ring

Syntax Description

external	External ring pattern is used for all incoming calls.
feature	Feature ring pattern is used for all incoming calls.
internal	Internal ring pattern is used for all incoming calls.
primary	(Optional) Ring pattern is used on primary number only.
secondary	(Optional) Ring pattern is used on secondary number only.

Command Default

Standard ring pattern is used.

Command Modes

Ephone-dn configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command allows you to select one of the three ring styles supported by SCCP—internal, external, or feature ring. The ring pattern is used for all types of incoming calls to this directory number, on all phones on which the directory number appears. If the phone is already in use, an incoming call is presented as a call-waiting call and uses the distinctive call-waiting beep.

If the **primary** or **secondary** keyword is used, the distinctive ring is used only if the incoming called number matches the primary number or secondary number defined for the ephone-dn. If there is no secondary number defined for the ephone-dn, the **secondary** keyword has no effect.

By default, Cisco Unified CME uses the internal ring pattern for calls between local IP phones and uses the external ring pattern for all other types of calls.

You can associate the feature ring pattern with a specific button on a phone by using the **button f** command. This assigns the ring pattern to the button on the phone so that different phones that share the same directory number can use a different ring style.

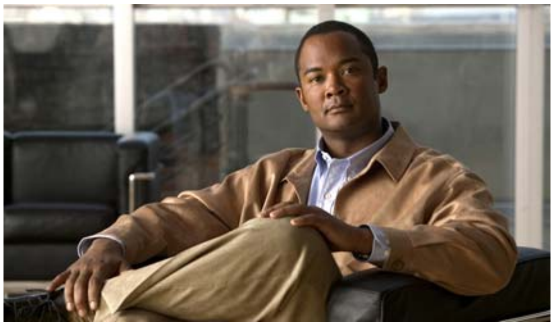
Examples

The following example sets external ringing for all incoming calls on extension 2389.

```
ephone-dn 24
 number 2389
 ring external
```

Related Commands

Command	Description
button	Associates ephone-dns with individual buttons on an IP phone and specifies line type or ring behavior.



Cisco Unified CME Commands: sast1 trustpoint through show fb-its-log

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This chapter contains commands to configure and maintain Cisco Unified Communications Manager Express (formally known as Cisco Unified CallManager Express). The commands are presented in alphabetical order. Some commands required for configuring Cisco Unified Communications Manager Express (Cisco Unified CME) may be found in other Cisco IOS command references. Use the command reference master index or search online to find these commands.

sast1 trustpoint

To specify the name of the SAST1 trustpoint, use the **sast1 trustpoint** command in CTL-client configuration mode. To return to the default, use the **no** form of this command.

sast1 trustpoint *label*

no sast1

Syntax Description

label	Name of the SAST1 trustpoint.

Command Default

No SAST1 trustpoint name is specified

Command Modes

CTL-client configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used with Cisco Unified CME phone authentication.

The **sast1 trustpoint** and **sast2 trustpoint** commands are used to set up the System Administrator Security Token (SAST) credentials, which are used to sign the CTL file. The SAST1 and SAST2 certificates must be different from each other, but to conserve memory either one of them can use the same certificate as Cisco Unified CME. The CTL file is always signed by SAST1 credentials. The SAST2 credentials are included in the CTL file so that if the SAST1 certificate is compromised, the CTL file can be signed by SAST2 to prevent the phones from being reset to their factory defaults.

Examples

The following example names sast1tp as the SAST1 trustpoint.

```
Router(config)# ctl-client
Router(config-ctl-client)# server capf 10.2.2.2 trustpoint capftrust
Router(config-ctl-client)# server cme 10.2.2.3 trustpoint cmetp
Router(config-ctl-client)# server tftp 10.2.2.4 trustpoint tftptp
Router(config-ctl-client)# sast1 trustpoint sast1tp
Router(config-ctl-client)# sast2 trustpoint sast2tp
Router(config-ctl-client)# regenerate
```

Related Commands

Command	Description
sast2 trustpoint	Specifies the name of the SAST2 trustpoint.

sast2 trustpoint

To specify the name of the SAST2 trustpoint, use the **sast2 trustpoint** command in CTL-client configuration mode. To return to the default, use the **no** form of this command.

sast2 trustpoint *label*

no sast2

Syntax Description

label	Name of the SAST2 trustpoint.

Command Default

No SAST2 trustpoint name is specified.

Command Modes

CTL-client configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used with Cisco Unified CME phone authentication.

The **sast1 trustpoint** and **sast2 trustpoint** commands are used to set up the System Administrator Security Token (SAST) credentials, which are used to sign the CTL file. The SAST1 and SAST2 certificates must be different from each other, but to conserve memory either one of them can use the same certificate as Cisco CME. The CTL file is always signed by SAST1 credentials. The SAST2 credentials are included in the CTL file so that if the SAST1 certificate is compromised, the CTL file can be signed by SAST2 to prevent the phones from being reset to their factory defaults.

Examples

The following example names sast2tp as the SAST2 trustpoint.

```
Router(config)# ctl-client
Router(config-ctl-client)# server capf 10.2.2.2 trustpoint capftrust
Router(config-ctl-client)# server cme 10.2.2.3 trustpoint cmetp
Router(config-ctl-client)# server tftp 10.2.2.4 trustpoint tftptp
Router(config-ctl-client)# sast1 trustpoint sast1tp
Router(config-ctl-client)# sast2 trustpoint sast2tp
Router(config-ctl-client)# regenerate
```

Related Commands

Command	Description
sast1 trustpoint	Specifies the name of the SAST1 trustpoint.

sdspfarm conference mute-on mute-off

To define mute-on and mute-off DTMF digits for use during conferencing, use the **sdspfarm conference mute-on mute-off** command in telephony-service configuration mode. To disable the mute-on and mute-off digits, use the **no** form of this command.

sdspfarm conference mute-on *mute-on-digits* **mute-off** *mute-off-digits*

no sdspfarm conference mute-on *mute-on-digits* **mute-off** *mute-off-digits*

Syntax Description

mute-on *mute-on-digits*	Defines the buttons you press on your phone to mute during a conference. Maximum: 3 digits. Valid values are the numbers and symbols that appear on your telephone keypad: 1, 2, 3, 4, 5, 6, 7, 8, 9, 0, *, and #.
mute-off *mute-off-digits*	Defines the buttons you press on your IP phone to unmute during a conference. Maximum: 3 digits. Valid values are the numbers and symbols that appear on your telephone keypad: 1, 2, 3, 4, 5, 6, 7, 8, 9, 0, *, and #.

Command Default

No mute-on or mute-off digits are defined.

Command Modes

Telephony-service configuration

Command History

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

Usage Guidelines

You must define mute-on and mute-off digits to mute or unmute your phone using the keypad during a conference. The mute-on digits and mute-off digits can be the same or different. You can mute and unmute your phone using the phone's mute button also. You must unmute the phone in the same way that you muted it, either with the keypad or the mute button.

Examples

The following example configures #5 as the buttons to press to mute and unmute the phone during a conference:

```
Router(config-telephony)# sdspfarm conference mute-on #5 mute-off #5
```

sdspfarm tag

To permit a digital-signal-processor (DSP) farm to be to registered to Cisco CallManager Express (Cisco CME) and associate it with a Skinny Client Control Protocol (SCCP) client interface's MAC address, use the **sdspfarm tag** command in telephony-service configuration mode. To delete a tag generated by the **sdspfarm tag** command, use the **no** form of this command.

sdspfarm tag *number device-name*

no sdspfarm tag *number device-name*

Syntax Description

number	Declares a numeric name for a DSP farm. Number from 1 to 5.
device-name	The MAC address of the SCCP client interface that is preceded by the Message Transfer Part (MTP).

Defaults

No default behavior or values

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.3(11)T	3.2	This command was introduced.

Usage Guidelines

DSP farms are sets of DSP resources used for conferencing and transcoding only. DSP farms do not include voice termination resources. Use the **show interface** command to find the MAC address of the SCCP client interface.

Examples

The following example declares tag 1 as the MAC address of mac000a.8aea.ca80. The **show interface** command is used to obtain the MAC address.

```
Router# show interface FastEthernet 0/0
.
.
.
FastEthernet0/0 is up, line protocol is up
Hardware is AmdFE, address is 000a.8aea.ca80 (bia 000a.8aea.ca80)
.
.
.
Router(config)# telephony-service
Router(config-telephony)# sdspfarm tag 1 mac000a.8aea.ca80
```

Related Commands

Command	Description
sdspfarm transcode	Specifies the maximum number of transcoding sessions allowed per Cisco CME router.
sdspfarm units	Specifies the maximum number of DSP farms that are allowed to be registered to the SCCP server.

sdspfarm transcode sessions

To specify the maximum number of transcoding sessions allowed per Cisco CallManager Express (Cisco CME) router, use the **sdspfarm transcode sessions** command in telephony-service configuration mode. To return to the default transcode session of 0, use the **no** form of this command.

sdspfarm transcode sessions *number*

no sdspfarm transcode sessions *number*

Syntax Description

number	Declares the number of DSP farm sessions. Valid values are numbers from 1 to 128.

Defaults

The default is 0.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.3(11)T	3.2	This command was introduced.

Usage Guidelines

The transcoding is allowed between G.711 and G.729. A session consists of two transcode streams. To configure this information, you must know how many digital-signal-processor (DSP) farms are configured on the network module (NM) farms in your Cisco CME router. DSP farms are sets of DSP resources used for conferencing and transcoding only. DSP farms do not include voice termination resources. To learn how many DSP farms have been configured on your Cisco CME router, use the **show sdspfarm** command.

Examples

The following example sets the maximum number of transcoding sessions allowed on the Cisco CME router to 20:

```
Router(config)# telephony-service
Router(config-telephony)# sdspfarm transcode sessions 20
```

Related Commands

Command	Description
sdspfarm tag	Declares a DSP farm and associates it with an SCCP client interface's MAC address.
sdspfarm unit	Specifies the maximum number of DSP farms that are allowed to be registered to the SCCP server.
show sdspfarm	Displays the status of the configured DSP farms and transcoding streams.

sdspfarm units

To specify the maximum number of digital-signal-processor (DSP) farms that are allowed to be registered to the Skinny Client Control Protocol (SCCP) server, use the **sdspfarm units** command in telephony-service configuration mode. To set the number of DSP farms to the default value of 0, use the **no** form of this command.

sdspfarm units *number*

no sdspfarm units *number*

Syntax Description

number	Declares the number of DSP farms. Valid values are numbers from 0 to 5.

Defaults

The default number is 0.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.3(11)T	3.2	This command was introduced.

Usage Guidelines

DSP farms are sets of DSP resources used for conferencing and transcoding only. DSP farms do not include voice termination resources.

Examples

The following example configures a Cisco CME router to register one DSP farm:

```
Router(config)# telephony-service
Router(config-telephony)# sdspfarm units 1
```

Related Commands

Command	Description
sdspfarm tag	Declares a DSP farm and associates it with an SCCP client interface's MAC address.
sdspfarm transcode	Specifies the maximum number of transcoding sessions allowed per Cisco CME router.

sdspfarm unregister force

To remove all transcoding streams associated with active calls, use the **sdspfarm unregister force** command in telephony-service configuration mode. To deactivate the removal of transcoding streams, use the **no** form of this command.

sdspfarm unregister force

no sdspfarm unregister force

Syntax Description

This command has no arguments or keywords.

Defaults

The default is transcoding streams are not removed.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.3(11)T	3.2	This command was introduced.

Usage Guidelines

If any of the SCCP server's associated streams are functioning in active calls, the default response for the **sdspfarm unregister force** command is to reject them. If no stream is used in a call, all of the transcoding streams associated with the DSP farm will be released, and SCCP server can recycle those streams for other DSP farms.

Examples

The following example removes all transcoding streams for active calls.

```
Router(config)# telephony-service
Router(config-telephony)# sdspfarm unregister force
```

Related Commands

Command	Description
sdspfarm tag	Declares a DSP farm and associates it with an SCCP client interface's MAC address.
sdspfarm unit	Specifies the maximum number of DSP farms that are allowed to be registered to the SCCP server.
show sdspfarm	Displays the status of the configured DSP farms and transcoding streams.

secondary start

To specify the ephone hunt group agent to receive parked calls that are forwarded to the secondary pilot number, use the **secondary start** command in ephone-hunt configuration mode. To disable this setting, use the **no** form of this command.

secondary start [**current** | **next** | *list-position*]

no secondary start [**current** | **next** | *list-position*]

Syntax Description

current	The ephone-dn that parked this call.
next	The ephone-dn that follows the parking ephone-dn in the list specified by the **list** command.
list-position	The ephone-dn at the specified position in the list specified by the **list** command. Range is from 1 to 20.

Command Default

No default behavior or values

Command Modes

Ephone-hunt configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

When a call that has been parked by a hunt group agent meets either of these conditions, the hunt group agent to receive the call can be specified with the **secondary start** command:

- The call is recalled from call park to the hunt group secondary pilot number.
- The call is transferred from call park to an ephone-dn that forwards the call to the hunt group secondary pilot number.

Examples

The following example specifies that the third hunt group member (3031) will receive calls that are recalled or forwarded to the secondary group hunt pilot number (3001) after the calls have been parked by an ephone-dn.

```
ephone-hunt 17 sequential
 pilot 3000 secondary 3001
 list 3011, 3021, 3031
 timeout 10
 final 7600
 secondary start 3
```

Related Commands

Command	Description
ephone-hunt	Defines an ephone hunt group and enters ephone-hunt configuration mode.
list	Creates a list of extensions that are members of an ephone hunt group

secondary-dialtone

To activate a secondary dial tone when a Cisco IP phone user dials a defined public switched telephone network (PSTN) access prefix, use the **secondary-dialtone** command in telephony-service configuration mode. To disable the secondary dial tone, use the **no** form of this command.

secondary-dialtone *digit-string*

no secondary-dialtone

Syntax Description

digit-string	String of up to 32 numbers that defines an access prefix.

Defaults

No secondary dial tone is enabled.

Command Modes

Telephony-service configuration

Command History

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

Usage Guidelines

The secondary dial tone is turned off when the next number after the access prefix is pressed. For example, if 8 is the access prefix and a person dials 8 555-0145, the secondary dial tone is turned off when the digit 5 is pressed.

Examples

The following example enables a secondary dial tone when a Cisco IP phone users press the digit 9 to get an outside line:

```
Router(config)# telephony-service
Router(config-telephony)# secondary-dialtone 9
```

Related Commands

Command	Description
telephony-service	Enters telephony-service configuration mode.

secure-signaling trustpoint

To specify the name of the PKI trustpoint with the certificate to use for TLS handshakes with IP phones on TCP port 2443, use the **secure-signaling trustpoint** command in telephony-service configuration mode. To return to the default, use the **no** form of this command.

secure-signaling trustpoint *label*

no secure-signaling trustpoint

Syntax Description

label	Name of a configured PKI trustpoint with a valid certificate.

Command Default

No trustpoint is specified.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used with Cisco Unified CME phone authentication to name the trustpoint that enables handshaking between Cisco Unified CME and a phone to ensure secure SCCP signaling on TCP port 2443.

Examples

The following example names server25, the CAPF server, as the trustpoint to enable secure SCCP signaling:

```
Router(config)# telephony-service
Router(config-telephony)# device-security-mode authenticated
Router(config-telephony)# secure-signaling trustpoint server25
Router(config-telephony)# tftp-server-credentials trustpoint server12
Router(config-telephony)# load-cfg-file slot0:Ringlist.xml alias Ringlist.xml sign create
Router(config-telephony)# exit
```

semi-attended enable (voice register template)

To enable call transfer at the alert call stage for Cisco IP Phone 7940s, Cisco IP Phone 7940Gs, Cisco IP Phones 7960s, and Cisco IP Phone and 7960Gs, use the **semi-attended enable** command in the voice register template mode. To disable call transfer, use the **no** form of this command.

semi-attended enable

no semi-attended enable

Syntax Description

This command has no arguments or keywords.

Defaults

Call transfer at the alert call stage is disabled

Command Modes

Voice register template

Command History

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

Examples

The following example shows how to enable call transfer at the alter call stage:

```
Router(config)# voice register template 1
Router(config-register-temp)# semi-attend enable
```

Related Commands

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

server

To specify the IP address of a presence server for sending presence requests from internal watchers to external presence entities, use the **server** command in presence configuration mode. To remove the server, use the **no** form of this command.

server *ip-address*

no server

Syntax Description

ip-address	IP address of the remote presence server.

Command Default

A remote presence server is not used.

Command Modes

Presence configuration

Command History

Release	Modification
12.4(11)XJ	This command was introduced.
12.4(15)T	This command was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

This command specifies the IP address of a presence server that handles presence requests when the watcher and presence entity (presentity) are not colocated. The router acts as the presence server and processes all presence requests and status notifications when a watcher and presentity are both internal. If a subscription request is for an external presentity, the request is sent to the remote server specified by this command.

Examples

The following example shows a presence server with IP address 10.10.10.1:

```
Router(config)# presence
Router(config-presence)# allow subscribe
Router(config-presence)# server 10.10.10.1
```

Related Commands

Command	Description
allow subscribe	Allows internal watchers to monitor external presence entities (directory numbers).
allow watch	Allows a directory number on a phone registered to Cisco Unified CME to be watched in a presence service.
max-subscription	Sets the maximum number of concurrent watch sessions that are allowed.
show presence global	Displays configuration information about the presence service.

Command	Description
show presence subscription	Displays information about active presence subscriptions.
watcher all	Allows external watchers to monitor internal presence entities (directory numbers).

server (CTL-client)

To enter trustpoint information for the CAPF server, Cisco Unified CME router, or TFTP server into the router configuration, use the **server** command in CTL-client configuration mode. To return to the default, use the **no** form of this command.

server {**capf** | **cme** | **cme-tftp** | **tftp**} *ip-address* **trustpoint** *label*

no server {**capf** | **cme** | **cme-tftp** | **tftp**} *ip-address*

Syntax Description

capf	CAPF server.
cme	Cisco Unified CME router.
cme-tftp	Combined Cisco Unified CME router and TFTP server.
tftp	TFTP server.
ip-address	IP address of the entity.
trustpoint *label*	Name of the PKI trustpoint for the entity.

Command Default

Trustpoint information about the CAPF server, Cisco Unified CME router, or TFTP server is not present in the security configuration.

Command Modes

CTL-client configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used with Cisco Unified CME phone authentication. Cisco IOS software stores credential information in a trustpoint. The trustpoint label in this command names the specified PKI trustpoint.

Note Repeat this command for each entity that requires a trustpoint.

Examples

The following example defines trustpoint names and IP addresses for the CAPF server, the Cisco Unified CME router, and the TFTP server:

```
Router(config)# ctl-client
Router(config-ctl-client)# server capf 10.2.2.2 trustpoint capftrust
Router(config-ctl-client)# server cme 10.2.2.3 trustpoint cmetp
Router(config-ctl-client)# server tftp 10.2.2.4 trustpoint tftptp
Router(config-ctl-client)# sast1 trustpoint sast1tp
Router(config-ctl-client)# sast2 trustpoint sast2tp
```

```
Router(config-ctl-client)# regenerate
```

server-security-mode

To change the security mode of the Cisco Unified CME phone authentication server, use the **server-security-mode** command in telephony-service configuration mode. To return to the default, use the **no** form of this command.

server-security-mode {**non-secure** | **secure**}

no server-security-mode {**non-secure** | **secure**}

Syntax Description

non-secure	Non-secure mode.
secure	Secure mode.

Command Default

When the CTL file is initially generated, the mode is set to secure.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used with Cisco Unified CME phone authentication.

This command has no effect until the CTL file is initially generated by the CTL client. When the CTL file is successfully generated, the CTL client automatically sets the server security mode to secure. You can toggle the mode thereafter.

This command must be followed by the **regenerate** command in CTL-client configuration mode, which regenerates the CTL file. This is necessary because if the security mode is non-secure, its credentials are zeroed out in the CTL file. If the security mode is secure, the CTL file contains the server's credentials.

Examples

The following example changes the mode of the server to non-secure.

```
telephony-service
 server-security-mode non-secure
```

Related Commands

Command	Description
regenerate	Creates a new CTLFile.tlv file after changes to the CTL client configuration.

service directed-pickup

To enable directed pickup in a Cisco Unified CME system, use the **service directed-pickup** command in telephony-service configuration mode. To globally disable the directed pickup feature and change the action of the PickUp soft key on IP phones, use the **no** form of this command.

service directed-pickup

no service directed-pickup

Syntax Description

This command has no arguments or keywords.

Command Default

Directed pickup is enabled.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is enabled by default.

To globally disable directed call pickup for all phone users, use the **no** form of this command. Using the **no** form of the command also changes the behavior of the PickUp soft key on all IP phones so that a user pressing it invokes local group pickup rather than directed call pickup.

To selectively remove the directed pickup function from one or more ephones, use the **features blocked** command in ephone-template mode. The **features blocked** command removes the PickUp soft key from IP phones and blocks directed call pickup on analog phones. When you use the **features blocked** command, it becomes part of an ephone template that you apply to one or more ephones on which you want the specified features to not appear as soft keys.

Examples

The following example globally disables directed call pickup.

```
telephony-service
 no service directed-pickup
```

Related Commands

Command	Description
features blocked	Prevents one or more features from being used on one or more ephones.
telephony-service	Enters telephony-service configuration mode.

service dnis dir-lookup

To allow the display of names associated with called numbers for incoming calls on IP phones, use the **service dnis dir-lookup** command in telephony-service configuration mode. To deactivate directory lookup, use the **no** form of this command.

service dnis dir-lookup

no service dnis dir-lookup

Defaults

The default is directory service lookup is inactive.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.3(11)T	3.2	This command was introduced.

Usage Guidelines

The **service dnis dir-lookup** command provides a called number to the name-lookup service to support display of the name associated with the called number for incoming calls to IP phones. The display name is obtained from the Cisco CME system's list of Cisco CME directory names created using the **directory entry** command in the telephony-service configuration mode.

Called numbers can be displayed for overlaid ephone-dn and for ephone-dns that are not overlaid. Secondary line are supported.

To allow a single ephone-dn to receive calls for multiple different called numbers (with different names), you must use wildcard "." characters in the number field for the ephone-dn.

To use the **service dnis dir-lookup** command in conjunction with the **ephone-hunt** command, you can configure the ephone-hunt group to use a pilot number that contains wildcard "." characters. This allows the ephone-hunt group to receive calls from different numbers. Individual ephone-dns that are configured as members of the hunt group with the **ephone-hunt list** command must not have wildcard characters in their number fields.

If the **service dnis dir-lookup** command is used at the same time as the **service dnis overlay** command, the directory-lookup service takes precedence in resolving the name for the called number.

Examples

The following is an example of an overlaid ephone-dn configuration, where the **service dnis dir-lookup** command allows one of three directory entry names to be displayed on three IP phones when a call is placed to a number declared in the **directory entry** command.

```
telephony-service
 service dnis dir-lookup

 directory entry 1 0001 name dept1
 directory entry 2 0002 name dept2
 directory entry 3 0003 name dept3

ephone-dn 1
```

```
 number 0001
ephone-dn 2
 number 0002

ephone-dn 2
 number 0002

ephone 1
 button 1o1,2,3
ephone 2
 button 1o1,2,3
ephone 3
 button 1o1,2,3
```

The following is an example of an ephone-dn configuration in which the overlay function is not in use. There are three IP phones, each with two buttons. Button 1 receives calls from user1, user2, and user3; button 2 receives calls from user4, user5, and user6.

```
telephony-service
 service dnis dir-lookup

 directory entry 1 5550001 name user1
 directory entry 2 5550002 name user2
 directory entry 3 5550003 name user3

 directory entry 4 5550010 name user4
 directory entry 5 5550011 name user5
 directory entry 6 5550012 name user6

ephone-dn 1
 number 555000.

ephone-dn 2
 number 5552001.

ephone 1
 button 1:1
 button 2:2
 mac-address 1111.1111.1111

ephone 2
 button 1:1
 button 2:2
 mac-address 2222.2222.2222
```

The following is an example of a hunt-group configuration. There are three phones, each with two buttons, and each button is assigned two numbers. When a person calls 5550341, Cisco CME matches the hunt-group pilot secondary number (555....) and rings button 1 on one of the two phones and displays “user1.” The selection of the phone is dependent on the **ephone-hunt** command settings. For more information about hunt groups, refer to the “Ephone Hunt Groups” section of the *Cisco CallManager Express 3.3 System Administrator Guide*.

```
telephony-service
 service dnis dir-lookup
 max-redirect 20
 directory entry 1 5550341 name user1
 directory entry 2 5550772 name user2
 directory entry 3 5550263 name user3
 directory entry 4 5550150 name user4

ephone-dn 1
 number 1001
ephone-dn 2
```

```
 number 1002
ephone-dn 3
 number 1003
ephone-dn 4
 number 1004

ephone 1
 button 1o1,2
 button 2o3,4
 mac-address 1111.1111.1111

ephone 2
 button 1o1,2
 button 2o3,4
 mac-address 2222.2222.2222


ephone-hunt 1 peer
 pilot 1000 secondary 555....
 list 1001, 1002, 1003, 1004
 final 5556000
 hops 5
 preference 1
 timeout 20
 no-reg
```

The following is an example of a secondary-line configuration. Ephone-dn 1 can accept calls from extension 1001 and from 5550000, 5550001, and 5550002.

```
telephony-service
 service dnis dir-lookup

 directory entry 1 5550000 name doctor1
 directory entry 2 5550001 name doctor2
 directory entry 3 5550002 name doctor3

ephone-dn 1
 number 1001 secondary 555000.

ephone 1
 button 1:1
 mac-address 2222.2222.2222
```

Related Commands

Command	Description
directory entry	Adds an entry to a local phone directory that can be displayed on IP phones.
ephone-hunt	Enters ephone-hunt configuration mode to create a hunt group for use in a Cisco CME system.
list	Creates a list of extensions that are members of a Cisco CME ephone hunt group.
service dnis overlay	Allows an ephone-dn name to appear on receiving IP phones' displays when the ephone-dn's number is called.

service dnis overlay

To allow incoming calls to an ephone-dn overlay to display called ephone-dn names, use the **service dnis overlay** command in telephony-service configuration mode. To deactivate the service dialed number identification service (DNIS) overlay, use the **no** form of this command.

service dnis overlay

no service dnis overlay

Defaults

The ephone-dn names in an ephone-dn overlay are not displayed on IP phones.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.3(11)T	3.2	This command was introduced.

Usage Guidelines

The **service dnis overlay** command allows phone users to determine which ephone-dn within an overlay set is being called. Up to ten ephone-dns are allowed per overlay set. When an incoming call is presented under a **service dnis overlay** command configuration, the phone displays the name of the individual ephone-dn according to the **name** command configured under the ephone-dn configuration mode. Note that for an ephone-dn name to be displayed on IP phones, you must first assign ephone-dn names with the **name** command in ephone-dn configuration mode.

The number of the first ephone-dn listed in the **button** command is the default display for all phones using the same set of overlaid ephone-dns. Calls to the first ephone-dn display the caller ID. Calls to the remaining ephone-dns display ephone-dn names. For example, if there are three phones with one overlay set containing five ephone-dns, the first ephone-dn's number listed is the default display for all three phones. A call to the first ephone-dn displays the caller ID on all phones until the call is picked up. When the call is answered by phone 1, the displays in phone 2 and phone 3 return to the default display. Calls to the last four ephone-dns display ephone-dn names.

If the **service dnis overlay** command is used at the same time as the **service dnis dir-lookup** command, the **service dnis dir-lookup** command takes precedence in determining the name to be displayed.

Examples

The following is an overlay configuration for two phones with button 1 assigned to pick up three 800 numbers from three ephone-dns that have been assigned names. The default display for button 1 is 18005550100. A call to 18005550100 displays the caller ID. Calls to 18005550001 and 18005550002 display "name1" and "name2," respectively.

```
telephony-service
 service dnis overlay

ephone-dn 1
 name mainnumber
 number 18005550100
ephone-dn 2
 name name1
```

```
 number 18005550001
ephone-dn 3
 name name2
 number 18005550002

ephone 1
 button 1o1,2,3
ephone 2
 button 1o1,2,3
```

Related Commands

Command	Description
name	Associates a name with a Cisco CME extension (ephone-dn).
service dnis dir-lookup	Allows directory entry lookup for the display of directory entry names on IP phones.

service dss

To enable DSS (Direct Station Select) in a Cisco Unified CME system, use the **service dss** command in telephony-service configuration mode. To globally disable the DSS feature, use the **no** form of this command.

service dss

no service dss

Syntax Description

This command has no arguments or keywords.

Command Default

DSS service is disabled.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(6)XE	Cisco Unified CME 4.0(2)	This command was introduced.
12.4(4)XC4	Cisco Unified CME 4.0(3)	This command was introduced.
12.4(11)T	Cisco Unified CME 4.0(3)	This command is integrated into Cisco IOS Release 12.4(11)T.

Usage Guidelines

This command enables phone users to quickly transfer calls to an extension selected by a speed-dial or monitor line button without having to press the Transfer button. If this command is enabled, a user can transfer a call when the call is in the connected state by simply pressing a speed-dial or monitor line button to select the transfer destination. The transfer action is automatically implied by CME if the **service dss** command is enabled.

This feature is supported only on phones on which monitor-line buttons for speed dial or speed-dial line buttons are configured.

Using the **no** form of the command changes the behavior of the speed-dial line button on all IP phones so that a user pressing a speed-dial button in the middle of a connected call will play out the speed-dial digits into the call without transferring the call. If the **service dss** command is disabled, the phone user must press the Transfer button before pressing the speed-dial line button or monitor line button to transfer the call.

For Cisco Unified CME 4.0 and earlier, the **transfer-system full-consult dss** command is used to select between blind transfers and consult transfers for the DSS case.

Examples

The following example globally enables directed call pickup.

```
telephony-service
 service dss
```

Related Commands

Command	Description
button	Associates ephone-dns with individual buttons on a Cisco Unified IP phone and to specify line type, such as monitor mode for a shared line.
speed-dial	Defines a unique speed-dial identifier, a digit string to dial, and an optional label to display next to a line button.

service local-directory

To enable the availability of the local directory service on IP phones served by the Cisco CallManager Express (Cisco CME) router, use the **service local-directory** command in telephony service configuration mode. To disable the display, use the **no** form of this command.

service local-directory [authenticate]

no service local-directory [authenticate]

Syntax Description

authenticate	(Optional) Requires authentication for local directory search requests.

Command Default

Local directory service is available on IP Phones.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco Product	Modifications
12.2(11)YT	Cisco ITS 2.1	This command was introduced.
12.2(15)T	Cisco ITS 2.1	This command was integrated into Cisco IOS Release 12.2(15)T.
12.2(15)ZJ	Cisco CME 3.0	The **authenticate** keyword was introduced.
12.3(4)	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Usage Guidelines

Use this command with Cisco IOS Telephony Services V2.1, Cisco CME 3.0, or a later version.

Examples

The following example specifies that the directory service should not be available on the IP phones served by this ITS router:

```
Router(config)# telephony-service
Router(config-telephony-service)# no service local-directory
```

Related Commands

Command	Description
telephony-service	Enters telephony-service configuration mode.

service phone

To modify the vendorConfig parameters in the configuration file, use the **service phone** command in telephony-service or ephone-template configuration mode. To disable a setting, use the **no** form of this command.

service phone *parameter-name parameter-value*

no service phone *parameter-name parameter-value*

Syntax Description

parameter-name	Name of the vendorConfig parameter in the configuration file. For valid parameter names, see Table 12. **Note** Parameter names are word- and case-sensitive.
parameter-value	Value for the vendorConfig parameter. For valid parameter values and defaults, see Table 12.

Command Default

The vendorConfig parameter defaults are enabled.

Command Modes

Telephony-service configuration
Ephone-template configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)XL	Cisco CME 3.2.1	This command was introduced.
12.3(14)T	Cisco CME 3.3	Support was added for the Cisco Unified IP Phone 7971G-GE., and this command was integrated into Cisco IOS Release 12.3(14)T.
12.4(4)XC	Cisco Unified CME 4.0	This command was made available in ephone-template configuration mode, support was added for the **videoCapability** parameter, and a new value (**2**) for allowing limited access was added to the **settingsAccess** parameter.
12.4(9)T	Cisco Unified CME 4.0	Support was added for the **videoCapability** parameter, a new value (**2**) for allowing limited access was added for the **settingsAccess** parameter, and this command this command in ephone-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.
12.4(6)XE	Cisco Unified CME 4.0(2)	Support was added for the Cisco Unified IP Phone 7931G, and support was added for the **backlight** parameters, the **spanToPCPort** parameter, and the **webAccess** parameter.

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC4	Cisco Unified CME 4.0(3)	Support was added for the Cisco Unified IP Phone 7931G, and support was added for the **backlight** parameters, the **spanToPCPort** parameter, and the **webAccess** parameter.
12.4(11)T	Cisco Unified CME 4.0(3)	Modifications to this command were integrated into Cisco IOS Release 12.4(11)T.
12.4(11)XJ	Cisco Unified CME 4.1	Support was added for the Cisco Unified IP Phone 7921G.
12.4(15)T	Cisco Unified CME 4.1	This command was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

This command in telephony-service configuration mode modifies vendorConfig parameters in the configuration file for phones in a Cisco Unified CME system.

In Cisco Unified CME 4.0 and later versions, support for creating configuration files at a per-phone level was added and this command in ephone-template configuration mode creates an ephone-template of vendorConfig parameters which can be applied to an individual Cisco Unified IP phone in Cisco Unified CME.

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

The vendorConfig section of a configuration file is read by a phone's firmware when that Cisco Unified IP phone is booted. Only the vendorConfig parameters supported by the currently loaded firmware are available. The number and type of parameters may vary from one firmware version to the next.

The IP phone that downloads the configuration file will implement only those parameters that it can support and ignore configured parameters that it cannot implement. For example, a Cisco Unified IP Phone 7970G does not have a backlit display and cannot implement Backlight parameters regardless of whether they are configured.

After modifying the vendorConfig parameters, you must use the **create cnf** command to generate a new configuration file. After generating the configuration file, use the **reset** command to reboot the IP phones to be configured and download the configuration.

Use the **show telephony-service tftp-binding** command to view the SEP*.cnf.xml files that are associated with individual phones. The following example entry from a Sep*.conf.xml file disables the PC port on a phone:

```
<vendorConfig>
<pcPort>1</pcPort>
</vendorConfig>
```

Table 12 lists the basic vendorConfig parameters in alphabetical order.

Note Parameter names are word- and case-sensitive and must be typed exactly as shown.

Table 12 vendorConfig Parameter-Name and Parameter-Value Descriptions

Parameter Name and Value	Description
adminPassword *password*	(For Cisco Unified IP Phone 7921G only) Creates password for accessing the web interface on a phone • *passwor*d—String of up to 32 characters.
autoSelectLineEnable {**0** | **1**}	Enables and disables auto line selection. • **0**—Disabled. • **1**—Enabled (default).
backlightIdleTimeout *HH***:***MM*	Sets the length of time after which the backlighting of the IP phone displays will switch off again once the phone is inactive. This parameter is applicable only on the days specified using the **daysBacklightNotActive** parameter. This parameter does not affect the display during the period of time specified using the **backlightOnDuration** parameter. • Default is one hour (01:00).
backlightOnDuration *HH***:***MM*	Sets the length of time for which IP phone displays will be backlit. This parameter does not affect the display on the days specified using the **daysBacklightNotActive** parameter. • Default is 10 hours (10:00).
backlightOnTime *HH***:***MM*	Sets the time at which backlighting of the IP phone displays is switched on, using a 24-hour time format. This parameter does not affect the display on the days specified using the **daysBacklightNotActive** parameter. • Default is 07:30.
daysBacklightNotActive *number***,** *number...*	Sets days of the week on which backlighting of the IP phone displays is switched off unless there is user interaction with the IP phone. The *number* argument represents the days of the week numerically, starting with Sunday (1) and ending with Saturday (7). Each number must be separated with a comma. No spaces are allowed. • Default is no backlighting on Sun (1) and Sat (7).
daysDisplayNotActive *number***,** *number...*	Sets days of the week on which IP phone displays will be blank. The *number* argument represents the days of the week numerically, starting with Sunday (1) and ending with Saturday (7). Each number must be separated with a comma. No spaces are allowed. • Default is an inactive display on Sun (1) and Sat (7).
displayIdleTimeout *HH***:***MM*	Sets the length of time for which IP phone displays will remain active, starting from the last time that the phone was used. • Default is one hour (01:00).
displayOnDuration *HH***:***MM*	Sets the length of time for which IP phone displays will be active. • Default is 10 hours (10:00).

Table 12 vendorConfig Parameter-Name and Parameter-Value Descriptions (continued)

Parameter Name and Value	Description
displayOnTime *HH***:***MM*	Sets the time at which IP phone displays are activated, using a 24-hour time format. • Default is 07:30.
disableSpeaker {**true** | **false**}	Enables and disables the speakerphone. • **true**—Enabled (default). • **false**—Disabled.
disableSpeakerAndHeadset {**true** | **false**}	Enables and disables the speakerphone and headset. • **true**—Enabled (default). • **false**—Disabled.
*HH***:***MM*	Hour (*HH*) and minute (*MM*). You must enter all four characters. For example, 9:05 a.m. must be entered as 09:05.
forwardingDelay {**0** | **1**}	Enables and disables the activation of the IP phone's PC Ethernet switch port when the IP phone boots to prevent Ethernet traffic from interfering with the boot up process. • **0**—Disabled. • **1**—Enabled (default).
garp {**0** | **1**}	Enables and disables IP phone response to gratuitous Address Resolution Protocol (ARP) messages from the IP phone's Ethernet interface. • **0**—Disabled. • **1**—Enabled (default).
loadServer [*hostname* | *IPaddress*]	(For Cisco Unified IP Phone 7921G only) Directs phone to use an alternative TFTP server such as a local server to obtain firmware loads and upgrades. Using this parameter can help to reduce install times, particularly for upgrades over a WAN. The specified server must be running TFTP services and have the firmware file in the TFTP path. **Note** If the firmware file is not found, the firmware will not install. The phone will not be redirected to the TFTP server that is specified by the **option 150 ip** command. • *hostname*—Name of server from which the Ip phone must retrieve phone firmware. Maximum length: 256 characters • *Ip address*—IP address of server from which the IP phone must retrieve phone firmware. • To disable this command and redirect the phone to use the TFTP server that is specified by the **option 150 ip** command to obtain its load files and upgrades, use this parameter name without a value.

Table 12 vendorConfig Parameter-Name and Parameter-Value Descriptions (continued)

Parameter Name and Value	Description
pcPort {**0** | **1**}	Enables and disables the Ethernet switch port on the phone so the IP phone can have access to an Ethernet connection for a PC connection through the phone. • **0**—Enabled (default). • **1**—Disabled.
PushToTalkURL *url*	(Cisco Unified IP Phone 7921G only) Provisions uniform resource locator (URL) to be contacted for application services such as Push-To-Talk services. • *url*—URL as defined in RFC 2396. Maximum length: 256 characters
settingsAccess {**0** | **1** | **2**}	Enables and disables the Settings button on an IP phone. • **0**—Disabled. • **1**—Enabled (default). Phone user can modify features by using the Settings menu. • **2**—Restricted. Phone user is allowed to access User Preferences and volume settings only.
spanToPCPort {**0** | **1**}	Enables and disables the path between Ethernet switch port of an IP phone and a connection to a PC. • **0**—Enabled (default). • **1**—Disabled.

specialNumbers *number*[**,***number*...]	(For Cisco Unified IP Phone 7921G only) Identifies a number that can be dialed on a phone regardless of whether the phone is locked or unlocked. For example, in the United States, the 911 emergency number is a good special number candidate to be dialed without unlocking the phone.
	• *number*—Numerical string. Maximum length: 16 characters.
	• To identify more than one special number, separate the numbers with a comma (**,**). Do not include spaces between numbers.
	• The following example shows how to configure 411, 511, and 911 as special numbers:
	`Router(config)# telephony-service`
	`Router(config telephony-service)# service phone specialNumbers 411,511,911`

videoCapability {**0** | **1**}	Enables and disables video capability for all applicable IP phones associated with a Cisco Unified CME router or template. • **0**—Disabled. • **1**—Enabled (default).

Table 12 vendorConfig Parameter-Name and Parameter-Value Descriptions (continued)

Parameter Name and Value	Description
voiceVlanAccess {**0** | **1**}	Enables and disables spanning, which is the IP phone's access to voice VLAN of the PC to which the IP phone's Ethernet port is connected. • **0**—Enabled (default). • **1**—Disabled. **Note** For Cisco Unified IP Phone 7985, default is Disabled (1).
webAccess {**0** | **1** | **2**}	Enables and disables web access that allows phone users to configure settings and features on User Option web pages. • **0**—Enabled (default). • **1**—Disabled. • **2**—Read Only. For Cisco Unified IP Phone 7921G only. Phone user can only view User Option web pages and cannot modify settings and features on the pages. **Note** For Cisco Unified IP Phone 7921G, default is Read Only (2).
WLanProfile {**1** | **2** | **3** | **4**} {**0** | **1**}	(For Cisco Unified IP Phone 7921G only) Locks or unlocks a specific profile. • **0**—Locked (default). • **1**—Unlocked. User can modify a profile. • Repeat this command for each profile to be locked or unlocked.

Examples

The following example shows how to configure multiple **service phone** parameters. This configuration is applied only in as much as IP phone firmware supports each parameter.

```
Router(config)# telephony-service
Router(config-telephony)# service phone disableSpeaker true
Router(config-telephony)# service phone disableSpeakerAndHeadset true
Router(config-telephony)# service phone forwardingDelay 1
Router(config-telephony)# service phone garp 1
Router(config-telephony)# service phone pcPort 1
Router(config-telephony)# service phone voiceVlanAccess 0
Router(config-telephony)# service phone settingsAccess 1
Router(config-telephony)# service phone videoCapability 1
Router(config-telephony)# service phone daysDisplayNotActive 1,7
Router(config-telephony)# service phone displayOnTime 07:30
Router(config-telephony)# service phone displayOnDuration 10:00
Router(config-telephony)# service phone displayIdleTimeout 01:00
Router(config-telephony)# service phone daysBacklightNotActive 1,7
Router(config-telephony)# service phone backlightOnTime 07:30
Router(config-telephony)# service phone backlightOnDuration 10:00
Router(config-telephony)# service phone backlightIdleTimeout 01:00
Router(config-telephony)# create cnf-files
Router(config-telephony)# reset all
```

The following example shows how to set the default values for backlighting the phone display for all Cisco Unified IP phones with backlight capabilities in Cisco Unified CME.

```
Router(config)# telephony-service
Router(config-telephony)# service phone daysBacklightNotActive 1,7
Router(config-telephony)# service phone backlightOnTime 07:30
Router(config-telephony)# service phone backlightOnDuration 10:00
Router(config-telephony)# service phone backlightIdleTimeout 01:00
Router(config-telephony)# create cnf-files
Router(config-telephony)# reset all
```

The following example shows how to set the Backlighting parameters so that there is no backlighting of the phone display for all Cisco Unified IP phones with backlight capabilities until there is user interaction with the phone. The **backlightIdleTimeout** parameter is configured so that the backlight will switch off again after 60 seconds of inactivity.

```
Router(config)# telephony-service
Router(config-telephony)# service phone daysBacklightNotActive 1,2,3,4,5,6,7
Router(config-telephony)# service phone backlightOnTime 07:30
Router(config-telephony)# service phone backlightOnDuration 10:00
Router(config-telephony)# service phone backlightIdleTimeout 00.01
Router(config-telephony)# create cnf-files
Router(config-telephony)# reset all
```

The following example shows how to set the Display parameters so that the phone display for all Cisco Unified IP phones with luminous displays are blank on Sunday (1), Monday (2), and Saturday (7):

```
Router(config)# telephony-service
Router(config-telephony)# service phone daysDisplayNotActive 1,2,7
Router(config-telephony)# create cnf-files
Router(config-telephony)# reset all
```

The following example shows how to disable the PC port on an individual IP phone (ephone 15) using an ephone template:

```
Router(config)# ephone-template 8
Router(config-ephone-template)# service phone pcPort 1
Router(config-ephone-template)# exit
Router(config)# ephone 15
Router(config-ephone)# ephone-template 8
Router(config-ephone)# exit
Router(config)# telephony-service
Router(config-telephony)# create cnf-files
Router(config-telephony)# exit
Router(config)# ephone 15
Router(config-ephone)# reset
```

Related Commands

Command	Description
create cnf-files	Builds XML configuration files that set IP phone displays and functionality.
reset (telephony-service)	Performs a complete reboot of one or all phones associated with a Cisco Unified CME router.
show telephony-service tftp-binding	Displays the current configuration files accessible to IP phones.

session-server

To specify a session manager to manage and monitor Register and Subscribe messages during a feature-server session, use the **session-server** command in voice register pool or ephone-dn configuration mode. To return to the default, use the **no** form of this command.

session-server *session-server-tag*[**, ...***session-server-tag*]

no session-server *session-server-tag*

Syntax Description

session-server-tag	Unique identifier of previously configured session manager in Cisco Unified CME. Range: 1 to 8. Can contain up to eight session-server-tags separated by commas (**,**) when configured in voice register dn or ephone-dn configuration mode.

Command Default

Session manager is not assigned.

Command Modes

Ephone-dn configuration (ephone-dn)
Voice register dn configuration (voice-register-dn)
Voice register pool configuration (voice-register-pool)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XW1	Cisco Unified CME 4.2	This command was introduced.

Usage Guidelines

Cisco Unified CME 4.2 and later provides a general interface to work with external feature servers, such as the Cisco Unified Contact Center Express application on a Cisco CRS, for managing and monitoring Register and Subscribe messages during a feature-server session.

Use this command in voice register pool configuration mode to specific that Register and Subscribe messages for a route point for an external feature server must contain a Cisco-referenceID field. Registration or subscription will be granted only for the specified route point.

Use this command in ephone-dn and voice register dn configuration modes to specify that Subscribe messages for a directory number must contain a Cisco-referenceID field. Registration or subscription will be granted only for the specified directory number.

Before using this command, configure a session manager by using the **voice register session-server** command.

A route point for each external feature server for which Register and Subscribe messages are to be managed by this session manager must already be configured as a SIP endpoint.

A directory number for which Subscribe messages are to be monitored by this session manager must already be configured in Cisco Unified CME.

Each route point can be managed by only one session manager.

Each session manager can manage multiple route points.

Each directory number can be monitored by up to eight session managers.

Each session manager can subscribe for multiple directory numbers.

Examples

The following example shows the configuration for specifying that session manager 1 can control a SIP endpoint (voice register pool) that is configured for an external feature server, such as Cisco Unified CCX on a Cisco CRS platform:

```
voice register pool 1
 session-server 1
```

The following example shows the configuration specifying which session managers can monitor Register and Subscribe messages to directory numbers associated with Cisco Unified CCX agent phones. Notice that several session managers (1, 3, 5, and 7) can subscribe for both directory numbers:

```
ephone-dn 1
 session-server 1,2,3,4,5,6,7,8
.
ephone-dn 2
 session-server 1,3,5,7
```

Related Commands

Command	Description
ephone-dn	Enters ephone-dn configuration mode to define a directory number for an SCCP phone.
voice register dn	Enters voice register dn configuration mode to define a directory number for a SIP phone, intercom line, voice port, or a message-waiting indicator (MWI).
voice register pool	Enters voice register pool configuration mode to set device-specific parameters for a a SIP device.
voice register session-server	Enters voice register session configuration mode for the purposes of configuring a session manager

session-transport

To specify the transport layer protocol that a SIP phone uses to connect to Cisco Unified CME, use the **session-transport** command in voice register pool or voice register template configuration mode. To reset to the default value, use the **no** form of this command.

session-transport {**tcp** | **udp**}

no session-transport

Syntax Description

tcp	Transmission Control Protocol (TCP) is used.
udp	User Datagram Protocol (UDP) is used. This is the default.

Command Default

UDP is the default protocol.

Command Modes

Voice register pool configuration
Voice register template configuration

Command History

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

Usage Guidelines

This command sets the transport layer protocol parameter in the phone's configuration file.

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

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

Note Although this command is not supported for the Cisco Unified IP Phone 7905, 7912, 7940, or 7960, it can be used to assign TCP as the session transport type for these phones. If TCP is selected for an unsupported phone using this command, calls to that phone will not complete successfully. The phone can originate calls but it uses UDP, although TCP has been assigned.

Examples

The following example sets the transport layer protocol to TCP for SIP phone 10:

```
Router(config)# voice register pool 10
Router(config-register-pool)# session-transport tcp
```

Related Commands

Command	Description
create profile	Generates the configuration profile files required for SIP phones.
show sip-ua status	Displays the status of SIP call service on a SIP gateway.

show capf-server

To display CAPF server configuration and session information, use the **show capf-server** command in privileged EXEC configuration mode.

show capf-server {**auth-string** | **sessions** | **summary**}

Syntax Description

auth-string	Display authentication strings for ephones.
sessions	Display information about active CAPF sessions.
summary	Display CAPF server configuration details.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used with Cisco Unified CME phone authentication.

Examples

The following example output displays CAPF server parameters:

```
Router# show capf-server summary

CAPF Server Configuration Details
        Trustpoint for TLS With Phone: cmeserver
        Trustpoint for CA operation: iosra
        Source Address: 10.1.1.1
        Listening Port: 3804
        Phone Key Size: 1024
        Phone KeyGen Retries: 100
        Phone KeyGen Timeout: 120 minutes
        Device Authentication Mode: Auth-String
```

The following example output displays the authentication strings that have been defined for the phones with the listed MAC addresses:

```
Router# show capf-server auth-string

Authentication Strings for configured Ephones
Mac-Addr        Auth-String
--------        -----------
000CCE3A817C    7012
001121116BDD    922
000D299D50DF    9182
000ED7B10DAC    3114
000F90485077    3328
0013C352E7F1    0678
```

The following example output displays active sessions between phones (identified by their MAC addresses) and the CAPF server. The phone ID field lists standard phone identifications, which include the letters "SEP" plus the MAC addresses of the phones. Table 13 defines the session states that can appear in the output.

```
Router# show capf-server sessions

Active CAPF Sessions
Phone ID               State
SEP000CCE3A817C        AWAIT-KEYGEN-RES
```

Table 13 show capf-server sessions State Descriptions

State	Description
IDLE	Phone is idle.
AWAIT AUTH RES	A TLS connection was established on the TCP port that is specified in the configuration file. After a successful handshake verified the server certificate, a dialog was started between the CAPF server and the phone's CAPF client. The server has challenged the phone by sending an authentication request and is waiting for a response.
AWAIT KEYGEN RESP	Phone authentication was successful. The CAPF server has sent a key generation request message to the phone and is waiting for a response.
AWAIT ENCRYPT MSG RESP	A key has been generated and the CAPF has used the phone's public key to start the enrollment process with PKI. The CAPF sent an encrypt-message request to the phone and is waiting for a response.
AWAIT CA RESP	The phone has signed the received message using its private key and the CAPF has continued the enrollment process. PKI has forwarded the certificate request to the CA and is waiting for a response.
AWAIT STORE CERT RESP	Upon receiving an certificate issued from the CA, the CAPF has sent a store-certificate request message to the phone. The store-certificate request contains the certificate to be written to the phone's flash memory. The CAPF is waiting for a store-certificate response message to confirm that the certificate has been stored.

show credentials

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

show credentials

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(14)T	Cisco SRST 3.3	This command was introduced for Cisco SRST.
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced for Cisco Unified CME.
12.4(9)T	Cisco Unified CME 4.0	This command for Cisco Unified CME was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Cisco Unified CME

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

Cisco Unified SRST

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

Examples

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

```
Router# show credentials

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

Table 14 describes the fields in the sample output.

Table 14 show credentials Field Descriptions

Field	Description
Credentials IP	Cisco Unified CME—IP address where the CTL provider is configured. Cisco Unified SRST—The specified IP address where certificates from Cisco Unified CallManager to the SRST router are received.
Credentials PORT	Cisco Unified CME—TCP port for credentials service communication. Default is 2444. Cisco Unified SRST—The port to which the SRST router connects to receive messages from the Cisco Unified IP phones. The port number is from 2000 to 9999. The default port number is 2445.
Trustpoint	Cisco Unified CME—CTL provider trustpoint label that will be used for TLS sessions with the CTL client. Cisco Unified SRST—The name of the trustpoint that is associated with the credentials service between the Cisco Unified CallManager client and the SRST router.

Related Commands

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

show ctl-client

To display information about the certificate trust list (CTL) client, use the **show ctl-client** command in privileged EXEC configuration mode.

show ctl-client

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)TC	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used with Cisco Unified CME phone authentication.

Examples

The following example displays trustpoints and IP addresses known to the CTL client.

```
Router# show ctl-client

CTL Client Information
-----------------------------
        SAST 1 Certificate Trustpoint: cmeserver
        SAST 1 Certificate Trustpoint: sast2
        List of Trusted Servers in the CTL
                CME     10.1.1.1        cmeserver
                TFTP    10.1.1.1        cmeserver
                CAPF    10.1.1.1        cmeserver
```

show ephone

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

show ephone [*mac-address* | *phone-type*]

Syntax Description

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

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco ITS 1.0 Cisco SRST 1.0	This command was introduced on the Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(2)XT	Cisco ITS 2.0 Cisco SRST 2.0	This command was implemented on the Cisco 1750 and Cisco 1751.
12.2(8)T	Cisco ITS 2.0 Cisco SRST 2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	Cisco ITS 2.0 Cisco SRST 2.0	This command was implemented on the Cisco 2600XM and Cisco 2691.
12.2(11)T	Cisco ITS 2.01 Cisco SRST 2.01	The **ata** keyword was added and this command was implemented on the Cisco 1760.
12.2(11)YT	Cisco ITS 2.1 Cisco SRST 2.1	The **7914** keyword was added.
12.2(15)ZJ	Cisco CME 3.0 Cisco SRST 3.0	The **7902**, **7905**, and **7912** keywords were added.
12.3(7)T	Cisco CME 3.1 Cisco SRST 3.1	The **7920** and **7936** keywords were added.
12.3(11)XL	Cisco CME 3.2.1 Cisco SRST 3.2.1	The **7970** keyword was added.
12.3(14)T	Cisco CME 3.3 Cisco SRST 3.3	The **7971** keyword was added, and this command was integrated into Cisco IOS Release 12.3(14)T.
12.4(4)XC	Cisco Unified CME 4.0 Cisco Unified SRST 4.0	The **7911**, **7941**, **7941GE**, **7961**, and **7961GE** keywords were added.
12.4(9)T	Cisco Unified CME 4.0 Cisco Unified SRST 4.0	The **7911**, **7941**, **7941GE**, **7961**, and **7961GE** keywords were integrated into Cisco IOS Release 12.4(9)T.
12.4(6)XE	Cisco Unified CME 4.0(2)	The **7931** keyword was added for Cisco Unified CME.
12.4(4)XC4	Cisco Unified CME 4.0(3)	The **7931** keyword was added for Cisco Unified CME.
12.4(11)T	Cisco Unified CME 4.0(3)	The **7931** keyword for Cisco Unified CME was integrated in Cisco IOS Release 12.4(11)T.
12.4(11)XJ2	Cisco Unified CME 4.1	The **7921** keyword was added.
12.4(15)T	Cisco Unified CME 4.1	The **7921** keywordwas integrated into Cisco IOS Release 12.4(15)T.

Examples

Significant fields in the output from this command are described in Table 13.

The following sample output shows general information for registered phones:

```
Router# show ephone

ephone-1 Mac:0003.E3E7.F627 TCP socket:[2] activeLine:0 REGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
IP:10.0.0.2 51671 Telecaster 7940 keepalive 28 max_line 2
button 1: dn 1 number 4444
ephone-2 Mac:0030.94C3.F43A TCP socket:[1] activeLine:0 REGISTERED
```

```
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
IP:10.0.0.3 50094 Telecaster 7960 keepalive 28 max_line 6
button 1: dn 3 number 5555
ephone-3 Mac:0003.6B40.99DA TCP socket:[3] activeLine:0 REGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
IP:10.2.168.200 51879 Telecaster 7960 keepalive 28 max_line 6
button 1: dn 2 number 3333
```

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

```
Router# show ephone 0003.E3E7.F627

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

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

```
Router# show ephone 7914

ephone-2 Mac:0007.0EA6.39F8 TCP socket:[2] activeLine:0 REGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
IP:10.2.205.206 49278 Telecaster 7960  sub=4 keepalive 2723 max_line 34
button 1: dn 21 number 60021 CH1 IDLE
button 2: dn 22 number 60022 CH1 IDLE
button 7: dn 11 number 60011 CH1 IDLE      monitor-ring
button 8: dn 12 number 60012 CH1 IDLE      monitor-ring
button 9: dn 13 number 60013 CH1 IDLE      monitor-ring
button 10: dn 14 number 60014 CH1 IDLE      monitor-ring
button 11: dn 15 number 60015 CH1 IDLE      monitor-ring
button 12: dn 16 number 60016 CH1 IDLE      monitor-ring
button 13: dn 17 number 60017 CH1 IDLE      monitor-ring
button 14: dn 18 number 60018 CH1 IDLE      monitor-ring
button 15: dn 19 number 60019 CH1 IDLE      monitor-ring
button 16: dn 20 number 60020 CH1 IDLE      monitor-ring
button 17: dn 39 number 60039 CH1 IDLE      CH2 IDLE      monitor-ring
button 18: dn 40 number 60040 CH1 IDLE      CH2 IDLE      monitor-ring
button 19: dn 23 number 60023 CH1 IDLE      monitor-ring
button 20: dn 24 number 60024 CH1 IDLE      monitor-ring
button 21: dn 25 number 60025 CH1 IDLE      monitor-ring
button 22: dn 26 number 60026 CH1 IDLE      monitor-ring
button 23: dn 27 number 60027 CH1 IDLE      monitor-ring
button 24: dn 28 number 60028 CH1 IDLE      monitor-ring
button 25: dn 29 number 60029 CH1 IDLE      monitor-ring
button 26: dn 30 number 60030 CH1 IDLE      monitor-ring
button 27: dn 31 number 60031 CH1 IDLE      CH2 IDLE      monitor-ring
button 28: dn 32 number 60032 CH1 IDLE      CH2 IDLE      monitor-ring
button 29: dn 33 number 60033 CH1 IDLE      CH2 IDLE      monitor-ring
button 30: dn 34 number 60034 CH1 IDLE      CH2 IDLE      monitor-ring
button 31: dn 35 number 60035 CH1 IDLE      CH2 IDLE      monitor-ring
button 32: dn 36 number 60036 CH1 IDLE      CH2 IDLE      monitor-ring
button 33: dn 37 number 60037 CH1 IDLE      CH2 IDLE      monitor-ring
button 34: dn 38 number 60038 CH1 IDLE      CH2 IDLE      monitor-ring
```

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

```
Router# show ephone 7910

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

Table 13 describes significant fields in the output.

Table 15 show ephone Field Descriptions

Field	Description
Active Call	An active call is in progress.
activeLine	Line (button) on the phone that is in use. Zero indicates that no line is in use.
auto-dial *number*	Intercom extension that automatically dials *number*.
button *number*: dn *number*	Phone button number and the extension (ephone-dn) dn-tag number associated with that button.
bytes	Total number of voice data bytes sent or received by the phone.
Called Dn, Calling Dn	Ephone-dn tag numbers of the called and calling ephone-dn. Set to -1 if the call is not to or from an ephone-dn, or if there is no active call.
cfa *number*	Call-forward-all to *number* is enabled for this extension.
CH1 CH2	Status of channel 1 and, if this is a dual-line ephone-dn, the status of channel 2.
debug	1 indicates that debug for the phone is enabled. 0 indicates that debug is disabled.
DnD	Do Not Disturb is set on this phone.
DP tag	Not used.
ephone-*number*	Unique sequence number used to identify this phone during configuration (phone-tag).
IP	Assigned IP address of the Cisco Unified IP phone.
Jitter	Amount of variation (in milliseconds) of the time interval between voice packets received by the Cisco Unified IP phone.
keepalive	Number of keepalive messages received from the Cisco Unified IP phone by the router.
Latency	Estimated playout delay for voice packets received by the Cisco Unified IP phone.
line *number*	Button number on an IP phone. Line 1 is the button nearest the top of the phone.
Lost	Number of voice packets lost, as calculated by the Cisco Unified IP phone, on the basis of examining voice packet time-stamp and sequence numbers during playout.
Mac	MAC address.
Max Conferences	Maximum number of allowable conference calls and number of active conference calls.

Table 15 show ephone Field Descriptions (continued)

Field	Description
max_line *number*	Maximum number of line buttons that can be configured on this phone.
mediaActive	1 indicates that an active conversation is in progress. 0 indicates that no conversation is ongoing.
monitor-ring	This button is set up as a monitor button.
number	Telephone or extension number associated with the Cisco Unified IP phone button and its dn-tag.
offhook	1 indicates that the phone is off-hook. 0 indicates that the phone is on-hook.
overlay	This button contains an overlay set. Use **show ephone overlay** to display the contents of overlay sets.
paging	1 indicates that the phone has received an audio page. 0 indicates that the phone has not received an audio page.
paging-dn	Ephone-dn that is dedicated for receiving audio pages on this phone. The paging-dn number is the number of the paging set to which this phone belongs.
Password	Authentication string that the phone user types when logging in to the web-based Cisco Unified CME GUI.
Port	Port used for TAPI transmissions.
REGISTERED	The Cisco Unified IP phone is active and registered. Alternative states are UNREGISTERED (indicating that the connection to the Cisco Unified IP phone was closed in a normal manner) and DECEASED (indicating that the connection to the Cisco Unified IP phone was closed because of a keepalive timeout).
reset	Pending reset.
reset_sent	Request for reset has been sent to the Cisco Unified IP phone.
ringing	1 indicates that the phone is ringing. 0 indicates that the phone is not ringing.
Rx Pkts	Number of received voice packets.
silent-ring	Silent ring has been set on this button and extension.
socket	TCP socket number used to connect to IP phone.
speed dial *speed-tag:digit-string label-text*	This button is a speed-dial button, assigned to the speed-dial sequence number *speed-tag*. It dials *digit-string* and displays the text *label-text* next to the button.
sub=3, sub=4	Subtype 3 means that one Cisco Unified IP Phone 7914 Expansion Module is attached to the main Cisco Unified IP Phones 7960 and 7960G, and subtype 4 means that two are attached.
Tag *number*	Dn-tag number, the unique sequence number that identifies an ephone-dn during configuration, followed by the type of ephone-dn it is.
TAPI Client IP Address	IP address of the PC running the TAPI client.
TCP socket	TCP socket number used to communicate with the Cisco Unified IP phone. This can be correlated with the output of other debug and show commands.

Table 15 show ephone Field Descriptions (continued)

Field	Description
Telecaster *model-number*	Type and model of the Cisco Unified IP phone. This information is received from the phone during its registration with the router.
Tx Pkts	Number of transmitted voice packets.
Username	Username that the phone user types when logging in to the web-based Cisco Unified CME GUI.

Related Commands

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

show ephone attempted-registrations

To display the log of ephones that unsuccessfully attempt to register with Cisco Unified CME, use the **show ephone attempted-registrations** command in privileged EXEC mode.

show ephone attempted-registrations

Syntax Description

This command has no keywords or arguments.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

The **no auto-reg-ephone** command blocks the automatic registration of ephones whose MAC addresses are not explicitly listed in the configuration. When automatic registration is blocked, Cisco Unified CME records the MAC addresses of phones that attempt to register but cannot because they are blocked.

Use the **show ephone attempted-registrations** command to view the list of phones that have attempted to register but have been blocked. The **clear telephony-service ephone-attempted-registrations** command clears the list.

Examples

The following example displays ephones that unsuccessfully attempted to register with Cisco Unified CME:

```
Router# show ephone attempted-registrations

Attempting Mac address:

Num    Mac Address         DateTime                          DeviceType
-----------------------------------------------------------------------------

1      C863.8475.5417      22:52:05 UTC Thu Apr 28 2005      SCCP Gateway (AN)
2      C863.8475.5408      22:52:05 UTC Thu Apr 28 2005      SCCP Gateway (AN)
.....
25     000D.28D7.7222      22:26:32 UTC Thu Apr 28 2005      Telecaster 7960
26     000D.BDB7.A9EA      22:25:59 UTC Thu Apr 28 2005      Telecaster 7960
...
47     C863.94A8.D40F      22:52:17 UTC Thu Apr 28 2005      SCCP Gateway (AN)
48     C863.94A8.D411      22:52:18 UTC Thu Apr 28 2005      SCCP Gateway (AN)

49     C863.94A8.D400      22:52:15 UTC Thu Apr 28 2005      SCCP Gateway (AN)
```

Table 16 describes the significant fields shown in the display.

Table 16 show ephone attempted-registrations Field Descriptions

Field	Description
Num	Index number.
Mac Address	MAC address of the ephone.
DateTime	Date and time that the attempt to register was made.
DeviceType	Type of ephone.

Related Commands

Command	Description
auto-reg-ephone	Enables automatic registration of ephones with the Cisco Unified CME system.
clear telephony-service ephone-attempted-registrations	Empties the log of ephones that unsuccessfully attempt to register with Cisco Unified CME.

show ephone cfa

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

show ephone cfa

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

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

Examples

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

```
Router# show ephone cfa

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

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

Related Commands

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

show ephone dn

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

show ephone dn [*dn-tag*]

Syntax Description

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

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

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

Examples

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

```
Router# show ephone dn 5

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

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

Related Commands

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

show ephone dnd

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

show ephone dnd

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

This command does not apply to Cisco Unified SRST.

Examples

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

```
Router# show ephone dnd

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

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

Related Commands

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

show ephone login

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

show ephone login

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

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

Examples

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

```
Router# show ephone login

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

Table 17 describes significant fields in this output.

Table 17 show ephone login Field Descriptions

Field	Description
ephone *phone-tag*	Phone identified with its unique phone-tag sequence number.

Table 17 show ephone login Field Descriptions

Field	Description
Pin enabled	TRUE indicates that a PIN has been defined for this phone. FALSE indicates that no PIN has been defined for this phone.
Logged-in	TRUE indicates that a phone user is currently logged in on this phone. FALSE indicates that no phone user is currently logged in on this phone.

Related Commands

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

show ephone offhook

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

show ephone offhook

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

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

Examples

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

```
Router# show ephone offhook

No ephone in specified type/condition.
```

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

```
Router# show ephone offhook

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

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

```
Router# show ephone offhook

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

```
button 5:dn 75 number 49943  auto dial 49943 CH1 IDLE
speed dial 1:57514 marketing
Active Call on DN 9 chan 1 :59943 10.23.84.71 22926 to 172.30.131.129 2000 via
172.30.151.1
G711Ulaw64k  160 bytes no vad
Tx Pkts 0 bytes 0 Rx Pkts 0 bytes 0 Lost 0
Jitter 0 Latency 0 callingDn -1 calledDn -1 (media path callID 19288 srcCallID 1
9289)
Username:user1 Password:newuser
```

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

Related Commands

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

show ephone overlay

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

show ephone overlay

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

This command does not apply to Cisco Unified SRST.

Examples

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

```
Router# show ephone overlay

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

The **show ephone** command describes significant fields in this output. Table 18 describes a field that is not in that table.

Table 18 show ephone overlay Field Descriptions

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

Related Commands

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

show ephone phone-load

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

show ephone phone-load

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

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

Examples

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

```
Router# show ephone phone-load

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

Table 19 describes significant fields in this output.

Table 19 show ephone phone-load Field Descriptions

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

Related Commands

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

show ephone registered

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

show ephone registered

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

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

Examples

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

```
Router# show ephone registered

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

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

Related Commands

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

show ephone remote

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

show ephone remote

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

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

Examples

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

```
Router# show ephone remote

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

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

Related Commands

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

show ephone ringing

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

show ephone ringing

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

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

Examples

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

```
Router# show ephone ringing

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

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

Related Commands

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

show ephone summary

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

show ephone summary

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

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

Examples

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

```
Router# show ephone summary

ephone-1 Mac:0030.94C3.37CB TCP socket:[-1] activeLine:0 REGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 debug:0
IP:10.1.1.1 Telecaster 7910  keepalive 45  1:1
sp1:5002 sp2:5003

ephone-2 Mac:0030.94C3.F96A TCP socket:[-1] activeLine:0 REGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 debug:0
IP:10.1.1.2 Telecaster 7960  keepalive 45  1:2 2:3 3:4
sp1:5004 sp2:5001

ephone-3 Mac:0030.94C3.F946 TCP socket:[-1] activeLine:0 REGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 debug:0
IP:10.2.1.2 Telecaster 7960  keepalive 59

ephone-4 Mac:0030.94C3.F43A TCP socket:[-1] activeLine:0 REGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 debug:0
IP:10.2.1.1 Telecaster 7960  keepalive 59

Max 48, Registered 1, Unregistered 0, Deceased 0, Sockets 1
```

```
Max Conferences 4 with 0 active (4 allowed)
Skinny Music On Hold Status
Active MOH clients 0 (max 72), Media Clients 0
No MOH file loaded
```

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

Related Commands

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

show ephone tapiclients

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

show ephone tapiclients

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

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

Examples

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

```
Router# show ephone tapiclients

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

Username:userB status:REGISTERED Socket :[5]
 Tapi Client IP address: 192.168.1.5  Port:2295
```

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

Related Commands

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

show ephone telephone-number

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

show ephone telephone-number *number*

Syntax Description

number	Telephone number that is associated with an ephone.

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

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

Examples

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

```
Router# show ephone telephone-number 91400

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

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

Related Commands

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

show ephone unregistered

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

show ephone unregistered

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

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

Examples

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

```
Router# show ephone unregistered

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

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

Related Commands

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

show ephone-dn

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

show ephone-dn [*dn-tag*]

Syntax Description

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

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco CME 1.0 Cisco SRST 1.0	This command was introduced on the Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(2)XT	Cisco CME 2.0 Cisco SRST 2.0	This command was implemented on the Cisco 1750 and Cisco 1751.
12.2(8)T	Cisco CME 2.0 Cisco SRST 2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	Cisco CME 2.0 Cisco SRST 2.0	This command was implemented on the Cisco 2600XM and Cisco 2691.
12.2(11)T	Cisco CME 2.01 Cisco SRST 2.01	This command was implemented on the Cisco 1760.
12.3(4)T	Cisco CME 3.0 Cisco SRST 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Examples

Cisco Unified CME

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

```
Router# show ephone-dn

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

```
 Non Linear Mute is disabled
 Non Linear Threshold is -21 dB
 Music On Hold Threshold is Set to -38 dBm
 In Gain is Set to 0 dB
 Out Attenuation is Set to 0 dB
 Echo Cancellation is enabled
 Echo Cancellation NLP mute is disabled
 Echo Cancellation NLP threshold is -21 dB
 Echo Cancel Coverage is set to 8 ms
 Playout-delay Mode is set to adaptive
 Playout-delay Nominal is set to 60 ms
 Playout-delay Maximum is set to 200 ms
 Playout-delay Minimum mode is set to default, value 40 ms
 Playout-delay Fax is set to 300 ms
 Connection Mode is normal
 Connection Number is not set
 Initial Time Out is set to 10 s
 Interdigit Time Out is set to 10 s
 Call Disconnect Time Out is set to 60 s
 Ringing Time Out is set to 180 s
 Wait Release Time Out is set to 30 s
 Companding Type is u-law
 Region Tone is set for US
 Station name None, Station number 91400

 Caller ID Info Follows:
 Standard BELLCORE
 Translation profile (Incoming):
 Translation profile (Outgoing):

 Digit Duration Timing is set to 100 ms

50/0/2 CH1 IDLE      CH2 IDLE

EFXS 50/0/2 Slot is 50, Sub-unit is 0, Port is 2
 Type of VoicePort is EFXS
 Operation State is DORMANT
 Administrative State is UP
 No Interface Down Failure
 Description is not set
 Noise Regeneration is enabled
 Non Linear Processing is enabled
 Non Linear Mute is disabled
 Non Linear Threshold is -21 dB
 Music On Hold Threshold is Set to -38 dBm
 In Gain is Set to 0 dB
 Out Attenuation is Set to 0 dB
 Echo Cancellation is enabled
 Echo Cancellation NLP mute is disabled
 Echo Cancellation NLP threshold is -21 dB
 Echo Cancel Coverage is set to 8 ms
 Playout-delay Mode is set to adaptive
 Playout-delay Nominal is set to 60 ms
 Playout-delay Maximum is set to 200 ms
 Playout-delay Minimum mode is set to default, value 40 ms
 Playout-delay Fax is set to 300 ms
 Connection Mode is normal
 Connection Number is not set
 Initial Time Out is set to 10 s
 Interdigit Time Out is set to 10 s
 Call Disconnect Time Out is set to 60 s
 Ringing Time Out is set to 180 s
 Wait Release Time Out is set to 30 s
 Companding Type is u-law
```

```
 Region Tone is set for US
 Station name None, Station number 91450

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

Cisco Unified SRST

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

```
Router# show ephone-dn 7

50/0/7 INVALID

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

 Caller ID Info Follows:
 Standard BELLCORE

 Voice card specific Info Follows:
 Digit Duration Timing is set to 100 ms
```

Table 20 describes significant fields in the output from this command.

Table 20 show ephone-dn Field Descriptions

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

Table 20 show ephone-dn Field Descriptions (continued)

Field	Description
Initial Time Out	Amount of time the system waits for an initial input digit from the caller.
Interdigit Time Out	Amount of time the system waits for a subsequent input digit from the caller.
Last 64 far-end disconnect cause codes	See the Mappings of PSTN Cause Codes to SIP Event table for a list of public switch telephone network (PSTN) cause codes that can be sent as an ISDN cause information element (IE) and the corresponding Session Interface Protocol (SIP) event.
Latency	The final voice packet receive latency reported by the IP phone at the end of the call.
Lost	Number of lost packets.
Music On Hold Threshold	Not applicable to the Cisco IP phone.
No Interface Down Failure	State of the interface.
Noise Regeneration	Not applicable to the Cisco IP phone.
Non Linear...	Not applicable to the Cisco IP phone.
Operation State	Operational state of the voice port.
Out Attenuation	Not applicable to the Cisco IP phone.
outgoing	The number of outgoing call attempts.
Playout-delay Maximum	Not applicable to the Cisco IP phone.
Playout-delay...	Not applicable to the Cisco IP phone.
Port	Port number for the interface associated with the voice interface card.
Region Tone	Not applicable to the Cisco IP phone.
ring	The number of calls that were disconnected by the far-end device when the local IP phone was in the ringing state (for example, if the call was not answered and the caller hung up).
Ringing Time Out	Duration, in seconds, for which ringing is to continue if a call is not answered. Set with the **timeouts ringing** command.
Rx Pkts, bytes	Number of packets and bytes received during the current or last call.
Signal Level to phone, peak	For G.711 calls only, this parameter indicates the most recent voice signal level in the voice IP packets sent from the router to the IP phone. This parameter is valid only for VoIP or PSTN G.711 calls to the IP phones. This parameter is not valid for calls between local IP phones, or calls that use codecs other than G.711. The peak field indicates the peak signal level seen during the entire call.
Slot	Slot used in the voice interface card for this port.
Station name	Station name.
Station number	Station number.
Sub-unit	Subunit used in the voice interface card for this port.
Tx Pkts, bytes	Number of packets and bytes transmitted during the current call or last call.

Table 20 show ephone-dn Field Descriptions (continued)

Field	Description
Type of VoicePort	Voice port type.
VAD	Voice activity detection.
Voice card specific info	Information specific to the voice card.
VPM STATE	State indication for the VPM software component.
VTSP STATE	State indication for the VTSP software component.
Wait Release Time Out	Time that a voice port stays in the call-failure state while the router sends a busy tone, reorder tone, or out-of-service tone to the port.

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

Table 21 Mappings of PSTN Cause Codes to SIP Events

PSTN Cause Code	Description	SIP Event
1	Unallocated number	410 Gone
3	No route to destination	404 Not found
16	Normal call clearing	BYE
17	User busy	486 Busy here
18	No user responding	480 Temporarily unavailable
19	No answer from the user	
21	Call rejected	603 Decline
22	Number changed	302 Moved temporarily
27	Destination out of order	404 Not found
28	Address incomplete	484 Address incomplete
29	Facility rejected	501 Not implemented
31	Normal unspecified	404 Not found
34	No circuit available	503 Service unavailable
38	Network out of order	
41	Temporary failure	
42	Switching equipment congestion	
44	Requested channel not available	
47	Resource unavailable	
55	Incoming class barred within CUG	603 Decline
57	Bearer capability not authorized	501 Not implemented
58	Bearer capability not presently available	

Table 21 Mappings of PSTN Cause Codes to SIP Events (continued)

PSTN Cause Code	Description	SIP Event
63	Service or option unavailable	503 Service unavailable
65	Bearer cap not implemented	501 Not implemented
79	Service or option not implemented	
87	User not member of CUG	603 Decline
88	Incompatible destination	400 Bad Request
95	Invalid message	
102	Recover on timer expiry	408 Request timeout
111	Protocol error	400 Bad request
127	Interworking unspecified	500 Internal server error
Any code other than those listed above		500 Internal server error

Related Commands

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

show ephone-dn callback

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

show ephone-dn callback

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

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

Examples

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

```
Router# show ephone-dn callback

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

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

```
Router# show ephone-dn callback

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

Significant fields in the output from this command are described in Table 22.

Table 22 show ephone-dn callback Field Descriptions

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

Related Commands

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

show ephone-dn conference

To display information about ad hoc and meet-me conferences in a Cisco Unified CallManager Express (Cisco Unified CME) environment, use the **show ephone-dn conference** command in privileged EXEC mode.

show ephone-dn conference [**ad-hoc** | **meetme** | **number** *number*]

Syntax Description

ad-hoc	(Optional) Ad hoc conferences.
meetme	(Optional) Meet-me conferences.
number *number*	(Optional) Conference telephone or extension number.

Command Modes

Privileged EXEC

Command History

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

Examples

The following sample output displays information for the 1234 conference number. There are three directory numbers and six inactive parties.

```
Router# show ephone-dn conference number 1234

type   active inactive  number
==================================
Meetme    0       6      1234
DN tags: 31, 32, 33
```

Table 23 describes the significant fields shown in the display.

Table 23 show ephone-dn conference Field Descriptions

Field	Description
active	Number of active parties in the conference.
DN tags	Directory numbers (DNs) in the conference.
inactive	Number of inactive parties in the conference.
number	Conference telephone or extension number.
type	Type of conference: meet-me or ad hoc.

Related Commands

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

show ephone-dn loopback

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

show ephone-dn loopback

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

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

Examples

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

```
Router# show ephone-dn loopback

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

Significant fields in the output from this command are described in Table 24, in alphabetical order.

Table 24 show ephone-dn loopback Field Descriptions

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

Related Commands

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

show ephone-dn park

To display information about call-park slots in the system, use the **show ephone-dn park** command in privileged EXEC mode.

show ephone-dn park

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.3(7)T	This command was introduced.

Examples

The following example shows information for a single call-park slot that uses an ephone-dn identifier of 50 and an extension number of 1560.

```
Router# show ephone-dn park

DN 50 (1560) park-slot state IDLE
Notify to () timeout 15 limit 20
```

Table 25 describes the significant fields shown in the display.

Table 25 show ephone-dn park Field Descriptions

Field	Description
DN	Ephone-dn tag (identifier) number for the call-park slot.
(1560)	Extension number associated with the call-park slot.
park-slot state	Whether the call-park slot is in use or idle.
Notify to ( )	Extension that has been specified for notification. Empty parentheses indicate that no extension was specified in the configuration.
timeout	Number of seconds between reminder rings, in seconds.
limit	Number of reminder rings before a call parked at this slot is disconnected.

Related Commands

Command	Description
park-slot	Creates a floating extension (ephone-dn) at which calls can be temporarily held (parked).

show ephone-dn statistics

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

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

Syntax Description

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

Command Modes

Privileged EXEC

Command History

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

Examples

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

```
Router# show ephone-dn statistics

Total Calls 103
Stats may appear to be inconsistent for conference or shared line cases

DN 1 chan 1 incoming 36 answered 21 outgoing 60 answered 30 busy 6
Far-end disconnect at:connect 29 alert 18 hold 7 ring 15
Last 64 far-end disconnect cause codes
17 17 17 17 17 17 16 16 16 16 16 16 16 16 16 16
16 16 16 16 65 16 65 65 65 65 16 65 65 65 16 16
16 16 16 16 16 16 16 16 16 16 16 16 16 65 47 65
47 47 16 16 16 16 16 16 16 16 16 16 16 16 16 16
local phone on-hook

DN 1 chan 1 (95011) voice quality statistics for last call
Call Ref 103 called 91500 calling 95011
Total Tx Pkts 0 bytes 0 Rx Pkts 0 bytes 0 Lost 0
Final Jitter 30 Latency 0 Lost 0
Signal Level to phone 0 (-78 dB) peak 0 (-78 dB)
Packets counted by router 0

DN 1 chan 2 incoming 0 answered 0 outgoing 1 answered 0 busy 0
Far-end disconnect at:connect 0 alert 0 hold 0 ring 0
Last 64 far-end disconnect cause codes
0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
```

```
0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
local phone on-hook

DN 1 chan 2 (95011) voice quality statistics for last call
Call Ref 86 called  calling
Total Tx Pkts 0 bytes 0 Rx Pkts 0 bytes 0 Lost 0
Final Jitter 0 Latency 0 Lost 0
Signal Level to phone 0 (-78 dB) peak 0 (-78 dB)
Packets counted by router 0

DN 3 chan 1 incoming 0 answered 0 outgoing 1 answered 1 busy 0
Far-end disconnect at:connect 0 alert 0 hold 0 ring 0
Last 64 far-end disconnect cause codes
0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0

DN 3 chan 1 (95021) voice quality statistics for current call
Call Ref 102 called 94011 calling 95021
Current Tx Pkts 241 bytes 3133 Rx Pkts 3304 bytes 515023 Lost 0
Jitter 30 Latency 0
Worst Jitter 30 Worst Latency 0
Signal Level to phone 201 (-39 dB) peak 5628 (-12 dB)
Packets counted by router 3305
```

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

```
Router# show ephone-dn 2 statistics

DN 2 chan 1 incoming 0 answered 0 outgoing 2 answered 0 busy 0
Far-end disconnect at: connect 0 alert 0 hold 0 ring 0
Last 64 far-end disconnect cause codes
28 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0
local phone on-hook

DN 2 chan 1 (91450) voice quality statistics for last call
Call Ref 2 called  calling
Total Tx Pkts 0 bytes 0 Rx Pkts 0 bytes 0 Lost 0
Final Jitter 0 Latency 0 Lost 0
Signal Level to phone 0 (-78 dB) peak 0 (-78 dB)
Packets counted by router 0
```

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

Related Commands

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

show ephone-dn summary

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

show ephone-dn summary

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

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

Examples

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

```
Router# show ephone-dn summary

PORT     DN STATE    CODEC    VAD VTSP STATE           VPM STATE
======== ==========  ======== === ==================== =========
50/0/1   DOWN        -         -  -                    EFXS_ONHOOK

50/0/2   DOWN        -         -  -                    EFXS_ONHOOK

50/0/3   DOWN        -         -  -                    EFXS_ONHOOK

50/0/4   INVALID     -         -  -                    EFXS_INIT

50/0/5   INVALID     -         -  -                    EFXS_INIT

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

Table 26 describes significant fields in the output from this command.

Table 26 show ephone-dn summary Field Descriptions

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

Related Commands

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

show ephone-dnd

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

show ephone dnd

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

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

Examples

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

```
Router# show ephone dnd

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

The **show capf-server** command describes significant fields in this output.

Related Commands

Command	Description
show ephone	Displays information about ephones.

show ephone-hunt

To display ephone-hunt configuration information and current status and statistics information, use the **show ephone-hunt** command in privileged EXEC mode.

show ephone-hunt [*tag* | **summary**]

Syntax Description

tag	(Optional) Hunt-group number that was used to identify a hunt group in the **ephone-hunt** command. Range is 1 to 100.
summary	(Optional) Displays hunt group configuration information.

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

The **show ephone-hunt** and **show ephone-hunt summary** commands display information for peer, sequential, and last-idle ephone hunt groups. Using the *tag* argument outputs data for a specific ephone hunt group.

The output is dependent on call activity. If there is no activity, no data is displayed.

Examples

The following examples are contained in this section:

- Verbose Output
- Summary Output
- Agent Status Control Conditions
- Automatic Agent Status Not-Ready Parameters

Verbose Output

The following is sample output from the **show ephone-hunt** command when no argument or keyword has been entered. The sample contains information for a peer hunt group, a sequential hunt group, and a longest-idle hunt group. See Table 28 for descriptions of significant fields in the output.

```
Router# show ephone-hunt

Group 1
    type: peer
    pilot number: 450, peer-tag 20123
    list of numbers:
        451, aux-number A450A0900, # peers 5, logout 0, down 1
            peer-tag  dn-tag  rna  login/logout  up/down
             [20122     42      0       login      up  ]
             [20121     41      0       login      up  ]
```

```
                [20120     40     0       login      up  ]
                [20119     30     0       login      up  ]
                [20118     29     0       login      down]
         452, aux-number A450A0901, # peers 4, logout 0, down 0
             peer-tag  dn-tag  rna  login/logout  up/down
                [20127     45     0       login      up  ]
                [20126     44     0       login      up  ]
                [20125     43     0       login      up  ]
                [20124     31     0       login      up  ]
         453, aux-number A450A0902, # peers 4, logout 0, down 0
             peer-tag  dn-tag  rna  login/logout  up/down
                [20131     48     0       login      up  ]
                [20130     47     0       login      up  ]
                [20129     46     0       login      up  ]
                [20128     32     0       login      up  ]
         477, aux-number A450A0903, # peers 1, logout 0, down 0
             peer-tag  dn-tag  rna  login/logout  up/down
                [20132     499    0       login      up  ]
     preference: 0
     preference (sec): 7
     timeout: 3, 3, 3, 3
     max timeout : 10
     hops: 4
     next-to-pick: 1
     E.164 register: yes
     auto logout: no
     stat collect: no
 Group 2
     type: sequential
     pilot number: 601, peer-tag 20098
     list of numbers:
         123, aux-number A601A0200, # peers 1, logout 0, down 0
             peer-tag  dn-tag  rna  login/logout  up/down
                [20097     56     0       login      up  ]
         622, aux-number A601A0201, # peers 3, logout 0, down 0
             peer-tag  dn-tag  rna  login/logout  up/down
                [20101     112    0       login      up  ]
                [20100     111    0       login      up  ]
                [20099     110    0       login      up  ]
         623, aux-number A601A0202, # peers 3, logout 0, down 0
             peer-tag  dn-tag  rna  login/logout  up/down
                [20104     122    0       login      up  ]
                [20103     121    0       login      up  ]
                [20102     120    0       login      up  ]
         *, aux-number A601A0203, # peers 1, logout 0, down 1
             peer-tag  dn-tag  rna  login/logout  up/down
                [20105     0      0       -          down]
         *, aux-number A601A0204, # peers 1, logout 0, down 1
             peer-tag  dn-tag  rna  login/logout  up/down
                [20106     0      0       -          down]
     final number: 5255348
     preference: 0
     preference (sec): 9
     timeout: 5, 5, 5, 5, 5
     max timeout : 40
     fwd-final: orig-phone
     E.164 register: yes
     auto logout: no
     stat collect: no
 Group 3
     type: longest-idle
     pilot number: 100, peer-tag 20142
     list of numbers:
         101, aux-number A100A9700, # peers 3, logout 0, down 3
```

```
            on-hook time stamp 7616, off-hook agents=0
            peer-tag  dn-tag  rna  login/logout  up/down
             [20141     132    0       login      down]
             [20140     131    0       login      down]
             [20139     130    0       login      down]
        *, aux-number A100A9701, # peers 1, logout 0, down 1
            on-hook time stamp 7616, off-hook agents=0
            peer-tag  dn-tag  rna  login/logout  up/down
             [20143     0      0       -          down]
        102, aux-number A100A9702, # peers 2, logout 0, down 2
            on-hook time stamp 7616, off-hook agents=0
            peer-tag  dn-tag  rna  login/logout  up/down
             [20145     142    0       login      down]
             [20144     141    0       login      down]
    all agents down!
    preference: 0
    preference (sec): 7
    timeout: 100, 100, 100
    hops: 0
    E.164 register: yes
    auto logout: no
    stat collect: no
```

Summary Output

The following example shows summary output. See Table 28 for descriptions of significant fields in the output.

```
Router# show ephone-hunt summary

Group 1
    type: peer
    pilot number: 5000
    list of numbers:
       5001
       5002
       5003
       5004
       5005
    final number: 5006
    preference: 0
    timeout: 180
    hops: 2
    E.164 register: yes
Group 2
    type: sequential
    pilot number: 6000
    list of numbers:
       5005
       5004
       5003
       5002
       5001
    final number: 5007
    preference: 5
    timeout: 3
    E.164 register: no
```

Agent Status Control Conditions

A portion of the **show ephone-hunt** command output displays the ready and not-ready agent status of extensions in hunt groups. An extension that is ready is available to receive hunt-group calls. An extension that is in not-ready status blocks hunt-group calls. An agent toggles an extension from ready to not ready and back to ready using the HLog soft key or a FAC.

The following examples display some output that reports different agent status not-ready conditions within a hunt group. In the hunt group used for these examples, there are four users: agent1 and agent4 share extension 8001, agent2 is on extension 8002, and agent3 is on extension 8003.

In the **show ephone-hunt** output, "logout 0" means that all instances of the extension are in ready status. Any number greater than zero next to "logout' indicates that at least one ephone using the extension has activated not-ready status.

If agent1 is in not-ready status, the **show ephone-hunt** command will display the following output. The logout value for extension 8001 is 1 because one phone is in not-ready status.

```
Router# show ephone-hunt
.
.
.
list of numbers:
 8001, aux-number A8000A100, # peers 2, logout 1 ...
 8002, aux-number A8000A101, # peers 1, logout 0...
 8003, aux-number A8000A102, # peers 1, logout 0...
.
```

If agent1 and agent2 place their phones in not-ready status, the **show ephone-hunt** command will display the following output:

```
Router# show ephone-hunt
.
.
.
list of numbers:
    8001, aux-number A8000A100, # peers 2, logout 1...
    8002, aux-number A8000A101, # peers 1, logout 1...
    8003, aux-number A8000A102, # peers 1, logout 0...
```

If all agents place their phones in not-ready status, the **show ephone-hunt** command displays the following output. Note that the logout value of 2 for extension 8001 indicates that both ephone-dns with that extension number (agent1 and agent4) are in not-ready status.

```
Router# show ephone-hunt
.
.
.
list of numbers:
   8001, aux-number A8000A100, # peers 2, logout 2...
   8002, aux-number A8000A101, # peers 1, logout 1...
   8003, aux-number A8000A102, # peers 1, logout 1...
all agents logout!
```

Automatic Agent Status Not-Ready Parameters

The **show ephone-hunt** command displays the parameters that have been set using the **auto logout** command, which is used for the Automatic Agent Status Not-Ready feature. Table 27 shows the possible values of the auto logout field. Table 28 describes other fields in the output.

```
Router# show ephone-hunt 1

Group 1
   type:sequential
   pilot number:8888, peer-tag 20029
```

```
list of numbers:
    8001, aux-number A8888A000, # peers 1, logout 0, down 0
        peer-tag:dn-tag [ 20028:1]
    8003, aux-number A8888A001, # peers 1, logout 0, down 0
        peer-tag:dn-tag [ 20030:3]
preference:0
preference (sec):9
timeout:5
E.164 register:yes
auto logout:no
stat collect:yes
```

Table 27 show ephone-hunt Auto Logout Examples

show ephone-hunt Output	Description	auto logout Command
`auto logout: no`	The Automatic Agent Status Not-Ready feature is disabled. This is also the default if this command is not used.	`no auto logout`
`auto logout: 1 type: both`	The Automatic Agent Status Not-Ready feature is enabled and no options have been used with the **auto logout** command. The number of unanswered calls is 1 and the command applies to both static and dynamic hunt group members by default.	`auto logout`
`auto logout: 2 type: both`	Two unanswered calls will be sent to a hunt group agent before the agent's status is automatically changed to not ready. The command applies to both static and dynamic hunt group members by default.	`auto logout 2`
`auto logout: 3 type: static`	Three unanswered calls will be sent to a hunt group agent before the agent's status is automatically changed to not ready. The command applies to static hunt group members only.	`auto logout 3 static`

Table 28 describes significant fields shown in **show ephone-hunt** command displays.

Table 28 show ephone-hunt Field Descriptions

Field	Description
auto logout	Indicates whether the Automatic Agent Status Not-Ready feature has been enabled. See the "Automatic Agent Status Not-Ready Parameters" section on page 763.
aux-number	Auxiliary number used to generate dial peers for a hunt group. This number is generated by the **list** command.
description	Description string entered for the ephone hunt group. This value is set using the **description (ephone-hunt)** command.
dn-tag	Directory number (DN) sequence number.

Table 28 show ephone-hunt Field Descriptions (continued)

Field	Description
E.164 register	Displays whether a pilot number registers with an H.323 gatekeeper. This value is set by the **no-reg** command.
final number	Last number in the ephone-hunt group, after which a call is no longer redirected. This value is set by the **final** command.
fwd-final	Final destination of an unanswered call that has been transferred into a hunt group: orig-phone means calls are returned to the transferring phone, and final means calls are sent to the final number specified in the configuration. This value is set by the **fwd-final** command.
hops	Number of hops before a call proceeds to the final number. This value is set by the **hops** command.
list of numbers	Extension numbers that are group members of the specified ephone hunt group. This value is set by the **list** command.
login/logout	Ready status of the agent: login means ready and accepting calls, and logout means not-ready and blocking hunt-group calls.
logout	Number of agents in the not-ready state (not accepting hunt-group calls).
max timeout	Maximum combined timeout for the no-answer periods for all ephone-dns in the ephone-hunt list. This value is set by the **max-timeout** command.
next-to-pick	(Peer hunt groups only) List number of the agent whose phone will ring when the next call comes in to the hunt group. (For example, if the order of agents in the **list** command is 451, 452, 453, 454, the list number 2 represents extension 452.)
off-hook agents	Number of agents who are currently off-hook.
on-hook time stamp	(Longest-idle hunt groups only) The last on-hook time of the agent, which is used to determine which agent to ring next time.
peers	Displays the number of ephone-dn dial peers.
peer-tag	Dial-peer sequence number.
pilot number	Number that callers dial to reach the ephone hunt group.
preference	Preference order set by the **preference (ephone-hunt)** command for the primary pilot number.
preference (sec)	Preference order set by the **preference (ephone-hunt)** command for the secondary pilot number.
rna	Number of unanswered hunt group calls (ring-no-answer) by this agent, used for the Automatic Agent Status Not-Ready feature.
stat collect	Indicates whether statistic are being Cisco Unified CME B-ACD data is being collected. See the **statistics collect** command.
timeout	Number of seconds after which a call that is not answered at one number is redirected to the next number in the hunt-group list. Multiple values in this field refer to the timeouts for the hops between ephone-dns in a hunt group as they appear in the **list** command. This value is set by the **timeout** command.
type	Type of ephone hunt group: longest-idle, peer, or sequential.
up/down	Dial peer is up or down.

Related Commands

Command	Description
auto logout	Enables automatic change of agent status to not-ready after a specified number of hunt-group calls are not answered.
ephone-hunt	Enters ephone-hunt configuration mode to create a hunt group for use in a Cisco Unified CME system.
hunt-group logout	Enables separate handling of DND and HLog functionality for hunt-group agents and the display of an HLog soft key on phones.
show ephone-hunt statistics	Displays hunt group call statistics.

show ephone-hunt statistics

To display ephone-hunt statistics information, use the **show ephone-hunt statistics** command in privileged EXEC mode.

show ephone-hunt *tag* **statistics** {**last** *hours* **hours** | **start** *day time* [**to** *day time*]}

Syntax Description

tag	The hunt-tag number that was used to identify a hunt group in an **ephone-hunt** command. Range is 1 to 100.
last	Displays information for the previous number of specified hours, counting backward from the current hour. Range is 1 to 167.
hours **hours**	Number of hours for which to display call statistics.
start	Defines the start of a period for which to display call statistics. Default duration is one hour.
day	Day of week. Use **sun**, **mon**, **tue**, **wed**, **thu**, **fri**, or **sat**.
time	Hour of day. Range is 0 to 23.
to	(Optional) Defines the stop time for display of call statistics.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)T	Cisco CME 3.2	This command was introduced.
12.4(4)XC	Cisco Unified CME 4.0	Call hold statistics were added.
12.4(9)T	Cisco Unified CME 4.0	Call hold statistics were integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

The **show ephone-hunt statistics last** and **show ephone-hunt statistics** commands provide expanded information regarding extension (list of numbers) and pilot numbers.

The output is dependent on call activity. If there is no activity, no data is displayed.

If your Cisco Unified CME system is configured with the basic automatic call distribution (B-ACD) and auto-attendant service, you can enable the collection of call statistics per ephone hunt group with the **statistics collect** command. Additional data is displayed for all agents combined and for individual agents. The additional data includes statistics such as: the number of calls received, the amount of time the calls waited to be answered, and the amount of time the calls spend on hold or in a queue.

The **statistics collect** command can be used to obtain other call statistics, such as direct calls to hunt group pilot numbers. For more information, see the "Cisco Unified CME Basic Automatic Call Distribution and Auto-Attendant Service" chapter in the *Cisco Unified CME B-ACD and TCL Call-Handling Applications* guide.

Once you have enabled statistics collection, you can use the **show ephone-hunt statistics** command to display call statistics, or you can use the **hunt-group report every hours** and **hunt-group report url** commands to transfer the statistics to files using TFTP.

Note Each year on the day that daylight saving time adjusts the time back by one hour at 2 a.m., the original 1 a.m. to 2 a.m. statistics for that day are lost because they are overwritten by the new 1 a.m. to 2 a.m. statistics.

Examples

The following sample output displays call statistics for the past hour for hunt group 2, which is associated with a Cisco Unified CME B-ACD service:

```
Router#show ephone-hunt 2 stat last 1 h
Thu 02:00 - 03:00
    Max Agents: 3
    Min Agents: 3
    Total Calls: 9
    Answered Calls: 7
    Abandoned Calls: 2
    Average Time to Answer (secs): 6
    Longest Time to Answer (secs): 13
    Average Time in Call (secs): 75
    Longest Time in Call (secs): 161
    Average Time before Abandon (secs): 8
    Calls on Hold: 2
    Average Time in Hold (secs): 16
    Longest Time in Hold (secs): 21
    Per agent statistics:
      Agent: 8004
        From Direct Call:
          Total Calls Answered : 3:
          Average Time in Call (secs) : 70
          Longest Time in Call (secs) : 150
          Total Calls on Hold : 1:
          Average Hold Time (secs) : 21
          Longest Hold Time (secs) : 21
        From Queue:
          Total Calls Answered : 3
          Average Time in Call (secs) : 55
          Longest Time in Call (secs) : 78
          Total Calls on Hold : 2:
          Average Hold Time (secs) : 19
          Longest Hold Time (secs) : 26
      Agent: 8006
        From Direct Call:
          Total Calls Answered : 3:
          Average Time in Call (secs) : 51
          Longest Time in Call (secs) : 118
          Total Calls on Hold : 1:
          Average Hold Time (secs) : 11
          Longest Hold Time (secs) : 11
        From Queue:
          Total Calls Answered : 1
          Average Time in Call (secs) : 4
          Longest Time in Call (secs) : 4
      Agent: 8044
        From Direct Call:
          Total Calls Answered : 1:
          Average Time in Call (secs) : 161
          Longest Time in Call (secs) : 161
        From Queue:
          Total Calls Answered : 1
          Average Time in Call (secs) : 658
          Longest Time in Call (secs) : 658
```

```
Queue related statistics:
  Total calls presented to the queue:  5
  Calls answered by agents:  5
  Number of calls in the queue:  0
  Average time to answer (secs):  2
  Longest time to answer (secs):  3
  Number of abandoned calls:  0
  Average time before abandon (secs):  0
  Calls forwarded to voice mail:  0
  Calls answered by voice mail:  0
```

Table 28 describes significant fields shown in **show ephone-hunt** command displays.

Table 29 show ephone-hunt statistics Field Descriptions

Field	Description
Average time before abandon (secs)	Average time that unanswered calls waited before hanging up, in seconds.
Average hold time (secs)	Average time that calls waited on hold for this agent, in seconds.
Average time in call	Average time that unanswered calls waited before going to an agent.
Average time in hold (secs)	Average time that calls were kept on hold for all agents.
Average time to answer (secs)	Average length of time that all calls to Cisco Unified CME B-ACD waited before being answered.
Calls answered by agents	Total number of calls to Cisco Unified CME B-ACD that were answered by ephone-dns or agents.
Calls answered by voice mail	Total number of calls to Cisco Unified CME B-ACD that were answered by voice mail.
Calls exited the queue	Total number of calls to Cisco Unified CME B-ACD that exited queues.
Calls forwarded to voice mail	Total number of calls to Cisco Unified CME B-ACD that were forwarded to voice mail.
Calls on hold	Total number of calls that were placed on hold.
Longest hold time (secs)	Longest time that a call to this agent spent between being placed on hold and being picked up, in seconds.
Longest time in call	Longest time in which calls to Cisco Unified CME B-ACD went to an agent and waited in a call queue.
Longest time in hold (secs)	Longest time that a call spent between being placed on hold and being picked up, in seconds, for all agents.
Longest time to answer (secs)	Longest time in which it took all calls to Cisco Unified CME B-ACD to be answered.
Number of abandoned calls:	Total number of calls to Cisco Unified CME B-ACD that hung up before they could be answered.
Total calls answered	Total number of calls to Cisco Unified CME B-ACD that were answered by an agent.
Total calls on hold	Total number of calls that were on hold for this agent.
Total calls presented to the queue	Total number of calls made to Cisco Unified CME B-ACD.

Related Commands

Command	Description
ephone-hunt	Enters ephone-hunt configuration mode to create a hunt group for use in a Cisco Unified CME system.
hunt-group report every hours	Sets the hourly interval at which Cisco Unified CME B-ACD call statistics are automatically transferred to a file.
hunt-group report url	Sets filename parameters and the URL path where Cisco Unified CME B-ACD call statistics are to be sent using TFTP.
statistics collect	Enables the collection of call statistics for an ephone hunt group.

show fb-its-log

To display information about the Cisco CallManager Express (Cisco CME) eXtensible Markup Language (XML) application program interface (API) configuration, statistics on XML API queries, and the XML API event logs, use the **show fb-its-log** command in privileged EXEC mode.

show fb-its-log [**summary**]

Syntax Description

summary	(Optional) Displays only the XML API configuration and the statistics for queries and logs, and not the logs themselves.

Command Modes

Privileged EXEC

Command History

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

Examples

The following is sample output from the **show fb-its-log summary** command:

```
Router# show fb-its-log summary

IP Keyswitch Logs:(21:11:30 UTC Wed Jul 1 2003)
  ---- Current Period ---
    extension events:4
    device events: 3
    overwrites:0
    missed:0
    deleted:0
 ---- History ------
    overwrites:0
    missed:0
    deleted:8
---- Threads ----
    max xml threads:2
    current thread:0
    read in process:FALSE
```

The following is sample output from the **show fb-its-log** command:

```
Router# show fb-its-log

IP Keyswitch Logs:(21:11:30 UTC Wed Jul 1 2003)
  ---- Current Period ---
    extension events:4
    device events: 3
    overwrites:0
    missed:0
    deleted:0

 ---- History ------
    overwrites:0
```

```
    missed:0
    deleted:8
---- Threads ----
    max xml threads:2
    cuttent thread:0
    read in process:FALSE

1  Time:21:11:06 UTC Wed Jul 1 2003
    Event:DN 1[2001] goes down
2  Time:21:11:06 UTC Wed Jul 1 2003
    Event:DN 2[2003] goes down
3  Time:21:11:06 UTC Wed Jul 1 2003
    Event:IP Phone 1[SEP003094C3F96A] unregistered
4  Time:21:11:06 UTC Wed Jul 1 2003
    Event:IP Phone 1[SEP003094C3F96A] unregistered
5  Time:21:11:54 UTC Wed Jul 2003
    Event:IP Phone 1[SEP003094C3F96A] registered
6  Time:21:11:57 UTC Wed Jul 2003
    Event:DN 1[2001] goes up
7  Time:21:11:57 UTC Wed Jul 2003
    Event:DN 2[2003] goes up
```

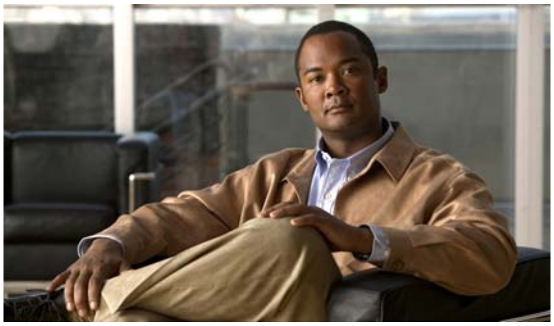
Table 30 describes the significant fields in this output.

Table 30 show fb-its-log Field Descriptions

Field	Description
Current Period	The time between the last retain-timer-triggered cleanup to the next cleanup.
extension events	Events related to extensions that have been captured in the internal buffer.
device events	Events related to devices that have been captured in the internal buffer.
overwrites	Events that are written over previously recorded events in the buffer. Overwrites occur when the internal buffer size is too small; new events overwrite old ones. The internal buffer size is set using the **max-size** keyword in the **log table** command.
missed	Events that happen too quickly for the system to record.
deleted	Events removed from the internal buffer.
History	Information since the last system restart.
Threads	Current number of threads configured in the system.
max xml threads	Maximum number of concurrent XML threads allowed.
current thread	XML API query thread.
read in process	TRUE indicates that the xml-test.html file is being read now. FALSE indicates that the file is not being read.
UTC	Coordinated Universal Time, which is used by the system clock on the Cisco CME router.

Related Commands

Command	Description
log table	Sets the maximum size of the table used to capture phone events used for the Cisco CME XML API.



Cisco Unified CME Commands: show presence global through system message

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This chapter contains commands to configure and maintain Cisco Unified Communications Manager Express (formally known as Cisco Unified CallManager Express). The commands are presented in alphabetical order. Some commands required for configuring Cisco Unified Communications Manager Express (Cisco Unified CME) may be found in other Cisco IOS command references. Use the command reference master index or search online to find these commands.

show presence global

To display configuration information about the presence service, use the **show presence global** command in user EXEC or privileged EXEC mode.

show presence global

Syntax Description

This command has no arguments or keywords.

Command Modes

User EXEC
Privileged EXEC

Command History

Release	Modification
12.4(11)XJ	This command was introduced.
12.4(15)T	This command was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

This command displays the configuration settings for presence.

Examples

The following example displays output from the **show subscription global** command:

```
Router# show subscription global

Presence Global Configuration Information:
=============================================
Presence feature enable            : TRUE
Presence allow external watchers   : FALSE
Presence max subscription allowed  : 100
Presence number of subscriptions   : 0
Presence allow external subscribe  : FALSE
Presence call list enable          : TRUE
Presence server IP address         : 0.0.0.0
Presence sccp blfsd retry interval : 60
Presence sccp blfsd retry limit    : 10
Presence router mode               : CME mode
```

Table 31 describes the significant fields shown in the display.

Table 31 show subscription global Field Descriptions

Field	Description
Presence feature enable	Indicates whether presence is enabled on the router with the **presence** command.
Presence allow external watchers	Indicates whether internal presentities can be watched by external watchers, as set by the **watcher all** command
Presence max subscription allowed	Maximum number of presence subscriptions allowed by the **max-subscription** command.

Table 31 show subscription global Field Descriptions (continued)

Field	Description
Presence number of subscriptions	Current number of active presence subscriptions.
Presence allow external subscribe	Indicates whether internal watchers are allowed to subscribe to status notifications from external presentities, as set by the **allow subscribe** command.
Presence call list enable	Indicates whether the Busy Lamp Field (BLF) call-list feature is enabled with the **presence call-list** command.
Presence server IP address	Displays the IP address of an external presence server defined with the **server** command.
Presence sccp blfsd retry interval	Retry timeout, in seconds, for BLF speed-dial numbers on SCCP phones set by the **sccp blf-speed-dial retry interval** command.
Presence sccp blfsd retry limit	Maximum number of retries allowed for BLF speed-dial numbers on SCCP phones set by the **sccp blf-speed-dial retry interval** command.
Presence router mode	Indicates whether the configuration mode is set to Cisco Unified CME or Cisco Unified SRST by the **mode** command.

Related Commands

Command	Description
allow watch	Allows a directory number on a phone registered to Cisco Unified CME to be watched in a presence service.
allow subscribe	Allows internal watchers to monitor external presence entities (directory numbers).
debug presence	Displays debugging information about the presence service.
presence enable	Allows the router to accept incoming presence requests.
server	Specifies the IP address of a presence server for sending presence requests from internal watchers to external presence entities.
show presence subscription	Displays information about active presence subscriptions.
watcher all	Allows external watchers to monitor internal presence entities (directory numbers).

show presence subscription

To display information about active presence subscriptions, use the **show presence subscription** command in user EXEC or privileged EXEC mode.

show presence subscription [**details** | **presentity** *telephone-number* | **subid** *subscription-id* | **summary**]

Syntax Description

details	(Optional) Displays detailed information about presentities, watchers, and presence subscriptions.
presentity *telephone-number*	(Optional) Displays information on the presentity specified by the destination telephone number.
subid *subscription-id*	(Optional) Displays information for the specific subscription ID.
summary	(Optional) Displays summary information about active subscription requests.

Command Default

Information for all active presence subscriptions is displayed.

Command Modes

User EXEC
Privileged EXEC

Command History

Release	Modification
12.4(11)XJ	This command was introduced.
12.4(15)T	This command was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

This command displays details about the currently active presence subscriptions

Examples

The following is sample output from the **show presence subscription details** command:

```
Presence Active Subscription Records Details:
=============================================

Subscription ID         : 1
  Watcher               : 6002@10.4.171.60
  Presentity            : 6005@10.4.171.34
  Expires               : 3600 seconds
  Subscription Duration : 1751 seconds
  line status           : idle
  watcher type          : local
  presentity type       : local
  Watcher phone type    : SIP Phone
  subscription type     : Incoming Indication
  retry limit           : 0
  sibling subID         : 0
  sdb                   : 0
```

```
  dp                     : 6555346C
  watcher dial peer tag : 40001
  number of presentity  : 1

Subscription ID         : 2
  Watcher               : 6002@10.4.171.60

Presence Active Subscription Records:
=============================================

Subscription ID         : 30
  Watcher               : 4085256003@10.4.171.34
  Presentity            : 5001@10.4.171.20
  Expires               : 3600 seconds
  line status           : idle
  watcher type          : local
  presentity type       : remote
  Watcher phone type    : SCCP [BLF Call List]
  subscription type     : Outgoing Request
  retry limit           : 0
  sibling subID         : 23
  sdb                   : 0
  dp                    : 0
  watcher dial peer tag : 0
```

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

```
Router# show presence subscription summary

Presence Active Subscription Records Summary: 15 subscription
Watcher                  Presentity               SubID  Expires SibID  Status
======================== ======================== ====== ======= ====== ======
6002@10.4.171.60         6005@10.4.171.34          1      3600    0      idle
6005@10.4.171.81         6002@10.4.171.34          6      3600    0      idle
6005@10.4.171.81         6003@10.4.171.34          8      3600    0      idle
6005@10.4.171.81         6002@10.4.171.34          9      3600    0      idle
6005@10.4.171.81         6003@10.4.171.34          10     3600    0      idle
6005@10.4.171.81         6001@10.4.171.34          12     3600    0      idle
6001@10.4.171.61         6003@10.4.171.34          15     3600    0      idle
6001@10.4.171.61         6002@10.4.171.34          17     3600    0      idle
6003@10.4.171.59         6003@10.4.171.34          19     3600    0      idle
6003@10.4.171.59         6002@10.4.171.34          21     3600    0      idle
6003@10.4.171.59         5001@10.4.171.34          23     3600    24     idle
6002@10.4.171.60         6003@10.4.171.34          121    3600    0      idle
6002@10.4.171.60         5002@10.4.171.34          128    3600    129    idle
6005@10.4.171.81         1001@10.4.171.34          130    3600    131    busy
6005@10.4.171.81         7005@10.4.171.34          132    3600    133    idle
```

The following is sample output from the **show presence subscription subid** command:

```
Router# show presence subscription subid 133

Presence Active Subscription Records:
=============================================

Subscription ID         : 133
  Watcher               : 6005@10.4.171.34
  Presentity            : 7005@10.4.171.20
  Expires               : 3600 seconds
  line status           : idle
  watcher type          : local
  presentity type       : remote
  Watcher phone type    : SIP Phone
  subscription type     : Outgoing Request
```

```
retry limit          : 0
sibling subID        : 132
sdb                  : 0
dp                   : 0
watcher dial peer tag : 0
```

Table 31 describes the significant fields shown in the display.

Table 32 show presence subscription Field Descriptions

Field	Description
Watcher	IP address of the watcher.
Presentity	IP address of the presentity.
Expires	Number of seconds until the subscription expires. Default is 3600.
line status	Status of the line: • Idle—Line is not being used. • In-use—User is on the line, whether or not this line can accept a new call. • Unknown—Phone is unregistered or this line is not allowed to be watched.
watcher type	Whether the watcher is local or remote.
presentity type	Whether the presentity is local or remote.
Watcher phone type	Type of phone, either SCCP or SIP.
subscription type	The type of presence subscription, either incoming or outgoing.
retry limit	Maximum number of times the router attempts to subscribe for the line status of an external SCCP phone when either the presentity does not exist or the router receives a terminated NOTIFY from the external presence server. Set with the **sccp blf-speed-dial retry-interval** command.
sibling subID	Sibling subscription ID if presentity is remote. If value is 0, presentity is local.
sdb	Voice port of the presentity.
dp	Dial peer of the presentity.
watcher dial peer tag	Dial peer tag of the watcher device.

Related Commands

Command	Description
allow watch	Allows a directory number on a phone registered to Cisco Unified CME to be watched in a presence service.
debug ephone blf	Displays debugging information for Busy Lamp Field (BLF) presence features.
debug presence	Displays debugging information about the presence service.
presence	Enables presence service and enters presence configuration mode.

Command	Description
presence enable	Allows the router to accept incoming presence requests.
show presence global	Displays configuration information about the presence service.

show sdspfarm

To display the status of the configured digital signal processor (DSP) farms and transcoding streams, use the **show sdspfarm** command in privileged EXEC mode.

show sdspfarm {**units** | **sessions** {**active** | **callID** *number* | **statistics** | **summary**}}

Syntax Description

units	Displays the configured and registered DSP farms.
sessions	Displays the transcoding streams.
active	Displays all active sessions.
callID	Displays activities for a specific caller ID.
number	Displays caller ID number displayed by the **show voip rtp connection** command.
statistics	Displays session statistics.
summary	Displays summary information.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.3(11)T	3.2	This command was introduced.

Examples

The following is sample output from the **show sdspfarm units** command:

```
Router# show sdspfarm units

mtp-1 Device:MTP123456782012 TCP socket:[-1]  UNREGISTERED
actual_stream:0 max_stream 0 IP:0.0.0.0  0  Unknown 0 keepalive 0

mtp-2 Device:MTP000a8aeaca80 TCP socket:[5]  REGISTERED
actual_stream:40 max_stream 40 IP:10.5.49.160  11001  MTP YOKO keepalive 12074
Supported codec:G711Ulaw
                 G711Alaw
                 G729
                 G729a
                 G729b
                 G729ab

 max-mtps:2, max-streams:240, alloc-streams:40, act-streams:0

The following is sample output from the show sdspfarm sessions active command:
Router# show sdspfarm sessions active

Stream-ID:3 mtp:2 1.5.49.160  20174  Local:2000 START
 usage:MoH  (DN=3  , CH=1) FE=TRUE
 codec:G729  duration:20 vad:0 peer Stream-ID:4

Stream-ID:4 mtp:2 1.5.49.160  17072  Local:2000 START
 usage:MoH  (DN=3  , CH=1) FE=FALSE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:3
```

The following is sample output from the **show sdspfarm sessions callID** command:

```
Router# show sdspfarm sessions callid 51M

Stream-ID:6, srcCall-ID:51, codec:G729AnnexA , dur:20ms, vad:0, dstCall-ID:52, confID:5,
mtp:2^
Peer Stream-ID:5, srcCall-ID:52, codec:G711Ulaw64k , dur:20ms, vad:0, dstCall-ID:51,
confID:5, mtp:2^
Router-2015# show sdspfarm sessions callid 52
Stream-ID:5, srcCall-ID:52, codec:G711Ulaw64k , dur:20ms, vad:0, dstCall-ID:51, confID:5,
mtp:2
Peer Stream-ID:6, srcCall-ID:51, codec:G729AnnexA , dur:20ms, vad:0, dstCall-ID:52,
confID:5, mtp:2
```

The following is sample output from the **show sdspfarm sessions statistics** command:

```
Router# show sdspfarm sessions statistics

Stream-ID:1 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:1014 in-pak:0 discard:0
Stream-ID:2 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:3 mtp:2 10.5.49.160  20174  Local:2000START  MoH  (DN=3  , CH=1) FE=TRUE
 codec:G729  duration:20 vad:0 peer Stream-ID:4
 recv-pak:0 xmit-pak:0 out-pak:4780 in-pak:0 discard:0
Stream-ID:4 mtp:2 10.5.49.160  17072  Local:2000START  MoH  (DN=3  , CH=1) FE=FALSE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:3
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:5 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:6 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:7 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:8 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:9 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:10 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:11 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:12 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:13 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:14 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:15 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
```

```
Stream-ID:16 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:17 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:18 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:19 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:20 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:21 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:22 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:23 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:24 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:25 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:26 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:27 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:28 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:29 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:30 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:31 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:32 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:33 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:34 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:35 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:36 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:37 mtp:2 0.0.0.0  0  Local:0IDLE
```

```
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:38 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:39 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
Stream-ID:40 mtp:2 0.0.0.0  0  Local:0IDLE
 codec:G711Ulaw64k  duration:20 vad:0 peer Stream-ID:0
 recv-pak:0 xmit-pak:0 out-pak:0 in-pak:0 discard:0
```

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

```
Router# show sdspfarm sessions summary

max-mtps:2, max-streams:240, alloc-streams:40, act-streams:2
  ID   MTP  State      CallID confID Usage                         Codec/Duration
==== ===== ====== =========== ====== ============================= ==============
1    2     IDLE   -1          0                                   G711Ulaw64k /20ms
2    2     IDLE   -1          0                                   G711Ulaw64k /20ms
3    2     START  -1          3      MoH  (DN=3  , CH=1) FE=TRUE  G729 /20ms
4    2     START  -1          3      MoH  (DN=3  , CH=1) FE=FALSE G711Ulaw64k /20ms
5    2     IDLE   -1          0                                   G711Ulaw64k /20ms
6    2     IDLE   -1          0                                   G711Ulaw64k /20ms
7    2     IDLE   -1          0                                   G711Ulaw64k /20ms
8    2     IDLE   -1          0                                   G711Ulaw64k /20ms
9    2     IDLE   -1          0                                   G711Ulaw64k /20ms
10   2     IDLE   -1          0                                   G711Ulaw64k /20ms
11   2     IDLE   -1          0                                   G711Ulaw64k /20ms
12   2     IDLE   -1          0                                   G711Ulaw64k /20ms
13   2     IDLE   -1          0                                   G711Ulaw64k /20ms
14   2     IDLE   -1          0                                   G711Ulaw64k /20ms
15   2     IDLE   -1          0                                   G711Ulaw64k /20ms
16   2     IDLE   -1          0                                   G711Ulaw64k /20ms
17   2     IDLE   -1          0                                   G711Ulaw64k /20ms
18   2     IDLE   -1          0                                   G711Ulaw64k /20ms
19   2     IDLE   -1          0                                   G711Ulaw64k /20ms
20   2     IDLE   -1          0                                   G711Ulaw64k /20ms
21   2     IDLE   -1          0                                   G711Ulaw64k /20ms
22   2     IDLE   -1          0                                   G711Ulaw64k /20ms
23   2     IDLE   -1          0                                   G711Ulaw64k /20ms
24   2     IDLE   -1          0                                   G711Ulaw64k /20ms
25   2     IDLE   -1          0                                   G711Ulaw64k /20ms
26   2     IDLE   -1          0                                   G711Ulaw64k /20ms
27   2     IDLE   -1          0                                   G711Ulaw64k /20ms
28   2     IDLE   -1          0                                   G711Ulaw64k /20ms
29   2     IDLE   -1          0                                   G711Ulaw64k /20ms
30   2     IDLE   -1          0                                   G711Ulaw64k /20ms
31   2     IDLE   -1          0                                   G711Ulaw64k /20ms
32   2     IDLE   -1          0                                   G711Ulaw64k /20ms
33   2     IDLE   -1          0                                   G711Ulaw64k /20ms
34   2     IDLE   -1          0                                   G711Ulaw64k /20ms
35   2     IDLE   -1          0                                   G711Ulaw64k /20ms
36   2     IDLE   -1          0                                   G711Ulaw64k /20ms
37   2     IDLE   -1          0                                   G711Ulaw64k /20ms
38   2     IDLE   -1          0                                   G711Ulaw64k /20ms
39   2     IDLE   -1          0                                   G711Ulaw64k /20ms
40   2     IDLE   -1          0                                   G711Ulaw64k /20ms
```

Table 33 describes the fields shown in the show sdspfarm command display.

Table 33 show sdspfarm Field Descriptions

Field	Description
act-streams	Active streams that are currently involved in calls.
alloc-streams	Number of transcoding streams that are actually allocated to all DSP farms that are registered to Cisco CME.
callID	Caller ID that the active stream is in.
Codec	Codec in use.
confID	ConfID that is used to communicate with DSP farms.
discard	Number of packets that are discarded.
dstCall-ID	Caller ID of the destination IP call leg.
Duration or dur	Packet rates, in milliseconds.
ID	Transcoding stream sequence number in Cisco CME.
in-pak	Number of incoming packets from the source call leg.
Local	Local port for voice packets.
max-mtps	Maximum number of Message Transfer Parts (MTPs) that are currently allowed to register in Cisco CME.
max-streams	Maximum number of transcoding streams that are currently allowed in Cisco CME.
mtp or MTP	MTP sequence number where the transcoding stream is located.
out-pak	Number of outgoing packets sending to source call leg.
peer Stream-ID	Stream sequence number of the other stream paired in the same transcoding session. (Two transcoding streams make up a transcoding session).
recv-pak	Number of voice packets received from DSP farm.
srcCall-ID	Source caller ID of the source IP call leg.
State	Current state of the transcoding stream, could be IDLE, SEIZE, START, STOP, or END.
Stream-ID	Transcoding stream sequence number in Cisco CME.
TCP-socket	Socket number for DSP farm (similar to TCP socket for **show ephone** output).
usage	Current usage of the stream; for example, Ip-Ip (IP to IP transcoding), MOH (for MOH transcoding) and Conf (conference).
vad	Voice-activity detection (VAD) flag for the transcoding stream. It should always be 0 (False).
xmit-pak	Number of packets that are sent to DSP farm.

Related Commands

Command	Description
sdspfarm tag	Permits a DSP farm to be to registered to Cisco CME and be associated with an SCCP client interface's MAC address.

Command	Description
sdspfarm transcode sessions	Specifies the maximum number of transcoding sessions allowed per Cisco CME router.
sdspfarm units	Specifies the maximum number of DSP farms that are allowed to be registered to Cisco CME.

show telephony-service admin

To display information about the Cisco CallManager Express (Cisco CME) system administrator, use the **show telephony-service admin** command in user EXEC or privileged EXEC mode.

show telephony-service admin

Syntax Description

This command has no arguments or keywords.

Command Modes

User EXEC
Privileged EXEC

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.2(2)XT	2.0	This command was introduced on the following platforms: Cisco 1750, Cisco 1751, Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(8)T	2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	2.0	This command was implemented on the Cisco 2600XM and Cisco 2691.
12.2(11)T	2.0.1	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760.

Examples

The following is sample output from this command:

```
Router# show telephony-service admin

admin_username Admin
admin_password word
edit DN through Web: enabled.
edit TIME through Web: enabled.
```

Table 34 describes the significant fields in this output.

Table 34 show telephony-service admin Field Descriptions

Field	Description
admin_username	Username of system administrator.
admin_password	Password of system administrator.
edit DN through Web	Whether editing of extensions through the GUI has been enabled using the **dn-webedit** command.
edit TIME through Web	Whether changing the router time through the GUI has been enabled using the **time-webedit** command.

Related Commands

Command	Description
dn-webedit	Enables adding of extensions (ephone-dns) through the web interface.
time-webedit	Enables setting of time through the web interface.

show telephony-service all

To display detailed configuration for phones, voice ports, and dial peers in a Cisco CallManager Express (Cisco CME) system, use the **show telephony-service all** command in user EXEC or privileged EXEC mode.

show telephony-service all

Syntax Description

This command has no arguments or keywords.

Command Modes

User EXEC
Privileged EXEC

Command History

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

Examples

The following is sample output from this command:

```
Router# show telephony-service all

CONFIG
======
ip source-address 10.0.0.1 port 2000
max-ephones 24
max-dn 24
dialplan-pattern 1 408734....
voicemail 11111
transfer-pattern 510734....
keepalive 30


ephone-dn 1
number 5001
huntstop

ephone-dn 2
number 5002
huntstop
call-forward noan 5001 timeout 8
```

```
ephone-dn 3
number 5003
huntstop


ephone 1
mac-address 0030.94C3.37CB
type 0
button  1:1
speed-dial 1 5002
speed-dial 2 5003
cos 0
!
ephone 2
mac-address 0030.94C3.F96A
type 0
button  1:2 2:3 3:4
speed-dial 1 5004
speed-dial 2 5001
cos 0
!


voice-port 50/0/1
 station-id number 5001
!
voice-port 50/0/2
 station-id number 5002
 timeout ringing 8
!

dial-peer voice 20025 pots
 destination-pattern 5001
 huntstop
 port 50/0/1

dial-peer voice 20026 pots
 destination-pattern 5002
 huntstop
 call-forward noan 5001
 port 50/0/2

dial-peer voice 20027 pots
 destination-pattern 5003
 huntstop
 port 50/0/3
```

Table 35 describes significant fields in this output, in alphabetical order.

Table 35 show telephony-service all Field Descriptions

Field	Description
button	Button on the Cisco IP phone.
call-forward noan	Call forward no answer is set.
cos	Not applicable; unused.
destination-pattern	Destination pattern (telephone number) configured for this dial peer.
dial-peer voice	Voice dial peer.
dialplan-pattern	Dial-plan pattern is set to expand the abbreviated extension numbers to fully qualified E.164 numbers.

Table 35 show telephony-service all Field Descriptions (continued)

Field	Description
ephone	Cisco IP phone.
ephone-dn	Cisco IP phone directory number.
huntstop	Huntstop is set.
ip source-address	IP address used by Cisco IP phones to register with the router for service.
keepalive	IP phone keepalive period, in seconds.
mac-address	MAC address.
max-dn	Maximum directory numbers.
max-ephones	Maximum numbers of Cisco IP phones.
number	Cisco IP phone number.
port	TCP port number used by Cisco IP phones to communicate with the router.
pots	POTS dial peer set.
speed-dial	Speed-dial is set.
station-id number	Number used for caller ID purposes when calls are made using the line.
timeout	Timeout is set.
timeout ringing	Maximum amount of time that the phone is allowed to ring before the call is disconnected.
transfer-pattern	Transfer pattern is set to allow transfer of calls to a specified number.
type	Not applicable; unused.
voicemail	A voice-mail (speed-dial) number is set.
voice-port	(Virtual) voice port designator.

Related Commands

Command	Description
show telephony dial-peer	Displays dial peers for extensions in a Cisco CME system.
show telephony voice-port	Displays virtual voice-port configuration for extensions in a Cisco CME system.

show telephony-service bulk-speed-dial

To display information about bulk speed-dial lists, use the **show telephony-service bulk-speed-dial** command in privileged EXEC mode.

show telephony-service bulk-speed-dial {**global** *list-id index-id* [**all**] | **local** *phone-tag list-id index-id* [**all**] | **summary**}

Syntax Description

global	Global lists that can be accessed by all users.
local	Personal lists that can be accessed by users configured to use the lists.
list-id	Digit that identifies the list. Range is from 0 to 9.
index-id	Identification number for an entry.
phone-tag	Ephone identifier (phone-tag).
summary	List of registered bulk speed-dial text files.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Examples

The following example displays the list of bulk speed-dial text files that have been configured in the system:

```
Router# show telephony-service bulk-speed-dial summary

List-id    Entries      Size    Reference  url
      0         40      3840    Global     tftp://192.168.254.254/phonedirs/uut.csv
      1         20      1920    Global     phoneBook.csv
      8         15      1440    Global     tftp://192.168.254.254/phonedirs/big.txt
      9         20      1920    Global     tftp://192.168.254.254/phonedirs/phoneBook.csv
      6      24879   2388384    ephone-2   tftp://192.168.254.254/phonedirs/big.txt1
      7         20      1920    ephone-2   phoneBook.csv
      6      24879   2388384    ephone-3   big.txt1
      7         20      1920    ephone-3   phoneBook.csv
4 Global List(s) 4 Local List(s)
```

The following example displays the single entry 1234 from list 9:

```
Router# show telephony-service bulk-speed-dial global 9 1234

Number: 1800 200 1345 name: Jay Smith Private: yes Extension: No
```

The following example displays all index entries starting with 1 for personal list number 7 for ephone 2:

```
Router# show telephony-service bulk-speed-dial local 2 7 1 all
    Index  Number                  Name                 Hide    Append
    1000   918005550164            ABC Co Front Desk      no      no
    1003   919005550167            ABC Co File room       no      no
    1100   918005550118                                   no      no
    1200   918005550184            ABC Co President       no      no
    1301   918005550152                                   no      no
    1342   91800,5550185           ABC Co Sales           no      no
    1682   91800555,,0115          ABC Co Service         no      no
```

Table 36 describes the significant fields shown in the display.

Table 36 show telephony-service bulk-speed-dial Field Descriptions

Field	Description
List-id	Digit that identifies the list. Range is from 0 to 9.
Entries	Number of entries in the speed-dial file.
Size	Size of the file, in KB.
Reference	Assignment of the list: global if assigned to all ephones, or a specific ephone number.
url	Location of the text file, in URL format.
Index	Identification number for an entry.
Number	Number to be dialed and displayed on the phone.
Name	Name to be displayed on the phone.
Hide	Yes indicates that this number should not be displayed when it is dialed.
Append	Yes indicates that additional digits can be dialed by the user after this number has been speed-dialed before the call is completed.

Related Commands

Command	Description
bulk-speed-dial list (ephone)	Enables a personal bulk speed-dial list for an ephone.
bulk-speed-dial list (telephony-service)	Enables a global bulk speed-dial list for all users of a Cisco Unified CME system.
bulk-speed-dial prefix	Sets the prefix code that phone users dial to access speed-dial numbers from a bulk speed-dial list.

show telephony-service conference hardware

To display information about hardware conferences in a Cisco CallManager Express (Cisco CME) system, use the **show telephony-service conference hardware** command in privileged EXEC mode.

show telephony-service conference hardware [**ad-hoc** [**detail**] | **detail** | **meetme** [**detail**] | **number** *telephone-number*]

Syntax Description

ad-hoc	(Optional) Ad hoc conferences.
detail	(Optional) Detailed information for all conferences.
meetme	(Optional) Meet-me conferences.
number	(Optional) Conference number.
telephone-number	(Optional) Telephone or extension number.

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

Use this command to display information about ad hoc and meet-me conferences, such as verifying which parties are still in the conference.

Examples

The following sample output displays information for a four-party ad hoc conference. Extension 8044 created the conference by calling extension 8012, then adding extension 8004 to the conference. The conference administrator, extension 8006, called into the conference after it was established.

```
Router# show telephony-service conference hardware detail

Conference   Type  Active  Max  Peak    Master         MasterPhone   Last
                                                       cur(initial)
=================================================================================
8893         Ad-hoc    4      8     4    8044           29    ( 29)   8006
Conference parties:
    8006  (admin)
    8004
    8012
    8044
```

Table 37 describes the significant fields shown in the display.

Table 37 show telephony-service conference hardware Field Descriptions

Field	Description
Active	Number of active parties in the conference.
admin	Ad hoc and meet-me hardware conference administrator. The administrator can: • Dial in to any conference directly through the conference number • Use the ConfList soft key to list conference parties • Remove any party from any conference
Conference	Conference directory number (DN).
Conference parties	DNs in the conference.
Last	Last party to join the conference.
Master	Conference creator.
MasterPhone cur(initial)	cur—Current master phone. The phone that hosts the conference creator now. (initial)—Initial master phone. The phone that hosted the conference creator when the conference was created. Because you can transfer the conference creator, the current master phone may be different from the initial master phone.
Max	Maximum parties allowed in the conference.
Peak	Maximum parties in the conference at any time.
Type	Type of conference: meet-me or ad hoc.

show telephony-service dial-peer

To display dial peer information for extensions in a Cisco CallManager Express (Cisco CME) system, use the **show telephony-service dial-peer** command in user EXEC or privileged EXEC mode.

show telephony-service dial-peer

Syntax Description

This command has no arguments or keywords.

Command Modes

User EXEC
Privileged EXEC

Command History

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

Usage Guidelines

The dial peers cannot be edited manually. To change values associated with dial peers, use the **ephone-dn** command.

Examples

The following is sample output from this command:

```
Router# show telephony-service dial-peer

dial-peer voice 20025 pots
 destination-pattern 5001
 huntstop
 port 50/0/1

dial-peer voice 20026 pots
 destination-pattern 5002
 huntstop
 call-forward noan 5001
 port 50/0/2

dial-peer voice 20027 pots
 destination-pattern 5003
 huntstop
 port 50/0/3
```

```
dial-peer voice 20028 pots
 destination-pattern 5004
 huntstop
 port 50/0/4
```

Table 38 describes significant fields in this output, in alphabetical order.

Table 38 show telephony-service dial-peer Field Descriptions

Field	Description
call-forward noan	Call forward no answer is set.
destination-pattern	Destination pattern (telephone number) configured for this dial peer.
dial-peer voice	Voice dial peer.
huntstop	Huntstop is set.
port	(Virtual) voice port designator.
pots	Plain old telephone service (POTS) dial peer set.

Related Commands

Command	Description
ephone	Enters ephone configuration mode.
ephone-dn	Enters ephone-dn configuration mode.
show telephony-service all	Displays detailed configuration for a Cisco CME system.
show telephony-service ephone-dn	Displays information for extensions (ephone-dns) in a Cisco CME system.
show telephony-service voice-port	Displays virtual voice-port configuration of extensions in a Cisco CME system.

show telephony-service directory-entry

To display the entries made using the **directory entry** command, use the **show telephony-service directory-entry** command in user EXEC or privileged EXEC mode.

show telephony-service directory-entry

Syntax Description

This command has no arguments or keywords.

Command Modes

User EXEC
Privileged EXEC

Command History

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

Usage Guidelines

This command lists directory entries that are made using the **directory entry** command but does not list entries that are made using the **name** and **number** commands in ephone-dn configuration mode.

Examples

The following is sample output from this command:

```
Router# show telephony-service directory-entry

directory entry 1 4085550123 name Smith, John
```

Table 39 describes significant fields in this output, in alphabetical order.

Table 39 show telephony-service directory-entry Field Descriptions

Field	Description
directory *directory-tag* (shown as 1 in the example)	Sequence number or unique identifier for a directory entry.
name (shown as Smith, John)	Name that appears in the directory associated with the number.
number (shown as 4085550123 in the example)	Telephone number or extension for the directory entry.

Related Commands

Command	Description
directory entry	Adds an entry to a local phone directory that can be displayed on IP phones.
show telephony-service all	Displays detailed configuration of a Cisco CME system.
show telephony-service ephone-dn	Displays information for extensions (ephone-dns) in a Cisco CME system.

show telephony-service ephone

To display configuration for the Cisco IP phones, use the **show telephony-service ephone** command in user EXEC or privileged EXEC mode.

show telephony-service ephone

Syntax Description

This command has no arguments or keywords.

Command Modes

User EXEC
Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco Unified CME 1.0	This command was introduced on the Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(2)XT	Cisco Unified CME 2.0	This command was implemented on the Cisco 1750 and Cisco 1751.
12.2(8)T	Cisco Unified CME 2.0	This command was implemented on the Cisco 1750 and Cisco 1751.
12.2(8)T1	Cisco Unified CME 2.0	This command was implemented on the Cisco 2600XM and Cisco 2691.
12.2(11)T	Cisco Unified CME 2.01	This command was implemented on the Cisco 2600XM and Cisco 2691.
12.4(11)XJ	Cisco Unified CME 4.1	The **conference add-mode**, **conference drop-mode**, and **conference admin fields** were added.
12.4(15)T	Cisco Unified CME 4.1	This command with modifications was integrated into Cisco IOS Release 12.4(15)T.

Examples

The following is sample output from this command:

```
Router# show telephony-service ephone

ephone 1
mac-address 0030.94C3.37CB
type 0
button  1:1
speed-dial 1 5002
speed-dial 2 5003
cos 0
conference drop-mode never
conference add-mode all
conference admin: Yes
!
ephone 2
mac-address 0030.94C3.F96A
type 0
```

```
button  1:2 2:3 3:4
speed-dial 1 5004
speed-dial 2 5001
cos 0
!
```

Table 40 describes significant fields in this output.

Table 40 show telephony-service ephone Field Descriptions

Field	Description
button	Button number on IP phone, separator to denote ring characteristics and ephone-dn tag. A colon (:) separator denotes a normal ring.
conference add-mode	Who can add parties to a conference: • creator—Only the creator can add parties. • all—Any party can add other parties if the creator remains in the conference.
conference drop-mode	When conferences are dropped: • creator—Conference terminates when the creator hangs up. • local—Conference terminates when the last local party in the conference hangs up or drops out of the conference. • never—Conference is not dropped, even if the creator hangs up, as long as three parties remain in the conference.
conference admin	Ad hoc and meet-me hardware conference administrator. The administrator can: • Dial in to any conference directly through the conference number • Use the ConfList soft key to list conference parties • Remove any party from any conference
cos	Not used.
ephone	Cisco IP phone.
mac-address	MAC address of the Cisco IP phone.
type	Not used.
speed-dial	Speed-tag (unique identifier) and the number that is programmed for that speed-tag.

Related Commands

Command	Description
show telephony-service all	Displays detailed configuration for a Cisco Unified CME system.
show telephony-service dial-peer	Displays dial-peer information for extensions in a Cisco Unified CME system.
show telephony-service ephone-dn	Displays information for extensions (ephone-dns) in a Cisco Unified CME system.
show telephony-service voice-port	Displays configurations for virtual voice ports in a Cisco Unified CME system.

show telephony-service ephone-dn

To display information about extensions (ephone-dns) in a Cisco CallManager Express (Cisco CME) system, use the **show telephony-service ephone-dn** command in user EXEC or privileged EXEC mode.

show telephony-service ephone-dn

Syntax Description

This command has no arguments or keywords.

Command Modes

User EXEC
Privileged EXEC

Command History

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

Examples

The following is sample output from this command:

```
Router# show telephony-service ephone-dn

ephone-dn 1
 number 5001
 huntstop

ephone-dn 2
 number 5002
 huntstop
 call-forward noan 5001 timeout 8

ephone-dn 3
 number 5003
 huntstop

ephone-dn 4
 number 5004
 huntstop
```

Table 41 describes significant fields in this output, in alphabetical order.

Table 41 show telephony-service ephone-dn Field Descriptions

Field	Description
call-forward noan	Call forwarding is set to no answer. Other available options are call-forward busy and call-forward all.
ephone-dn	Cisco IP phone directory number.
huntstop	Huntstop is set.
number	Cisco IP phone number.
timeout	Timeout setting for call forwarding when an extension does not answer.

Related Commands

Command	Description
show telephony-service all	Displays the detailed configuration of all the Cisco IP phones.
show telephony-service dial-peer	Displays dial peer information for extensions (ephone-dns) in a Cisco CME system.
show telephony-service voice-port	Displays configurations for virtual voice ports in a Cisco CME system.

show telephony-service ephone-dn-template

To display information about ephone-dn-template configurations, use the **show telephony-service ephone-dn-template** command in user EXEC or privileged EXEC mode.

show telephony-service ephone-dn-template

Syntax Description

This command has no arguments or keywords.

Command Modes

User EXEC
Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command displays contents of ephone-dn templates. Use the **show running-config** command to display the association of templates to particular ephone-dns.

Examples

The following is sample output from this command:

```
Router# show telephony-service ephone-dn-template

ephone-template 1
 softkeys idle  Newcall Redial Cfwdall Dnd Pickup Gpickup Login
 codec g711ulaw
 User Locale: US
 Network Locale: US

ephone-template 2
 softkeys idle  Redial Newcall Dnd Cfwdall Pickup Gpickup Login
 codec g711ulaw
 User Locale: US
 Network Locale: US
```

Related Commands

Command	Description
ephone-dn-template	Creates an ephone-dn template and enters ephone-dn-template configuration mode.

show telephony-service ephone-template

To display the contents of ephone-templates, use the **show telephony-service ephone-template** command in user EXEC or privileged EXEC mode.

show telephony-service ephone-template

Syntax Description

This command has no arguments or keywords.

Command Modes

User EXEC
Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)T	Cisco CME 3.2	This command was introduced.
12.4(11)XJ	Cisco Unified CME 4.1	The **conference add-mode**, **conference drop-mode**, and **conference admin fields** were added.
12.4(15)T	Cisco Unified CME 4.1	This command with the **conference add-mode**, **conference drop-mode**, and **conference admin fields** was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

This command displays the contents of each ephone template that has been defined. Use the **show running-config** command to display the association of templates to particular ephones.

Examples

The following is sample output from this command:

```
Router# show telephony-service ephone-template

ephone-template 1
softkey hold Join Newcall Resume Select
softkey idle Cfwdall ConfList Dnd Gpickup HLog Join Login Newcall Pickup Redial RmLstC
softkey seized Callback Cfwdall Endcall Gpickup HLog Meetme Pickup Redial
softkey alerting Acct Callback Endcall
softkey connected Acct ConfList Confrn Endcall Flash HLog Hold Join Park RmLstC Select
Trnsfer
conference drop-mode local
conference add-mode creator
conference admin: Yes
Always send media packets to this router: No
Preferred codec: g711ulaw
button-layout 7931 1
User Locale: US
Network Locale: US
```

Table 40 describes significant fields in this output.

Table 42 show telephony-service ephone Field Descriptions

Field	Description
ephone-template	Identifier for the ephone template.
softkey hold	Soft keys displayed during the hold call stage.
softkey idle	Soft keys displayed during the call-idle call stage.
softkey seized	Soft keys displayed during the call-seized call stage.
softkey alerting	Soft keys displayed during the call-alerting call stage.
softkey connected	Soft keys displayed during the call-connected call stage.
conference drop-mode	When conferences are dropped: • creator—Conference terminates when the creator hangs up. • local—Conference terminates when the last local party in the conference hangs up or drops out of the conference. • never—Conference is not dropped, even if the creator hangs up, as long as three parties remain in the conference.
conference add-mode	Who can add parties to a conference: • creator—Only the creator can add parties. • all—Any party can add other parties if the creator remains in the conference.
conference admin	Ad hoc and meet-me hardware conference administrator. The administrator can: • Dial in to any conference directly through the conference number • Use the ConfList soft key to list conference parties • Remove any party from any conference
Always send media packets to this router	Always send media packets to this Cisco Unified CME router, which acts as a proxy and forwards the packets to the destination, instead of sending them directly to the destination IP phone.
Preferred codec	Codec to use when initiating a call.
button-layout	Type of IP phone and number of fixed line or feature set. • 1—Button 24=Menu. Button 23=Headset. • 2—Button 24=Menu. Button 23=Headset. Button 22=Directories. Button 21=Messages.
User Locale	Locale that is associated with the phone user interface. The user locale identifies a set of detailed information, including language and font, to support users.
Network Locale	Locale that is associated with the phone. The network locale contains a definition of the tones and cadences that are used by the phones and gateways in the device pool in a specific geographic area.

Related Commands

Command	Description
ephone-template	Creates an ephone template and enters ephone-template configuration mode.

show telephony-service fac

To display current feature access codes (FACs), use the **show telephony-service fac** command in privileged EXEC mode.

show telephony-service fac

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Phone users dial FACs to access phone features. The set of standard FACs must be enabled using the **fac standard** command before phone users can use them. Individual FACs can be changed using the **fac custom** command.

Examples

The following example displays the set of standard FACs:

```
Router# show telephony-service fac

telephony-service fac standard
 callfwd all **1
 callfwd cancel **2
 pickup local **3
 pickup group **4
 pickup direct **5
 park **6
 dnd **7
 redial **8
 voicemail **9
 ephone-hunt join *3
 ephone-hunt cancel #3
```

Related Commands

Command	Description
fac	Enables standard FACs or creates a custom FAC.

show telephony-service security-info

To display the security-related information that is configured under telephony-service, use the **show telephony-service security-info** command in privileged EXEC configuration mode.

show telephony-service security-info

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used with Cisco Unified CME phone authentication.

Examples

The following example displays security information that was configured under telephony-service.

```
Router# show telephony-service security-info

Skinny Server Trustpoint for TLS: cisco1
TFTP Credentials Trustpoint: cisco1
Server Security Mode: Secure
Global Device Security Mode: Authenticated
```

show telephony-service tftp-bindings

To display the current configuration files accessible to IP phones, use the **show telephony-service tftp-bindings** command in user EXEC or privileged EXEC mode.

show telephony-service tftp-bindings

Syntax Description

This command has no arguments or keywords.

Command Modes

User EXEC
Privileged EXEC

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.2(11)YT	2.1	This command was introduced.
12.2(15)T	2.1	This command was integrated into Cisco IOS Release 12.2(15)T.

Usage Guidelines

Use this command with Cisco IOS Telephony Services V2.1, Cisco CallManager Express 3.0, or a later version.

This command provides a list of configuration files that are accessible to IP phones using TFTP, including the dictionary, language, and tone configuration files that are associated with the ISO-3166 codes that have been selected using the **user-locale** and **network-locale** commands.

Examples

The following is sample output from the **show telephony-service tftp-bindings** command when the ISO-3166 code for Germany has been selected for both language and tones:

```
Router(config)# show telephony-service tftp-bindings

tftp-server system:/its/SEPDEFAULT.cnf
tftp-server system:/its/SEPDEFAULT.cnf alias SEPDefault.cnf
tftp-server system:/its/XMLDefault.cnf.xml alias XMLDefault.cnf.xml
tftp-server system:/its/ATADefault.cnf.xml
tftp-server system:/its/XMLDefault7960.cnf.xml alias SEP00036B54BB15.cnf.xml
tftp-server system:/its/germany/7960-font.xml alias German_Germany/7960-font.xml
tftp-server system:/its/germany/7960-dictionary.xml alias
German_Germany/7960-dictionary.xml
tftp-server system:/its/germany/7960-kate.xml alias German_Germany/7960-kate.xml
tftp-server system:/its/germany/SCCP-dictionary.xml alias
German_Germany/SCCP-dictionary.xml
tftp-server system:/its/germany/7960-tones.xml alias Germany/7960-tones.xml
```

Related Commands

Command	Description
network-locale	Sets the definition of the tones and cadences on the Cisco IP Phones 7940 and 7940G and the Cisco IP Phones 7960 and 7960G for a specific geographic area.
user-locale	Sets language for displays on the Cisco IP Phones 7940 and 7940G and the Cisco IP Phones 7960 and 7960G.

show telephony-service voice-port

To display configurations of virtual voice ports in a Cisco CallManager Express (Cisco CME) system, use the **show telephony-service voice-port** command in user EXEC or privileged EXEC mode.

show telephony-service voice-port

Syntax Description

This command has no arguments or keywords.

Command Modes

User EXEC
Privileged EXEC

Command History

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

Usage Guidelines

This command displays virtual voice-port configurations for a Cisco CME system. Each ephone-dn corresponds to a virtual voice port. For example, the ephone-dn with dn-tag 7 corresponds to virtual voice port 50/0/7. The virtual voice port provides the telephone line associated with the Cisco IP phone extension (ephone-dn).

Examples

The following is sample output from this command:

```
Router# show telephony-service voice-port

voice-port 50/0/1
 station-id number 5001
!
voice-port 50/0/2
 station-id number 5002
 timeout ringing 8
!
voice-port 50/0/3
 station-id number 5003
!
voice-port 50/0/4
 station-id number 5004
!
```

Table 43 describes significant fields in this output, in alphabetical order.

Table 43 show telephony-service voice-port Field Descriptions

Field	Description
station-id number	Phone number used for caller ID purposes for calls made from this voice port.
timeout ringing	Maximum amount of time that a phone is allowed to ring before the call is disconnected.
voice-port	Virtual voice port.

Related Commands

Command	Description
show telephony-service all	Displays the detailed configuration of all the Cisco IP phones.
show telephony-service dial-peer	Displays dial-peer information for extensions in a Cisco CME system.
show telephony-service ephone-dn	Displays information for extensions (ephone-dns) in a Cisco CME system.

show voice register all

To display all Cisco Unified Session Initiation Protocol (SIP) Survivable Remote Site Telephony (SRST) and 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.

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 P0S3-07-4-00
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
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
keep-conference is enabled

Dialpeers created:
```

```
Statistics:
Active registrations : 0

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


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

Dialpeers created:

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

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

Statistics:
Active registrations : 2

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


Nothing configured yet
Pool Tag 8
Config:
```

```
Mac address is 0006.D737.CC42
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 47 describes significant fields shown in this output.

Table 44 show voice register all Field Descriptions

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

Table 44 show voice register all Field Descriptions (continued)

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

Related Commands

Command	Description
show sip-ua status registrar	Displays all the SIP endpoints currently registered with the contact address.
show voice register all	Displays all Cisco 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 credential

To display configuration information associated with a credential file used for authorization, use the **show voice register credential** command in privileged EXEC mode.

show voice register credential

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

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

Examples

The following is sample output from this command:

```
Router# show voice register credential

username: Jsmith, password: 1234abc, service: PRESENCE , file index 3
username: Ksample, password: xyz1234, service: PRESENCE , file index 3
username: Mmore, password: updwssc, service: PRESENCE , file index 3
username: Sstove, password: 12bms, service: PRESENCE , file index 3
username: Yjones, password: 3571lvrus, service: PRESENCE , file index 5
username: Yjones2, password: 55rrtuv, service: PRESENCE OOD_REFER , file index 5
username: vtemp, password: 1234567, service: PRESENCE , file index 5
```

Table 49 contains descriptions of fields shown in the output, listed in order of appearance.

Table 45 show voice register credential Field Descriptions

Field	Description
username	Username that is authorized.
password	Password that is authorized.
service	Type of service for which the credential file is used; presence or Out-of-dialog REFER (OOD-R).
file index	Identification number of the credential file defined with the **authenticate** command.

Related Commands

Command	Description
authenticate (voice register global)	Defines the authenticate mode for SIP phones in a Cisco Unified CME system.

Command	Description
credential load	Reloads a credential file into Flash memory.
show voice register all	Displays all Cisco Unified CME and Cisco Unified SIP SRST configurations and register information.

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

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 Cisco SIP SRST 3.4	This command was added to Cisco CME.

Examples

Cisco Unified SIP SRST

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

```
Router# show voice register dial-peers

dial-peer voice 40024 voip
corlist incoming call91
preference 5
destination-pattern 91011
redirect ip2ip
session target ipv4:192.168.0.2
session protocol sipv2
translate-outgoing called 1
voice-class codec 1

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

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

```
dial-peer voice 40027 voip
preference 1
destination-pattern 91011
redirect ip2ip
session target ipv4:10.2.161.187
session protocol sipv2
voice-class codec 1
monitor probe icmp-ping 10.2.161.187
```

Related Commands

Command	Description
show sip-ua status registrar	Displays all the SIP endpoints currently registered with the contact address.
show voice register all	Displays all Cisco 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 dialplan

To display all configuration information for a specific SIP dial plan, use the **show voice register dialplan** command in privileged EXEC mode.

show voice register dialplan *tag*

Syntax Description

tag	Number that identifies the SIP dial plan. Range: 1 to 24.

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

Use this command to verify the configuration of SIP dial plans. You define a dial plan with the **voice register dialplan** command and assign it to a SIP phone with the **dialplan** command.

Examples

The following is sample output from this command displaying information for dial plan 1:

```
Router# show voice register dialplan 1

Dialplan Tag 1
Config:
  Type is 7940-7960-others
  Pattern 1 is 2..., timeout is 10, user option is ip, button is default
  Pattern 2 is 1234, timeout is 0, user option is ip, button is 4
  Pattern 3 is 65..., timeout is 0, user option is phone, button is default
  Pattern 4 is 1..., timeout is 0, user option is phone, button is default
```

Table 47 contains descriptions of significant fields shown in this output, listed in alphabetical order.

Table 46 show voice register dialplan Field Descriptions

Field	Description
Config:	List of configuration options defined for this SIP dial plan.
Dialplan Tag	Tag number of the requested SIP dial plan.
Pattern	Dial pattern defined for a SIP dial plan with the **pattern** command in voice register dialplan configuration mode.
Type	Phone type defined for a SIP dial plan with the **type** command.

Related Commands

Command	Description
dialplan	Assigns a dial plan to a SIP phone.
pattern (voice register dialplan)	Defines a dial pattern for a SIP dial plan.
show voice register all	Displays all Cisco Unified CME configurations and register information.
show voice register pool	Displays all configuration information associated with a particular voice register pool.
type (voice register dialplan)	Defines a phone type for a SIP dial plan.
voice register dialplan	Enters voice register dialplan configuration mode to define a dial plan for SIP phones.
voice register pool	Enters voice register pool configuration mode for SIP phones.

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*

Syntax Description

tag	Tag number of the voice register dn for which to display information. Range is 1 to 750.

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.

Usage Guidelines

This command can also be used for Cisco SIP SRST.

Examples

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

```
Router# show voice register dn 148

Dn Tag 148
Config:
  Number is 1100
  Preference is 0
  Huntstop is disabled
  Auto answer is disabled
```

Table 47 contains descriptions of significant fields shown in this output, listed in alphabetical order.

Table 47 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 sip-ua status registrar	Displays all the SIP endpoints currently registered with the contact address.
show voice register all	Displays all Cisco SIP SRST and Cisco CME configurations and register information.
show voice register dial-peer	Displays details of all dynamically created VoIP dial peers associated with the Cisco SIP SRST or Cisco CME register event.
show voice register pool	Displays all configuration information associated with a particular voice register pool.
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 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.

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 P0S3-07-4-07
Time-format is 12
Date-format is M/D/Y
Time-zone is 5
Hold-alert is disabled
Mwi stutter is disabled
Mwi registration for full E.164 is disabled
Forwarding local is enabled
Dst auto adjust is enabled
 start at Apr week 1 day Sun time 02:00
 stop  at Oct week 8 day Sun time 02:00
Max redirect number is 5
Telnet Level: 2
Tftp path is system:/cme/sipphone
Generate text file is disabled
Tftp files are created, current syncinfo 0002830590524159
OS79XX.TXT is not created
Router#
```

Cisco 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 48 contains descriptions of significant fields shown in this output, listed in alphabetical order.

Table 48 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.
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 particular voice register pool, use the **show voice register pool** command in privileged EXEC mode.

show voice register pool *pool-tag*

Syntax Description

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

Command Modes

Privileged EXEC

Command History

Cisco IOS Release	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	This command was introduced.

Examples

Cisco Unified CME

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

```
Router# show voice register pool 33
Pool Tag 33
Config:
 Mac address is 0009.B7F7.532E
 Type is 7960
 Number list 1 : DN 1
 Number list 2 : DN 2
 Number list 3 : DN 3
 Number list 4 : DN 4
 Number list 5 : DN 5
 Number list 6 : DN 6
 Proxy Ip address is 0.0.0.0
 DTMF Relay is disabled
 Call Waiting is enabled
 keep-conference is enabled
 template is 1
```

Cisco Unified SIP SRST

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

```
Router# show voice register pool 1

Pool Tag 1
Config:
Network address is 192.168.0.0, Mask is 255.255.0.0
Number list 1 : Pattern is 50.., Preference is 2
Proxy Ip address is 0.0.0.0
Default preference is 2
```

```
Incoming called number is
Translate outgoing called tag is 1
Class of Restriction List Tag: default
Incoming corlist name is allowall
Application is default.new

Dialpeers created:

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

Statistics:
Active registrations : 2

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

Table 49 contains descriptions of significant fields shown in the Cisco Unified SIP SRST and Cisco Unified CME output, listed in alphabetical order.

Table 49 show voice register pool Field Descriptions

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

Table 49 show voice register pool Field Descriptions (continued)

Field	Description
Incoming corlist name	Used with the **all** and **pool** keywords. Shows the **cor** command configuration.
keep-conference	Status of **keep-conference** command.
Mac address	MAC address of this SIP phone as defined by using the **id** command.
Network address and Mask	Used with the **all** and **pool** keywords. Shows network address and mask information if the **id** command is configured.
Number list, Pattern, and Preference	Used with the **all** and **pool** keywords. Shows the **number** (voice register pool) command configuration.
Pool Tag	Used with the **all** and **pool** keywords. Shows the assigned tag number of the current pool.
Proxy Ip address	Used with the **all** and **pool** keywords. Shows the **proxy** command configuration; that is, the IP address of external SIP server.
Registration failed	Used with the **all**, **pool**, and **statistics** keywords. Shows the failed registrations.
Registration requests	Used with the **all**, **pool**, and **statistics** keywords. Shows the incoming registration requests.
Registration success	Used with the **all**, **pool**, and **statistics** keywords. Shows the successful registrations.
Statistics:	Used with the **all**, **pool**, and **statistics** keywords. Shows the registration statistics for this pool.
Template	Template-tag number for template that is applied to this SIP phone.
Translate outgoing called tag	Used with the **all** and **pool** keywords. Shows the **translate-outgoing** command configuration.
Total Registration Statistics	Used with the **all**, **pool**, and **statistics** keywords. Shows the total registration statistics for this pool.
Type	Phone type identified for this SIP phone using the **type** command.
unRegister requests	Used with the **all**, **pool**, and **statistics** keywords. Shows the incoming unregister/registration expire requests.
unRegister success	Used with the **all**, **pool**, and **statistics** keywords. Reports the number of successful unregisters.
unRegister failed	Used with the **all**, **pool**, and **statistics** keywords. Reports the number of failed unregisters.
Username Password	Values within authentication credential.

Related Commands

Command	Description
show sip-ua status registrar	Displays all the SIP endpoints currently registered with the contact address.
show voice register all	Displays all Cisco 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 pool	Enters voice register pool configuration mode for SIP phones.

show voice register profile

To display the content of configuration files that are in ASCII text format, use the **show voice register profile** command in privileged EXEC mode.

show voice register profile text *tag*

Syntax Description

tag	Unique identifier for voice register profile to be displayed. Range is 1–500.

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

Use this command to display ASCII configuration files for the Cisco IP Phone 7905 and 7905G, Cisco IP Phone 7912 and 7912G, Cisco ATA-186, or Cisco ATA-188. To generate ASCII text files, use the **file text** command.

Examples

The following is sample output from this command displaying information in the configuration profile for voice register pool 4:

```
Router# show voice register profile text 4
 Pool Tag: 4
#txt
 AutoLookUp:0
 DirectoriesUrl:0
…
 CallWaiting:1
 CallForwardNumber:0
 Conference:1
 AttendedTransfer:1
 BlindTransfer:1
…
 SIPRegOn:1
 UseTftp:1
 UseLoginID:0
 UIPassword:0
 NTPIP:0.0.0.0
 UID:2468
…
```

Table 50 contains descriptions of significant fields shown in this output, listed in alphabetical order.

Table 50 show voice register profile Field Descriptions

Field	Description
Attended Transfer	Setting of soft key for attended transfer in a SIP phone template as defined by using the **transfer-attended** command. “1” indicates that the soft key is enabled; “0” indicates that the soft key is disabled.
Auto Lookup	1 indicates that Auto Lookup is enabled. 0 indicates that it is disabled.
Blind Transfer	Setting of soft key for blind transfer in a SIP phone template as defined by using the **transfer-blind** command. “1” indicates that the soft key is enabled; “0” indicates that the soft key is disabled.
Call Waiting	Setting of the call-waiting option on a SIP phone as defined by using the **call-waiting** command. “1” indicates that the soft key is enabled; “0” indicates that the soft key is disabled.
Call Forward Number	Number to which incoming calls are forwarded
Conference	Setting of soft key for conference in a SIP phone template as defined by using the **conference** command. “1” indicates that the soft key is enabled; “0” indicates that the soft key is disabled.
Directories URL	1 indicates that the Directories feature button for the phone is enabled. 0 indicates that it is disabled.
NTPIP	IP address for the NTP source
Pool tag	Pool tag of the configuration file being requested.
SIP Reg On	1 indicates that the registration with external proxy server for the phone is enabled. 0 indicates that it is disabled.
UI Password	1 indicates that the UI password is enabled on the phone. 0 indicates that dit is disabled.
UID	Authenticatuion credential for SIP phone.
Use Login ID	1 indicates that “use login id” for phone is enabled. 0 indicates that it is disabled.

Related Commands

Command	Description
create profile (voice register global)	Generates the configuration profiles required for SIP phone.
file text (voice register global)	Generates ASCII text files for the Cisco IP Phone 7905 and 7905G, Cisco IP Phone 7912 and 79012G, Cisco ATA-186, or Cisco ATA-188.
reset (voice register global)	Performs a complete reboot of all SIP phones associated with a Cisco CME router.
voice register global	Enters voice register global configuration mode in order to set global parameters for all supported Cisco SIP phones in a Cisco CME or Cisco SIP SRST environment.

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

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 Cisco SIP SRST 3.4	This command was added to Cisco CME.

Usage Guidelines

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

Examples

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

```
Router# show voice register statistics

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

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

Table 47 describes significant fields shown in this output.

Table 51 show voice register statistics Field Descriptions

Field	Description
Statistics:	Used with the **all**, **pool**, and **statistics** keywords. Shows the registration statistics for this pool.
Active registrations	Used with the **all**, **pool**, and **statistics** keywords. Shows the current active registrations.
Total Registration Statistics	Used with the **all**, **pool**, and **statistics** keywords. Shows the total registration statistics for this pool.
Registration requests	Used with the **all**, **pool**, and **statistics** keywords. Shows the incoming registration requests.
Registration success	Used with the **all**, **pool**, and **statistics** keywords. Shows the successful registrations.
Registration failed	Used with the **all**, **pool**, and **statistics** keywords. Shows the failed registrations.
unRegister requests	Used with the **all**, **pool**, and **statistics** keywords. Shows the incoming unregister/registration expire requests.
unRegister success	Used with the **all**, **pool**, and **statistics** keywords. Reports the number of successful unregisters.
unRegister failed	Used with the **all**, **pool**, and **statistics** keywords. Reports the number of failed unregisters.
Global statistics	Used with the **statistics** keyword. Details all active registrations.
Register pool *number* statistics	Used with the **statistics** keyword. Details specific pool statistics.

Related Commands

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

show voice register template

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

show voice register template *template-tag*

Syntax Description

template-tag	Tag number of the template for which to display information. Range is 1 to 5.

Command Modes

Privileged EXEC

Command History

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

Examples

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

```
Router# show voice register template 1

Temp Tag 1
Config:
 Attended Transfer is enabled
 Blind Transfer is enabled
 Conference is enabled
 Caller-ID block is enabled
 DnD control is enabled
 Anonymous call block is enabled
 Vad is enabled
 Voicemail is 56789, timeout 15
```

Table 52 contains descriptions of significant fields shown in this output, listed in alphabetical order.

Table 52 show voice register template Field Descriptions

Field	Description
Anonymous call block	Status of anonymous caller blocking defined with the **anonymous block** command.
Attended Transfer	Status of attended transfer soft key defined with the **transfer-attended** command.
Blind Transfer	Status of blind transfer soft key defined with the **transfer-blind** command.
Conference	Status of conference soft key defined with the **conference** command.
Config:	List of configuration options defined for this template.
Caller-ID block	Status of caller-id feature defined with the **caller-id block** command.
Dnd controls	Status of Do-Not-Disturb soft key defined with the **dnd-control** command.

Table 52 show voice register template Field Descriptions (continued)

Field	Description
Temp Tag	Tag number of the requested template.
VAD	Status of voice activity detection defined with the **vad** command
Voicemail	Voicemail extension and timeout value defined with the **voice-mail** command.

Related Commands

Command	Description
show voice register all	Displays all voice register information, including statistics, pools, and dial peers.
voice register template	Enters voice register template configuration mode and defines a template of common parameters for SIP phones.

show voice register tftp-bind

To display the current configuration files accessible to SIP phones, use the **show voice register tftp-bind** command in privileged EXEC mode.

show voice register tftp-bind

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

This command provides a list of configuration files that are accessible to SIP phones using TFTP.

Examples

The following is sample output from this command:

```
Router(config)# show voice register tftp-bind
 tftp-server SIPDefault.cnf url system:/cme/sipphone/SIPDefault.cnf
 tftp-server syncinfo.xml url system:/cme/sipphone/syncinfo.xml
 tftp-server SIP0009B7F7532E.cnf url system:/cme/sipphone/SIP0009B7F7532E.cnf
 tftp-server SIP000ED7DF7932.cnf url system:/cme/sipphone/SIP000ED7DF7932.cnf
 tftp-server SIP0012D9EDE0AA.cnf url system:/cme/sipphone/SIP0012D9EDE0AA.cnf
 tftp-server gk123456789012 url system:/cme/sipphone/gk123456789012
 tftp-server gk123456789012.txt url system:/cme/sipphone/gk123456789012.txt
```

Table 53 contains descriptions of significant fields shown in this output, listed in alphabetical order.

Table 53 show voice register tftp-bind Field Descriptions

Field	Description
ata<mac-address>	Cisco SIP configuration profile for a particular Cisco ATA-186 or Cisco ATA-188 as indicated by the <mac-address>. This file is generated by using the **create profile** command.
ata<mac-address>.txt	ASCII text file of a Cisco SIP configuration profile for a particular Cisco ATA-186 or Cisco ATA-188 as indicated by the <mac-address>. This file is generated by using the **file text** command.
gk<mac-address>	Cisco SIP configuration profile for a particular Cisco IP Phone 7912 or Cisco IP Phone 7912G as indicated by the <mac-address>. This file is generated by using the **create profile** command.

Table 53 show voice register tftp-bind Field Descriptions (continued)

Field	Description
gk<mac>.txt	ASCII text file of a Cisco SIP configuration profile for a particular Cisco IP Phone 7912 or Cisco IP Phone 7912G as indicated by the <mac-address>. This file is generated by using the **file text** command.
Id<mac-address>	Cisco SIP configuration profile for a particular Cisco IP Phone 7905 or Cisco IP Phone 7912G as indicated by the <mac-address>. This file is generated by using the **create profile** command.
Id<mac-address>.txt	ASCII text file of a Cisco SIP configuration profile for a particular Cisco IP Phone 7905 or Cisco IP Phone 7912G as indicated by the <mac-address>. This file is generated by using the **file text** command.
SIPDefault.cnf	Configuration file to be shared by all Cisco SIP IP Phone 7940s and Cisco SIP IP Phone 7960s. This file is automatically generated by the router through the **source-address** command and is placed in router memory. The SIPDefault.cnf file contains the IP address that the phones use to register for service, using the Session Initiation Protocol (SIP).
SIP<mac-address>.cnf	Cisco SIP configuration profile for a particular Cisco IP Phone 7940 or Cisco IP Phone 7960 as indicated by the <mac-address>. This file is generated by using the **create profile** command.
syncinfo.xml	Configuration file to be shared by all Cisco SIP IP Phone 7940s and Cisco SIP IP Phone 7960s. This file is generated by using the **create profile** command.

Related Commands

Command	Description
create profile (voice register global)	Generates the configuration profiles required for SIP phones.
reset (voice register dn)	Performs a complete reboot of one phone associated with a Cisco CME router.
reset (voice register pool)	Performs a complete reboot of one or all phones associated with a Cisco CME router.
text file (voice register global)	Generates an ASCII format text file of the Cisco SIP configuration profile for Cisco IP Phone 7905s and 7905Gs, Cisco IP phone 7912s and 7912Gs, Cisco ATA-186s, and Cisco ATA-188s.
tftp-path (voice register global)	Specifies the directory to which the provisioning file for SIP phones in a Cisco CallManager Express (Cisco CME) system will be written.
voice register global	Enters voice register global configuration mode in order to set global parameters for all supported Cisco SIP phones in a Cisco CME or Cisco SIP SRST environment.

show voice-huntgroup

To display configuration information associated with one or all voice hunt groups in a Cisco CallManager (Cisco CME) system, use the **show voice-huntgroup** command in privileged EXEC mode.

show voice-huntgroup *hunt-tag* [**brief**] {**longest-idle** | **parallel** | **peer** | **sequential**}

Syntax Description

hunt-tag	(Optional) Unique sequence number that identifies the voice hunt group. Range is 1 to 100
brief	(Optional) To display brief information about one or all voice hunt groups in a Cisco CME system.
longest-idle	(Optional) To display summary of one or all longest-idle voice hunt groups.
parallel	(Optional) To display summary of one or all parallel voice hunt groups.
peer	(Optional) To display summary of one or all peer voice hunt groups.
sequential	(Optional) To display summary of one or all sequential voice hunt groups.

Command Modes

Privileged EXEC

Command History

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

Usage Guidelines

Use this command instead of using show running-config, as an alternate way to get information about voice hunt-group configuration on the gateway.

Use the **show voice-huntgroup** and **show voice-huntgroup brief** commands to display hunt group configuration information for all voice hunt groups in a Cisco CME system. Use the **show voice-huntgroup** *tag* command to display data regarding a specific hunt-tag configuration created by the **voice hunt-group** command. Use the **longest-idle**, **parallel**, **peer**, or **sequential** keywords to display data regarding a specific type of voice hunt group configuration created by the **voice hunt-group** command.

Examples

The following example is a sample of the output from this command when there is no voice hunt-group configured:

```
router# show voice hunt-group
 no voice hunt-groups configured
router# show voice hunt-group brief
 no voice hunt-groups configured
router# show voice hunt-group longest-idle
 no voice hunt-groups configured
router#
```

The following example is the sample output from this command displaying the configuration for all the configured voice hunt-groups:

```
router# show voice hunt-group
Group 5
 type: parallel
 pilot number: 1234, peer-tag 1234
 list of numbers: 9498889994,9498889993,
 secondary number: 5678, peer-tag 5678
 list preference: 5
 preference (sec): 8
 timeout: 180
 final_number: 4444

Group 8
 type: longest-idle
 pilot number: 6666, peer-tag 6666
 list of numbers: 5106575902,4088531111,4083911375,4089306067,8869395033,88686619633
 preference: 0
 preference (sec): 0
 timeout: 180
 final_number:
 hops: 6

Group 10
 type: longest-idle
 pilot number: 7777777, peer-tag 7777777
 secondary number: 88888888, peer-tag 88888888
 list of numbers: 7654321,87654321,987654321,
 preference: 0
 preference (sec): 0
 timeout: 180
 final_number:
 hops: 3

Group 15
 type: peer
 pilot number: 56789, peer-tag 56789
 list of numbers: 87654321,9876,87654,
 preference: 0
 preference (sec): 0
timeout: 180
 final_number:
 hops: 3
```

The following is sample output from this command displaying information for a particular voice hunt group as specified by a *hunt- tag* number:

```
Router# show voice hunt-group 5
Group 5
 type: parallel
 pilot number: 1234, peer-tag 1234
 secondary number: 5678, peer-tag 5678
 list of numbers: 9498889994,9498889993,
 preference: 5
 preference (sec): 8
 timeout: 20
 final_number: 4444
```

The following is sample output from this command displaying information about all the voice hunt groups of a particular type:

```
router# sh voice hunt-group longest-idle
Group 8
 type: longest-idle
 pilot number: 6666, peer-tag 6666
 list of numbers: 5106575902,4088531111,4083911375,4089306067,8869395033,88686619633,
```

```
 preference: 0
 preference (sec): 0
 timeout: 180
 final_number:
 hops: 6
Group 10
 type: longest-idle
 pilot number: 7777777, peer-tag 7777777
 secondary number: 88888888, peer-tag 88888888
 list of numbers: 7654321,87654321,987654321,
 preference: 0
 preference (sec): 0
 timeout: 180
 final_number:
 hops: 3
```

The following example is a sample output of this command plus the brief keyword:

```
router# show voice-huntgroup brief
TAG TYPE PILOT     LIST
=== ==== ========  =====================================================
5   PAR  1234      9498889-, 9498889-
         5678      9498889-, 9498889-
8   LON  6666      5106575-, 4088531-, 4083911-, 4089306-, 8869395-,.....
10  LON  7777777   7654321, 8765432-, 9876543-
         8888888-  7654321, 8765432-, 9876543-
15  PER  56789     8765432-, 9876, 87654
```

Table 54 contains descriptions of significant fields shown in this output, listed in alphabetical order.

Table 54 show voice-huntgroup Field Descriptions

Field	Description
Final_number	Last number in the voice hunt group, after which a call is no longer redirected
Hops	Number of hops before a call proceeds to the final number.
List of numbers	Numbers of the extensions configured in the **voice hunt-group** command's hunt-tag identifier.
Pilot number	Number that callers dial to reach the voice hunt group.
Preference	Preference order for the extension or telephone number associated with a dial peer. Range is 0 to 8. Default is 0.
Preference (sec)	Preference order for the secondary pilot number. Range is from 0 to 10, where 0 is the highest preference and 10 is the lowest preference. Default is 9.
Secondary number	Additional pilot number for the voice hunt group.
Timeout	Number of seconds after which a call that is not answered at one number is redirected to the next number in the hunt-group list.
Type	**voice hunt-group** command type. Can be longest-idle, parallel, peer, or sequential.

Related Commands

final (voice hunt-group)	Defines the last extension in a voice hunt group.
hops (voice hunt-group)	Defines the number of times that a call is redirected to the next directory number in a peer voice hunt-group list before proceeding to the final directory number.
list (voice hunt-group)	Defines the directory numbers that participate in a directory number hunt group.
pilot **(voice hunt-group)**	Defines the voice dn that callers dial to reach a Cisco CallManager Express (Cisco CME) voice hunt group.
timeout (voice hunt-group)	Sets the number of seconds after which a call that is not answered is redirected to the next number in the hunt-group list and defines the last directory number in the hunt group.
voice hunt-group	Defines the type of hunt group.

softkeys alerting

To configure an ephone template for soft-key display during the alerting call stage, use the **softkeys alerting** command in ephone-template configuration mode. To remove a **soft key alerting** configuration, use the **no** form of this command.

softkeys alerting {[**Acct**] [**Callback**] [**Endcall**]}

no softkeys alerting

Syntax Description

Acct	(Optional) Soft-key name that appears on the IP phone during the alerting call stage. Short for "account code." Provides access to configured accounts.
Callback	(Optional) Soft-key name that appears on the IP phone during the alerting call stage. Requests callback notification when a busy called line becomes free.
Endcall	(Optional) Soft-key name that appears on the IP phone during the alerting call stage. Ends the current call.

Defaults

The default soft keys for the alerting call stage and the order in which they appear on IP phones are, from first to last, Acct, Callback, and Endcall.

Command Modes

Ephone-template configuration

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.3(11)T	3.2	This command was introduced.

Usage Guidelines

The alerting call stage is when the remote point is being notified of an incoming call, and the status of the remote point is being relayed to the caller as either ringback or busy.

The number and order of soft keys listed in the **softkeys alerting** command correspond to the number and order of soft keys that will appear on IP phones.

Examples

In the following example, ephone template 1 is configured for the alerting stage and for the seized and connected call stages:

```
Router(config)# telephony-service
Router(config-telephony)# ephone-template 1
Router(config-ephone-template)# softkeys seized Redial Cfwdall Pickup
Router(config-ephone-template)# softkeys alerting Callback Endcall
Router(config-ephone-template)# softkeys connected Confrn Hold Endcall
```

Related Commands

Command	Description
ephone-template (ephone)	Applies an ephone template to an ephone.
softkeys connected	Configures an ephone template for soft-key display during the connected call stage.
softkeys idle	Configures an ephone template for soft-key display during the idle call stage.
softkeys seized	Configures an ephone template for soft-key display during the seized call stage.

softkeys connected

To configure an ephone template for soft-key display during the connected call stage, use the **softkeys connected** command in ephone-template configuration mode. To remove a **softkeys connected** configuration, use the **no** form of this command.

softkeys connected {[**Acct**] [**ConfList**] [**Confrn**] [**Endcall**] [**Flash**] [**HLog**] [**Hold**] [**Join**] [**Park**] [**RmLstC**] [**Select**] [**Trnsfer**]}

no softkeys connected

Syntax Description

Acct	(Optional) Soft-key name that appears on the IP phone during the connected call stage. Provides access to configured accounts.
ConfList	(Optional) Soft-key name that appears on the IP phone during the connected call stage. Lists all parties in a conference.
Confrn	(Optional) Soft-key name that appears on the IP phone during the connected call stage. Connects callers to a conference call.
Endcall	(Optional) Soft-key name that appears on the IP phone during the connected call stage. Ends the current call.
Flash	(Optional) Soft-key name that appears on the IP phone during the connected call stage. Also called “hookflash.” Provides hookflash functionality for public switched telephony network (PSTN) services on calls connected to the PSTN via a foreign exchange office (FXO) port.
HLog	(Optional) Soft-key name that appears on the IP phone during the connected call stage. Places a phone into not-ready status, in which it does not accept hunt-group calls. You must set the **hunt-group logout** command to HLog for this soft key to be visible. This key is a toggle; pressing it a second time returns the phone to ready status, in which it is available to receive calls.
Hold	(Optional) Soft-key name that appears on the IP phone during the connected call stage. Places an active call on hold and resumes the call.
Join	(Optional) Soft-key name that appears on the IP phone during the connected call stage. Joins an established call to conference.
Park	(Optional) Soft-key name that appears on the IP phone during the connected call stage. Places an active call on hold, so it can be retrieved from another phone in the system.
RmLstC	(Optional) Soft-key name that appears on the IP phone during the connected call stage. Removes the last party added to the conference. This soft key only works for the conference creator.
Select	(Optional) Soft-key name that appears on the IP phone during the connected call stage. Selects a call or a conference on which to take action.
Trnsfer	(Optional) Soft-key name that appears on the IP phone during the connected call stage. Transfers active calls to another extension.

Command Default

The default soft keys for the connected call stage and the order in which they appear on IP phones are, from first to last:

- With HLog support: Hold, EndCall, Trnsfer, Confrn, Acct, HookFlash, Park, HLog

- Without HLog support: Hold, EndCall, Trnsfer, Confrn, Acct, HookFlash, Park, Resume, NewCall

Command Modes

Ephone-template configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)T	Cisco CME 3.2	This command was introduced.
12.4(4)XC	Cisco Unified CME 4.0	The **HLog** keyword was added.
12.4(9)T	Cisco Unified CME 4.0	This command with modifications was integrated into Cisco IOS Release 12.4(9)T.
12.4(11)XJ	Cisco Unified CME 4.1	The **ConfList**, **Join**, **RmLstC**, and **Select** keywords were added.
12.4(15)T	Cisco Unified CME 4.1	This command was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

The connected call stage is when the connection to a remote point has been established.

The number and order of soft keys listed in the **softkeys connected** command correspond to the number and order of soft keys that will appear on IP phones.

Configure the ConfList, Join, and RmLstC soft keys for conferencing functions. Use these soft keys with hardware conferencing only.

Note The ConfList (including the Remove, Update, and Exit soft keys within the ConfList function) and RmLstC soft keys do not work on the Cisco Unified IP Phone 7902 and Cisco Unified IP Phone 7935 and 7936.

Examples

In the following example, ephone template 1 is configured for the connected stage and for the seized and alerting call stages:

```
Router(config)# ephone-template 1
Router(config-ephone-template)# softkeys seized Redial Cfwdall Pickup
Router(config-ephone-template)# softkeys alerting Callback Endcall
Router(config-ephone-template)# softkeys connected Confrn Hold Endcall
```

Related Commands

Command	Description
ephone	Enters ephone configuration mode for an IP phone.
ephone-template	Declares and names an ephone template to configure IP phone soft-key display and to enter ephone-template configuration mode.
ephone-template (ephone)	Applies an ephone template to an ephone.
fxo-hook-flash	Enables display of the Flash soft key.
hunt-group logout	Enables separate handling of DND and HLog functionality for hunt-group agents and the display of the HLog soft key on phones.

Command	Description
softkeys alerting	Configures an ephone template for soft-key display during the alerting call stage.
softkeys idle	Configures an ephone template for soft-key display during the idle call stage.
softkeys seized	Configures an ephone template for soft-key display during the seized call stage.

softkeys connected (voice register template)

To modify the soft-key display for the connected call state on SIP phones, use the **softkeys connected** command in voice register template configuration mode. To remove a **softkeys connected** configuration, use the **no** form of this command.

softkeys connected {[**Confrn**] [**Endcall**] [**Hold**] [**Trnsfer**]}

no softkeys connected

Syntax Description

Confrn	(Optional) Soft-key name that appears on the IP phone during the connected call state. Short for "conference." Connects callers to a conference call.
Endcall	(Optional) Soft-key name that appears on the IP phone during the connected call state. Ends the current call.
Hold	(Optional) Soft-key name that appears on the IP phone during the connected call state. Places an active call on hold and resumes the call.
Trnsfer	(Optional) Soft-key name that appears on the IP phone during the connected call state. Short for "call transfer." Transfers active calls to another extension.

Command Default

The default soft keys for the connected call state and the order in which they appear on SIP phones are, from first to last, Hold, Endcall, Trnsfer, and Confrn.

Command Modes

Voice register template configuration

Command History

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

Usage Guidelines

The connected call state is when the connection to a remote point is established.

The number and order of soft keys used in this command correspond to the number and order of soft keys that will appear on SIP phones. Any soft key that is not explicitly specified with this command is disabled if this command is used to change the default soft keys.

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

Examples

In the following example, SIP phone template 1 is configured for the connected and seized call states:

```
Router(config)# voice register template 1
Router(config-register-template)# softkeys seized Redial Cfwdall EndCall
Router(config-register-template)# softkeys connected Confrn Hold Endcall
```

Related Commands

Command	Description
softkeys hold (**voice register template**)	Configures a SIP phone template for soft-key display during the hold call state.
softkeys idle (**voice register template**)	Configures a SIP phone template for soft-key display during the idle call state.
softkeys seized (**voice register template**)	Configures a SIP ephone template for soft-key display during the seized call state.
template (**voice register pool**)	Applies a phone template to a SIP phone.

softkeys hold

To configure an ephone template to modify soft-key display during the call-hold call stage, use the **softkeys hold** command in ephone-template configuration mode. To remove a **softkeys hold** configuration, use the **no** form of this command.

softkeys hold {[**Join**] [**Newcall**] [**Resume**] [**Select**]}

no softkeys hold

Syntax Description

Join	(Optional) Soft-key name that appears on an IP phone during the hold call stage. Joins an established call to a conference.
Newcall	(Optional) Soft-key name that appears on an IP phone during the hold call stage. Opens a line on a speaker phone to place a new call.
Resume	(Optional) Soft-key name that appears on an IP phone during the hold call stage. Reconnects with the call on hold.
Select	(Optional) Soft-key name that appears on an IP phone during the hold call stage. Selects a call or a conference on which to take action.

Command Default

The default soft keys for the hold call stage and the order in which they appear on IP phones are alphabetical, from first to last, Join, Newcall, Resume, and Select.

Command Modes

Ephone-template configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.
12.4(11)XJ	Cisco Unified CME 4.1	The **Join** and **Select** keywords were added.
12.4(15)T	Cisco Unified CME 4.1	This command was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

You reach the call-hold state by pressing the Hold soft key while you are in the connected state. From the hold state, you can press Resume to return to the connected state or NewCall to start another call, leaving the original call in the call-hold state.

The number and order of soft keys listed in the **softkeys hold** command correspond to the number and order of soft keys that will appear on IP phones.

Configure the Join and Select soft keys for conferencing functions. These soft keys should be used with hardware conferencing only.

Examples

In the following example, ephone template 1 is configured for the idle, alerting, connected, and hold call stages. It is applied to ephone 25. When ephone 25 has a call on hold, the only soft key that will be available is the Resume soft key.

```
Router(config)# telephony-service
Router(config-telephony)# ephone-template 1
Router(config-ephone-template)# softkeys idle Redial Cfwdall Pickup
Router(config-ephone-template)# softkeys alerting Callback Endcall
Router(config-ephone-template)# softkeys connected Confrn Hold Endcall
Router(config-ephone-template)# softkeys hold Resume
Router(config-ephone-template)# exit

Router(config)# ephone 25
Router(config-ephone)# button 1:39
Router(config-ephone)# ephone-template 1
```

Related Commands

Command	Description
ephone	Enters ephone configuration mode for an IP phone.
ephone-template	Declares and names an ephone template to configure IP phone soft-key display and enters ephone-template configuration mode
ephone-template (ephone)	Applies an ephone template to an ephone.
softkeys alerting	Configures an ephone template for soft-key display during the alerting call stage.
softkeys connected	Configures an ephone template for soft-key display during the connected call stage.
softkeys idle	Configures an ephone template for soft-key display during the idle call stage.
softkeys seized	Configures an ephone template for soft-key display during the seized call stage.

softkeys idle

To configure an ephone template for soft-key display during the idle call stage, use the **softkeys idle** command in ephone template configuration mode. To remove a **softkeys idle** configuration, use the **no** form of this command.

softkeys idle {[**Cfwdall**] [**ConfList**] [**Dnd**] [**Gpickup**] [**HLog**] [**Join**] [**Login**] [**Newcall**] [**Pickup**] [**Redial**] [**RmLstC**]}

no softkeys idle

Syntax Description

Cfwdall	(Optional) Soft-key name that appears on the IP phone during the idle call stage. Forwards all calls.
ConfList	(Optional) Soft-key name that appears on the IP phone during the idle call stage. Lists all parties in a conference.
Dnd	(Optional) Soft-key name that appears on the IP phone during the idle call stage. Enables the Do-Not-Disturb features.
Gpickup	(Optional) Soft-key name that appears on the IP phone during the idle call stage. Selectively picks up calls coming into a phone number that is a member of a pickup group.
HLog	(Optional) Soft-key name that appears on the IP phone during the connected call stage. Places a phone into not-ready status, in which it does not accept hunt-group calls. You must set the **hunt-group logout** command to HLog for this soft key to be visible. This key is a toggle; pressing it a second time returns the phone to ready status, in which it is available to receive calls.
Join	(Optional) Soft-key name that appears on the IP phone during the idle call stage. Joins an established call to a conference.
Login	(Optional) Soft-key name that appears on the IP phone during the idle call stage. Provides personal identification number (PIN)-controlled access to restricted phone features.
Newcall	(Optional) Soft-key name that appears on the IP phone during the idle call stage. Opens a line on a speaker phone to place a new call.
Pickup	(Optional) Soft-key name that appears on the IP phone during the idle call stage. Selectively picks up calls coming into another extension.
Redial	(Optional) Soft-key name that appears on the IP phone during the idle call stage. Redials the last number dialed.
RmLstC	(Optional) Soft-key name that appears on the IP phone during the idle call stage. Removes the last party added to the conference. This soft key only removes the last party when the conference creator presses it.

Command Default

The default soft keys for the idle call stage and the order in which they appear on IP phones are:

- FXO Trunk: Redial, NewCall, DoNotDisturb
- With HLog support: Redial, NewCall, CFwdAll, CallPickUp, GrpCallPickUp, DoNotDisturb, Login, HLog
- Without HLog support: Redial, NewCall, CFwdAll, CallPickUp, GrpCallPickUp, DoNotDisturb, Login

Command Modes

Ephone-template configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)T	Cisco CME 3.2	This command was introduced.
12.4(4)XC	Cisco Unified CME 4.0	The **HLog** keyword was added.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.
12.4(11)XJ	Cisco Unified CME 4.1	The **ConfList**, **Join**, and **RmLstC** keywords were added.
12.4(15)T	Cisco Unified CME 4.1	This command was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

The idle calling stage occurs before a call is made and after a call is complete.

The number and order of soft keys listed in the **softkeys idle** command correspond to the number and order of soft keys on IP phones.

Configure the ConfList, Join, and RmLstC soft keys for conferencing functions. These soft keys should be used with hardware conferencing only.

Note The ConfList (including the Remove, Update, and Exit soft keys within the ConfList function) and RmLstC soft keys do not work on the Cisco Unified IP Phone 7902 and Cisco Unified IP Phone 7935 and 7936.

Examples

In the following example, ephone template 1 is configured for the idle stage and for the alerting and connected call stages:

```
Router(config)# telephony-service
Router(config-telephony)# ephone-template 1
Router(config-ephone-template)# softkeys idle Redial Cfwdall Pickup
Router(config-ephone-template)# softkeys alerting Callback Endcall
Router(config-ephone-template)# softkeys connected Confrn Hold Endcall
```

Related Commands

Command	Description
ephone	Enters ephone configuration mode for an IP phone.
ephone-template	Declares and names an ephone template to configure IP phone soft-key display and enters ephone-template configuration mode
ephone-template (ephone)	Applies an ephone template to an ephone.
hunt-group logout	Enables separate handling of DND and HLog functionality for hunt-group agents and the display of an HLog soft key on phones.
softkeys alerting	Configures an ephone template for soft-key display during the alerting call stage.

Command	Description
softkeys connected	Configures an ephone template for soft-key display during the connected call stage.
softkeys seized	Configures an ephone template for soft-key display during the seized call stage.

softkeys idle (voice register template)

To modify the soft-key display for the idle call state on SIP phones, use the **softkeys idle** command in voice register template configuration mode. To remove a **softkeys idle** configuration, use the **no** form of this command.

softkeys idle {[**Cfwdall**] [**Newcall**] [**Redial**]}

no softkeys idle

Syntax Description

Cfwdall	(Optional) Soft-key name that appears on the IP phone during the idle call state. Short for “call forward all.” Forwards all calls.
Newcall	(Optional) Soft-key name that appears on the IP phone during the idle call state. Opens a line on a speakerphone to place a new call.
Redial	(Optional) Soft-key name that appears on the IP phone during the idle call state. Redials the last number dialed.

Command Default

The default soft keys for the idle call state and the order in which they appear on SIP phones are, from first to last, Redial, Newcall, and Cfwdall.

Command Modes

Voice register template configuration

Command History

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

Usage Guidelines

The idle calling state occurs before a call is made and after a call is complete.

The number and order of soft keys used in this command correspond to the number and order of soft keys that will appear on SIP phones. Any soft key that is not explicitly specified with this command is disabled if this command is used to change the default soft keys.

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

Examples

In the following example, SIP phone template 1 is configured for the idle and connected call states:

```
Router(config)# voice register template 1
Router(config-register-template)# softkeys idle Redial Cfwdall
Router(config-register-template)# softkeys connected Confrn Hold Endcall
```

Related Commands

Command	Description
softkeys connected (voice register template)	Configures a SIP phone template for soft-key display during the connected call state.
softkeys hold (voice register template)	Configures a SIP phone template for soft-key display during the hold call state.
softkeys seized (voice register template)	Configures a SIP phone template for soft-key display during the seized call state.
template (voice register pool)	Applies a phone template to a SIP phone.

softkeys ringing

To configure an ephone template for soft-key display during the ringing call state, use the **softkeys ringing** command in ephone-template configuration mode. To remove the **softkeys ringing** configuration, use the **no** form of this command.

softkeys ringing {[**Answer**] [**Dnd**] [**HLog**]}

no softkeys ringing

Syntax Description

Answer	(Optional) Soft-key name that appears on the IP phone during the ringing call state.
Dnd	(Optional) Soft-key name that appears on the IP phone during the ringing call state.
HLog	(Optional) Soft-key name that appears on the IP phone during the ringing call state.

Command Default

The following soft keys are displayed in alphabetical order, first to last, on IP phones during the ringing call state: Answer, Dnd, HLog

Command Modes

Ephone-template configuration (config-ephone-template)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XW1	Cisco Unified CME 4.2	This command was introduced.

Usage Guidelines

Use this command in ephone-template configuration mode to create a template in which you can specify which soft keys are displayed, and in what order, on an IP phone during the ringing call state. The ringing calling state is when a call is received and before the call is connected.

Any soft key that is not explicitly configured is disabled.

You can enter any of the keywords in any order. The number and order of soft keys listed in the **softkeys ringing** command corresponds to the number and order of soft keys that will appear on IP phones during the ringing call state.

Configure the **Answer** keyword with this command to enable a phone user to answer an incoming call on a line button that is unavailable; for example, if a line button is configured with a dual-line directory number and a call is holding on one channel of the directory number and another call is ringing on the second channel, the phone user can use the Answer soft key to pick up the incoming call on the second channel.

Configure the **HLog** keyword with this command to *display* the Hlog soft key during the ringing call state. To enable HLog softkey *functionality* during the call ringing state, you must also configure the **hunt-group logout HLog** command. If you configure the Hlog soft key and do not configure the

hunt-group logout HLog command, the Hlog soft key appears on the phone screen but is not functional. The HLog softkey is a toggle for enabling or disabling the not-ready status, in which the directory number does not accept hunt-group calls.

Configure the **Dnd** keyword with this command to enable the phone user to place the phone into Do-Not-Disturb mode. Configure the Dnd soft key and the **hunt-group logout DND** command to enable the phone user to invoke DND mode and log the phone out of hunt groups in which it is a member.

For information about hunt groups and hunt-group agents, see the "Configuring Call Coverage" module of the *Cisco Unified CME Administrator Guide*.

To apply an ephone template to phone, configure the **ephone-template (ephone)** command in the ephone configuration mode.

Examples

In the following example, ephone template 1 is configured for the ringing state, and for the alerting and connected call states:

```
Router(config)# telephony-service
Router(config-telephony)# ephone-template 1
Router(config-ephone-template)# softkeys ringing Answer Dnd Hlog
Router(config-ephone-template)# softkeys alerting Callback Endcall
Router(config-ephone-template)# softkeys connected Confrn Hold Endcall
```

Related Commands

Command	Description
dnd feature ring	Allows phone buttons configured with the feature-ring option to not ring when their phones are in do-not-disturb (DND) mode.
ephone-template (ephone)	Applies an ephone template to an ephone.
hunt-group logout	Enables separate handling of DND and HLog functionality for hunt-group agents.
softkeys alerting	Configures an ephone template for soft-key display during the alerting call state.
softkeys connected	Configures an ephone template for soft-key display during the connected call state.
softkeys idle	Configures an ephone template for soft-key display during the idle call state.
softkeys seized	Configures an ephone template for the soft-key display during the seized call state.

softkeys seized

To configure an ephone template for soft-key display during the seized call stage, use the **softkeys seized** command in ephone-template configuration mode. To remove a **softkeys seized** configuration, use the **no** form of this command.

softkeys seized {[**CallBack**] [**Cfwdall**] [**Endcall**] [**Gpickup**] [**HLog**] [**MeetMe**] [**Pickup**] [**Redial**]}

no softkeys seized

Syntax Description

CallBack	(Optional) Soft-key name that appears on the IP phone during the seized call stage. Forwards all calls.
Cfwdall	(Optional) Soft-key name that appears on the IP phone during the seized call stage. Forwards all calls.
Endcall	(Optional) Soft-key name that appears on the IP phone during the seized call stage. Ends the current call.
Gpickup	(Optional) Soft-key name that appears on the IP phone during the seized call stage. Selectively picks up calls coming into a phone number that is a member of a pickup group.
HLog	(Optional) Soft-key name that appears on the IP phone during the connected call stage. Places a phone into not-ready status, in which it does not accept hunt-group calls. You must set the **hunt-group logout** command to HLog for this soft key to be visible. This key is a toggle; pressing it a second time returns the phone to ready status, in which it is available to receive calls.
MeetMe	(Optional) Soft-key name that appears on the IP phone during the seized call stage. Initiates a meet-me conference.
Pickup	(Optional) Soft-key name that appears on the IP phone during the seized call stage. Selectively picks up calls to another extension.
Redial	(Optional) Soft-key name that appears on the IP phone during the seized call stage. Redials the last number dialed.

Command Default

The default soft keys for the seized call stage and the order in which they appear on IP phones are:

- With HLog support: Redial, EndCall, CFwdAll, CallPickUp, GrpCallPickUp, CallBack, HLog
- Without HLog support: Redial, EndCall, CFwdAll, CallPickUp, GrpCallPickUp, CallBack

Command Modes

Ephone-template configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)T	Cisco CME 3.2	This command was introduced.
12.4(4)XC	Cisco Unified CME 4.0	The **HLog** keyword was added.
12.4(9)T	Cisco Unified CME 40	This command was integrated into Cisco IOS Release 12.4(9)T.

Cisco IOS Release	Cisco Product	Modification
12.4(11)XJ	Cisco Unified CME 4.1	The **MeetMe** keyword was added.
12.4(15)T	Cisco Unified CME 4.1	This command was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

The seized calling stage is when a caller is attempting a call but has not yet been connected.

The number and order of soft keys listed in the **softkeys seized** command correspond to the number and order of soft keys on IP phones.

You must configure the MeetMe soft key to initiate a meet-me conference. Use this soft key for hardware conferencing only.

Examples

In the following example, ephone template 1 is configured for the seized stage and for the alerting and connected call stages:

```
Router(config)# telephony-service
Router(config-telephony)# ephone-template 1
Router(config-ephone-template)# softkeys seized Redial Cfwdall Pickup
Router(config-ephone-template)# softkeys alerting Callback Endcall
Router(config-ephone-template)# softkeys connected Confrn Hold Endcall
```

Related Commands

Command	Description
ephone	Enters ephone configuration mode for an IP phone.
ephone-template	Declares and names an ephone template to configure IP phone soft-key display and enters ephone-template configuration mode
ephone-template (ephone)	Applies an ephone template to an ephone.
hunt-group logout	Enables separate handling of DND and HLog functionality for hunt-group agents and the display of an HLog soft key on phones.
softkeys alerting	Configures an ephone template for soft-key display during the alerting call stage.
softkeys connected	Configures an ephone template for soft-key display during the connected call stage.
softkeys idle	Configures an ephone template for soft-key display during the idle call stage.

softkeys seized (voice register template)

To modify the soft-key display for the seized call state on SIP phones, use the **softkeys seized** command in voice register template configuration mode. To remove a **softkeys seized** configuration, use the **no** form of this command.

softkeys seized {[**Cfwdall**] [**Endcall**] [**Redial**]}

no softkeys seized

Syntax Description

Cfwdall	(Optional) Soft-key name that appears on the IP phone during the seized call state. Short for "Call forward all." Forwards all calls.
Endcall	(Optional) Soft-key name that appears on the IP phone during the seized call state. Ends the current call.
Redial	(Optional) Soft-key name that appears on the IP phone during the seized call state. Redials the last number dialed.

Command Default

The default soft keys for the seized call state and the order in which they appear on SIP phones are, from first to last, Redial, Endcall, and Cfwdall.

Command Modes

Voice register template configuration

Command History

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

Usage Guidelines

The seized calling state is when a caller goes offhook and before any other action is taken.

The number and order of soft keys used in this command correspond to the number and order of soft keys that will appear on SIP phones. Any soft key that is not explicitly specified with this command is disabled if this command is used to change the default soft keys.

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

Examples

In the following example, SIP phone template 1 is configured for the seized and connected call states:

```
Router(config)# voice register template 1
Router(config-register-template)# softkeys seized Redial Cfwdall
Router(config-register-template)# softkeys connected Confrn Hold Endcall
```

Related Commands

Command	Description
softkeys connected (voice register template)	Configures a SIP phone template for soft-key display during the connected call state.
softkeys hold (voice register template)	Configures a SIP phone template for soft-key display during the hold call state.
softkeys idle (voice register template)	Configures a SIP phone template for soft-key display during the idle call state.
template (voice register pool)	Applies a template to a SIP phone.

source-addr

To specify the IP address of the certification authority proxy function (CAPF) server on the Cisco Unified CME router, use the **source-addr** command in CAPF-server configuration mode. To return to the default, use the **no** form of this command.

source-addr *ip-address*

no source-addr

Syntax Description

ip-address	IP address of the Cisco Unified CME router.

Command Default

No IP address is entered for the CAPF server in the router configuration.

Command Modes

CAPF-server configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used with Cisco Unified CME phone authentication.

Examples

The following example identifies the IP address for the CAPF server as 10.10.10.1:

```
Router(config)# capf-server
Router(config-capf-server)# source address 10.10.10.1
Router(config-capf-server)# trustpoint-label server25
Router(config-capf-server)# cert-oper upgrade all
Router(config-capf-server)# cert-enroll-trustpoint server12 password 0 x8oWiet
Router(config-capf-server)# auth-mode auth-string
Router(config-capf-server)# auth-string generate all
Router(config-capf-server)# port 3000
Router(config-capf-server)# keygen-retry 5
Router(config-capf-server)# keygen-timeout 45
Router(config-capf-server)# phone-key-size 2048
```

source-address (voice register global)

To identify the IP address and port through which SIP phones communicate with a Cisco CallManager Express (Cisco CME) router, use the **source-address** command in voice register global configuration mode. To disable the router from receiving messages from SIP phones, use the **no** form of this command.

source-address *ip-address* [**port** *port*]

no source-address

Syntax Description

ip-address	Preexisting router IP address, typically one of the addresses of the Ethernet port of the router.
port *port*	(Optional) TCP/IP port number to use for Skinny Client Control Protocol (SCCP). Range is 2000 to 9999. Default is 2000.

Defaults

Port number: 2000

Command Modes

Voice register global configuration

Command History

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

Usage Guidelines

This is a mandatory command. The Cisco CallManager Express router cannot communicate with the Cisco CME phones if the IP address is not provided. If the port number is not provided, the default is port 2000. The IP address is usually the IP address of the Ethernet port to which the phones are connected.

This command enables a router to receive messages from Cisco IP phones through the specified IP address and port.

For systems using ITS V2.1, Cisco CME 3.0, or later versions, the IP phones receive their initial configuration information and phone firmware from the TFTP server associated with the router. The TFTP server address obtained by the Cisco IP phones points to the router IP address. The Cisco IP phones transfer a configuration file called SIPDefault.cnf. This file is automatically generated by the router through the **source-address** command and is placed in router memory. The SIPDefault.cnf file contains the IP address that the phones, using the Session Initiation Protocol (SIP), use to register for service. This IP address corresponds to a valid Cisco CME router IP address (and may be the same as the router TFTP server address).

Examples

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

```
Router(config)# voice register global
Router(config-register-global)# source-address 10.6.21.4 port 6000
```

Related Commands

Command	Description
create profile (voice register global)	Generates the configuration profiles required for SIP phones.
file text (voice register global)	Generates ASCII text files for SIP phones.
tftp-path (voice register global)	Specifies the directory to which the provisioning file for SIP phones in a Cisco CallManager Express (Cisco CME) system will be written.
voice register global	Enters voice register global configuration mode in order to set global parameters for all supported Cisco SIP phones in a Cisco CME or Cisco SIP SRST environment.

speed-dial

To create speed-dial definitions for a Cisco Unified IP phone or analog phone that uses an analog telephone adaptor (ATA) in a Cisco Unified CME system, use the **speed-dial** command in ephone or ephone-template configuration mode. To disable a speed-dial definition, use the **no** form of this command.

speed-dial *speed-tag digit-string* [**label** *label-text*]

no speed-dial *speed-tag*

Syntax Description

speed-tag	Unique sequence number that identifies a speed-dial definition during configuration tasks. Range is from 1 to 33.
digit-string	Digits to be dialed when the speed-dial button is pressed on an IP phone or the digits to be dialed when the associated code is entered from an analog phone with an ATA device. For IP phones, if the first character of this string is the plus sign (+), this speed-dial number is locked and cannot be changed at the phone. If the only character in this string is a pound sign (#), a user-programmable speed-dial button with no speed-dial number attached is defined.
label *label-text*	(Optional) String that contains identifying text to be displayed next to the speed-dial button. Enclose the string in quotation marks if the string contains a space.

Command Default

No speed-dial definitions are created.

Command Modes

Ephone configuration
Ephone-template configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco ITS 1.0	This command was introduced on the Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(2)XT	Cisco ITS 2.0	This command was implemented on the Cisco 1750 and Cisco 1751.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	Cisco ITS 2.0	This command was implemented on the Cisco 2600XM and Cisco 2691.
12.2(11)T	Cisco ITS 2.01	This command was implemented on the Cisco 1760.

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	The number of speed-dial definitions that can be created was increased from 4 to 33. The ability to program speed-dial numbers at the phone and the ability to lock speed-dial numbers were introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.3(11)XL	Cisco CME 3.2.1	This feature was modified to allow IP phones to access more speed-dial numbers than the number of available buttons on their phones and to allow analog phones to access up to 33 speed-dial numbers.
12.3(14)T	Cisco CME 3.3	This command was integrated into Cisco IOS Release 12.3(14)T.
12.4(4)XC	Cisco Unified CME 4.0	This command was made available in ephone-template configuration mode.
12.4(9)T	Cisco Unified CME 4.0	This command in ephone-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

The *speed-tag* argument in this command is a unique identifier for a speed-dial definition on the phone that is being configured.

This command must be followed by a quick reboot of the phone using the **restart** command.

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

This command defines speed-dial numbers that are local to the ephone that is being configured. The **directory entry** command defines additional, systemwide speed-dial numbers.

IP Phones

For IP phones, speed-dial numbers can be defined by administrators using this command and the *digit-string* argument. The numbers are locked if the *digit-string* argument begins with a plus sign (+). Locked numbers cannot be changed at the phone. Speed-dial definitions without speed-dial numbers (those defined with only a pound sign) and speed-dial instances with unlocked *digit-string* arguments can be changed by users at their IP phones. Changes made to speed-dial definitions are saved in the router nonvolatile random-access memory (NVRAM) configuration after a timer-based delay.

On Cisco Unified IP phones, speed-dial definitions are assigned to available extension buttons that have not been assigned to extensions. Speed-dial definitions are assigned in the order of their identifier (tag) numbers. For example, if you define speed-dial 1, it is assigned to the first phone button that is available after the buttons that have been assigned to extensions. If you have used two buttons for extensions on a phone, speed-dial 1 is assigned to the third physical button on the phone. When you define speed-dial 2, it is assigned to the fourth physical button on the phone, and so on.

If more speed-dial definitions are created than are supported by the IP phone setup, the extra speed-dial configurations can be dialed from IP phones using this procedure:

1. With the phone on-hook, an IP phone user presses a two-digit speed-dial code (that is, 05 for the entry with tag 5). A new soft key, Abbr, appears in the phone display.

2. The phone user picks up the phone handset and presses the Abbr soft key. The full telephone number associated with the speed-dial tag is dialed.

Prior to Cisco IOS Releases 12.3(11)XL and 12.3(14)T, speed-dial entries that were in excess of the number of physical phone buttons available were ignored.

Analog Phones

Analog phone users who use a Cisco ATA-186, Cisco ATA-188, or Cisco VG 224 to connect to a Cisco Unified CME system use a different method to access speed-dial numbers. Analog phone users press the asterisk (*) key and the speed-dial identifier (tag number) to dial a speed-dial number. For instance, an analog phone user presses *1 to speed dial the number that has been programmed as speed-dial 1 on that ephone. Analog phones can have up to 33 local speed-dial numbers programmed by the system administrator. The numbers cannot be programmed from the phone.

Prior to Cisco IOS Releases 12.3(11)XL and 12.3(14)T, analog phones were limited to nine speed-dial numbers.)

Examples

The following example sets speed-dial button 2 to dial the phone user's assistant at extension 5001 and locks the setting so that the phone user cannot change it at the phone:

```
Router(config)# ephone 23
Router(config-ephone)# speed-dial 2 +5001 label "Assistant"
```

Related Commands

Command	Description
directory entry	Adds a systemwide phone directory entry or speed-dial entry.
restart (ephone)	Performs a fast reboot of a single IP phone in a Cisco Unified CME system.
restart (telephony-service)	Performs a fast reboot of one or all phones in a Cisco Unified CME system.

speed-dial (voice logout-profile and voice user-profile)

To create speed-dial definitions in a user profile or logout profile to be downloaded by the Extension Mobility feature in Cisco Unified CME, use the **speed-dial** command in voice user-profile or voice logout-profile configuration mode. To disable a speed-dial definition, use the no form of this command.

speed-dial *speed-tag number* [**label** *label*] [**blf**]

no speed-dial *speed-tag*

Syntax Description

speed-tag	Unique sequence number that identifies a speed-dial definition during configuration tasks. Range: 1 to 36.
number	Digits to be dialed when the speed-dial button is pressed on an IP phone or the digits to be dialed when the associated code is entered from an analog phone with an ATA device. For IP phones, if the first character of this string is the plus sign (+), this speed-dial number is locked and cannot be changed at the phone. If the only character in this string is a pound sign (#), a user-programmable speed-dial button with no speed-dial number attached is defined.
label *label*	(Optional) String that contains identifying text to be displayed next to the speed-dial button. Enclose the string in quotation marks if the string contains a space.
blf	(Optional) Enables Busy Lamp Field (BLF) monitoring for a speed-dial number.

Command Default

No speed-dial definition is created.

Command Modes

Voice logout-profile configuration (voice-logout-profile)
Voice user-profile configuration (voice-user-profile)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XW1	Cisco Unified CME 4.2	This command was introduced.

Usage Guidelines

This command in voice user-profile configuration mode creates a speed-dial definition to be downloaded to the IP phone when a user is logged into an IP phone registered in Cisco Unified CME and enabled for extension mobility.

This command in voice logout-profile configuration mode creates a speed-dial definition to be downloaded to the IP phone when no user is logged into an IP phone registered in Cisco Unified CME and enabled for extension mobility.

For button appearance, extension mobility will associate directory numbers then speed-dial definitions in the logout profile or user profile to phone buttons in a sequential manner. If the profile contains more directory and speed-dial numbers than there are buttons on the physical phone to which the profile is downloaded, the remaining numbers in the profile are ignored.

On Cisco Unified IP phones, speed-dial definitions are assigned to available extension buttons that have not been assigned to extensions. Speed-dial definitions are assigned in the order of their identifier (tag) numbers, from 1 to 36.

Examples

The following example shows the configuration for a voice-user profile to be downloaded when the a phone user logs into a Cisco Unified IP phone that is enabled for extension mobility. Which lines and speed-dial buttons in this profile are configured on an IP phone after the user logs in depends on phone type. For example, if the user logs into a Cisco Unified IP Phone 7970, all buttons are configured according to voice-user profile1. However, if the phone user logs into a Cisco Unified IP Phone 7960, all six lines are mapped to phone buttons and the speed dial is ignored because there is no button available for speed dial.

```
pin 12345
user me password pass123
number 2001 type silent-ring
number 2002 type beep-ring
number 2003 type feature-ring
number 2004 type monitor-ring
number 2005,2006 type overlay
number 2007,2008 type cw-overly
speed-dial 1 3001
speed-dial 2 3002 blf
```

Related Commands

Command	Description
logout-profile	Enables Cisco Unified IP phone for extension mobility and assigns a logout profile to this phone.

speed-dial (voice register pool)

To create a speed-dial definition for a SIP phone or analog phone that uses an analog telephone adaptor (ATA) in a Cisco CallManager Express (Cisco CME) system, use the **speed-dial** command in voice register pool configuration mode. To disable a speed-dial definition, use the **no** form of this command.

speed-dial *speed-tag digit-string* [**label** *label-text*]

no speed-dial *speed-tag*

Syntax Description

speed-tag	Unique sequence number that identifies a speed-dial definition during configuration tasks. Range is 1 to 5.
digit-string	Digits to be dialed when the speed-dial button is pressed on an IP phone or the digits to be dialed when the associated code is entered from an analog phone with an ATA device. For IP phones, if the first character of this string is the plus sign (+), this speed-dial number is locked and cannot be changed at the phone. If the only character in this string is a pound sign (#), a user-programmable speed-dial button with no speed-dial number attached is defined.
label *label-text*	(Optional) Text string that appears next to the speed-dial button. Enclose the string in quotation marks if the string contains a space.

Defaults

This command has no arguments or keywords.

Command Modes

Voice register pool configuration

Command History

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

Usage Guidelines

The *speed-tag* argument in this command is a unique identifier for a speed-dial definition on the phone that is being configured. On Cisco IP phones, speed-dial definitions are assigned to available extension buttons that have not been assigned to extensions. Speed-dial definitions are assigned in the order of their identifier numbers.

For example, if you define speed-dial 1, it is assigned to the first phone button that is available after the buttons that are assigned to extensions. If you used two buttons for extensions on a phone, speed-dial 1 is assigned to the third physical button on the phone. When you define speed-dial 2, it is assigned to the fourth physical button on the phone.

For IP phones, speed-dial numbers can be assigned by the administrator by using the *digit-string* argument and can be locked if the *digit-string* argument begins with a plus sign (+). Locked numbers cannot be changed at the phone. Speed-dial instances without speed-dial numbers (those defined with only a pound sign) and speed-dial instances with unlocked *digit-string* arguments can be changed by users at their IP phones.

If more speed-dial definitions are created than are supported by the IP phone setup, the extra speed-dial configurations are ignored.

Changes made to speed-dial buttons are saved in the router's nonvolatile random-access memory (NVRAM) configuration after a timer-based delay.

Analog phone users who use a Cisco ATA-186 or Cisco ATA-188 to connect to Cisco CME systems use a different method to access speed-dial numbers. Instead of pressing a speed-dial button, phone users with ATA devices press the asterisk (star) key and a *speed-tag* number (speed-dial identifier) to dial a speed-dial number. For instance, a phone user with a Cisco ATA-186 presses *1 to dial the number that has been programmed as speed-dial 1 on that phone. Phones with ATA devices are limited to a maximum of nine speed-dial numbers that must be programmed by the system administrator. The numbers cannot be programmed from the phone. With phones that use ATA devices, system administrators must be sure to tell phone users when speed-dial numbers have been programmed for their phones.

After you configure this command, restart the phone by using the **reset** command.

Examples

The following example shows how to set speed-dial button 2 to dial the head office at extension 5001 and lock the setting so that the phone user cannot change it at the phone:

```
Router(config)# voice register pool 23
Router(config-register-pool)# speed-dial 2 +5001 label "Head Office"
```

Related Commands

Command	Description
reset (voice register pool)	Performs a fast reboot of a single IP phone in a Cisco CME system.
reset (voice register global)	Performs a reboot of all SIP phones in a Cisco CME system.
voice register pool	Enters voice register pool configuration mode for SIP phones.

srst dn line-mode

To specify line mode for the ephone-dns that are automatically created in Survivable Remote Site Telephony (SRST) mode on a Cisco Unified CME router, use the **srst dn line-mode** command in telephony-service configuration mode. To return to the default, use the **no** form of this command.

srst dn line-mode {**dual** | **single**}

no srst dn line-mode

Syntax Description

dual	SRST fallback ephone-dns will be dual-line ephone-dns.
single	SRST fallback ephone-dns will be single-line ephone-dns.

Command Default

Single-line mode

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command specifies whether ephone-dns that are created during fallback should be dual-line or single-line ephone-dns. It applies only to the ephone-dns that are "learned" automatically from ephone configuration information, and not to ephone-dns that are manually prebuilt using Cisco Unified CME commands.

Use the **show telephony-service ephone-dn** command to display Cisco Unified CME parameters for ephone-dns.

Examples

The following example specifies dual-line mode for all SRST fallback ephone-dns.

```
telephony-service
 srst dn line-mode dual
```

Related Commands

Command	Description
show telephony-service ephone-dn	Displays parameters for ephone-dns.

srst dn template

To specify an ephone-dn template to be used in Survivable Remote Site Telephony (SRST) mode on a Cisco Unified CallManager Express (Cisco Unified CME) router, use the **srst dn template** command in telephony-service configuration mode. To return to the default, use the **no** form of this command.

srst dn template *template-tag*

no srst dn template

Syntax Description

template-tag	Identifying number of an existing ephone-dn template. Range is from 1 to 15.

Command Default

No ephone-dn template is specified.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command applies the specified ephone-dn template to all SRST fallback ephone-dns. Ephone-dn templates are created with the **ephone-dn-template** command.

Use the **show telephony-service ephone-dn-template** command to display the contents of ephone-dn templates.

Examples

The following example applies ephone-dn template 2 to all SRST fallback ephone-dns.

```
telephony-service
 srst dn template 2
```

Related Commands

Command	Description
ephone-dn-template	Enters ephone-dn-template configuration mode to create an ephone-dn template.
show telephony-service ephone-dn-template	Displays the contents of ephone-dn templates.

srst ephone description

To specify a description to be associated with an ephone in Survivable Remote Site Telephony (SRST) mode on a Cisco Unified CallManager Express (Cisco Unified CME) router, use the **srst ephone description** command in telephony-service configuration mode. To return to the default, use the **no** form of this command.

srst ephone description *string*

no srst ephone description

Syntax Description

string	Description to be associated with an ephone. Maximum string length is 100 characters.

Command Default

No description is specified.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Use the **show telephony-service ephone** command to display the ephone description to be associated with SRST fallback phones.

Examples

The following example applies a description to all SRST fallback ephones.

```
telephony-service
 srst ephone description srst fallback auto-provision phone
```

The following excerpt displays a time-stamped SRST description for ephone 1:

```
Router# show running-config

ephone 1
 description srst fallback auto-provision phone : Jul 07 2005 17:45:08
 ephone-template 5
 mac-address 100A.7052.2AAE
 button 1:1 2:2
```

Related Commands

Command	Description
show telephony-service ephone	Displays ephone settings.

srst ephone template

To specify an ephone template to be used in Survivable Remote Site Telephony (SRST) mode on a Cisco Unified CallManager Express (Cisco Unified CME) router, use the **srst ephone template** command in telephony-service configuration mode. To return to the default, use the **no** form of this command.

srst ephone template *template-tag*

no srst ephone template

Syntax Description

template-tag	Identifying number of an existing ephone template. Range is from 1 to 20.

Command Default

No ephone template is specified.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Ephone templates are created with the **ephone-template** command. This command applies the specified ephone template to all SRST fallback ephones.

Use the **show telephony-service ephone-template** command to display the contents of ephone templates.

Examples

The following example applies ephone template 3 to all SRST fallback ephones.

```
telephony-service
 srst ephone template 3
```

Related Commands

Command	Description
ephone-template	Enters ephone-template configuration mode to create an ephone template.
show telephony-service ephone-template	Displays the contents of ephone templates.

srst mode auto-provision

To enable Survivable Remote Site Telephony (SRST) mode for a Cisco Unified CallManager Express (Cisco Unified CME) router, use the **srst mode auto-provision** command in telephony-service configuration mode. To return to the default, use the **no** form of this command.

srst mode auto-provision {**all** | **dn** | **none**}

no srst mode auto-provision

Syntax Description

all	Includes information for learned ephones and ephone-dns in the running configuration.
dn	Includes information for learned ephone-dns in the running configuration.
none	Does not include information for learned ephones or learned ephone-dns in the running configuration. Use this keyword when you want Cisco Unified CME to provide SRST fallback services for Cisco Unified CallManager.

Command Default

SRST mode is disabled.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command puts a Cisco Unified CME router into SRST mode to provide fallback call-processing services for IP phones that have lost connection to their Cisco Unified CallManager. The phones can be preconfigured manually or the Cisco Unified CME-SRST router can dynamically learn their configuration. The keywords in this command allow you to specify how much of the learned phone configurations you want to include in the running configuration of the Cisco Unified CME-SRST router.

Use the **none** keyword to enable the Cisco Unified CME router to provide SRST fallback services for Cisco Unified CallManager. Use the **dn** or **all** keyword to enable the Cisco Unified CME router to learn the ephone-dn or ephone and ephone-dn configuration from Cisco Unified CallManager and include the information in its running configuration.

Note We do not recommend that you use the **dn** or **all** keyword if you want Cisco Unified CME to provide SRST fallback services. After the Cisco Unified CME-SRST router learns the phone configuration from Cisco Unified CallManager and you save the configuration, the fallback phones are treated as locally configured phones on the Cisco Unified CME-SRST router which can adversely impact the fallback behavior of those phones.

Examples

The following example shows how to enable the Cisco Unified CME router to provide SRST fallback services for phones connected to Cisco Unified CallManager. Information for learned ephone-dns and ephones is not included in the running configuration.

```
telephony-service
 srst mode auto-provision none
```

Related Commands

Command	Description
show telephony-service all	Displays detailed configuration for phones, voice ports, and dial peers in a Cisco Unified CME system.
srst dn line-mode	Specifies line mode for the ephone-dns that are automatically created in SRST mode on a Cisco Unified CME router.

statistics collect

To enable the collection of call statistics for an ephone hunt group, use the **statistics collect** command in ephone-hunt configuration mode. To stop statistics collection and to delete statistics that have been collected, use the **no** form of this command.

statistics collect

no statistics collect

Syntax Description

This command has no arguments or keywords.

Defaults

The default is no call statistics data is collected.

Command Modes

Ephone-hunt configuration

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.3(11)XL	3.2.1	This command was introduced.
12.3(14)T	3.3	This command was integrated into Cisco IOS Release 12.3(14)T.

Usage Guidelines

This command is used for the collection of call statistics, such as direct calls to hunt group pilot numbers, or calls to the Basic Automatic Call Distribution (B-ACD) and Auto Attendant service. For detailed information, see *Cisco Unified CME B-ACD and Tcl Call-Handling Applications*.

The **statistics collect** command can be used to activate statistics collection for any number of ephone hunt groups.

Statistics collection begins at the time that the **statistics collect** command is entered. A maximum of one week (168 hours) of statistics can be stored at a time. You can display the statistics with the **show hunt-group** command or transfer statistics automatically to files using TFTP. The **no statistics collect** command deletes all statistics that have been collected.

All or some of the statistics can be output with the **show hunt-group** command or sent to files automatically using TFTP by using the **hunt-group report url** command **hunt-group report every hours** commands.

Examples

The following example enables the collection of call statistics for ephone hunt group 1 and ephone hunt group 2:

```
Router(config)# ephone-hunt 1
Router(config-ephone-hunt)# statistics collect

Router(config)# ephone-hunt 2
Router(config-ephone-hunt)# statistics collect
```

Related Commands

Command	Description
ephone-hunt	Defines an ephone hunt group and enters ephone-hunt configuration mode.
hunt-group report delay hours	Delays the automatic transfer of Cisco CME B-ACD call statistics to a file.
hunt-group report every hours	Sets the hourly interval at which Cisco CME B-ACD call data is automatically transferred to a file.
hunt-group report url	Sets filename parameters and the URL path where Cisco CME B-ACD call statistics are to be sent using TFTP.
show ephone-hunt statistics	Displays ephone-hunt configuration information and current status and statistics information.

supplementary-service sip

To enable SIP supplementary service capabilities for call forwarding and call transfers across a SIP network, use the **supplementary-service sip** command in dial-peer or voice service voip configuration mode. To disable supplementary service capabilities, use the **no** form of this command.

supplementary-service sip {**moved-temporarily** | **refer**}

no supplementary-service sip {**moved-temporarily** | **refer**}

Syntax Description

moved-temporarily	SIP redirect response for call forwarding.
refer	SIP REFER message for call transfers.

Command Default

SIP supplementary service capabilities are enabled globally.

Command Modes

Dial-peer configuration
Voice-service voip configuration

Command History

Release	Modification
12.4(11)XJ	This command was introduced.
12.4(15)T	This command was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

This command allows you to disable a supplementary service feature (call forwarding or call transfer) if the destination gateway does not support the supplementary service. You can disable the feature either globally or for a specific SIP trunk (dial peer) by using the **no** form of this command.

The **no supplementary-service sip moved-temporarily** command prevents the router from sending a redirect response to the destination for call forwarding. The **no supplementary-service sip refer** command prevents the router from forwarding a REFER message to the destination for call transfers. The router instead attempts to initiate a hairpin call to the new target.

If this command is enabled globally and disabled on a dial peer, the functionality is disabled for the dial peer.

If this command is disabled globally and either enabled or disabled on a dial peer, the functionality is disabled for the dial peer.

This command is supported for calls between SIP phones and for calls between SCCP phones. It is not supported for a mixture of SCCP and SIP phones; for example, it has no effect for calls from a SCCP phone to a SIP phone.

Examples

The following example shows how to disable SIP call transfer capabilities for dial peer 37.

```
Router(config)# dial-peer voice 37 voip
Router(config-dial-peer)# destination-pattern 555....
Router(config-dial-peer)# session target ipv4:10.5.6.7
```

```
Router(config-dial-peer)# no supplementary-service sip refer
```

The following example shows how to disable SIP call forwarding capabilities globally:

```
Router(config)# voice service voip
Router(conf-voi-serv)# no supplementary-service sip moved-temporarily
```

Related Commands

Command	Description
supplementary-service h450.2 (voice-service)	Globally enables H.450.3 capabilities for call transfer.
supplementary-service h450.3 (voice-service)	Globally enables H.450.3 capabilities for call forwarding.

system message

To set a text message for display on idle Cisco IP Phones 7940 and 7940G and Cisco IP Phones 7960 and 7960G in a Cisco CallManager Express (Cisco CME) system, use the **system message** command in telephony-service configuration mode. To return to the default, use the **no** form of this command.

system message *text-message*

no system message

Syntax Description

text-message	Alphanumeric string of approximately 30 characters maximum to display when the phone is idle.

Defaults

The message "Cisco CallManager Express" is displayed.

Command Modes

Telephony-service configuration

Command History

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

Usage Guidelines

The number of characters that can be displayed is not fixed because IP phones typically use a proportional (as opposed to a fixed-width) font. There is room for approximately 30 alphanumeric characters.

The display message is refreshed with a new message after any of the following events occurs:

- A busy phone goes back on-hook.
- An idle phone receives a keepalive message.
- A phone is restarted.

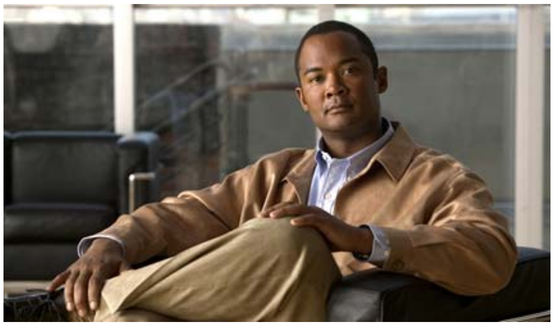
Examples

The following example sets the message "ABC Company" to display instead of "Cisco CallManager Express" on idle the Cisco IP Phones 7940 and 7940G and the Cisco IP Phones 7960 and 7960G:

```
Router(config)# telephony-service
Router(config-telephony)# system message ABC Company
```

Related Commands

Command	Description
telephony-service	Enters telephony-service configuration mode.



Cisco Unified CME Commands: T

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This chapter contains commands to configure and maintain Cisco Unified Communications Manager Express (formally known as Cisco Unified CallManager Express). The commands are presented in alphabetical order. Some commands required for configuring Cisco Unified Communications Manager Express (Cisco Unified CME) may be found in other Cisco IOS command references. Use the command reference master index or search online to find these commands.

telephony-service

To enter telephony-service configuration mode for configuring Cisco Unified CME, use the **telephony-service** command in global configuration mode. To remove the entire Cisco Unified CME configuration for SCCP IP phones, use the **no** form of this command.

telephony-service [**setup**]

no telephony-service

Syntax Description

setup	(Optional) Interactive setup tool for configuring Cisco Unified IP Phone 7910s, 7940s, and 7960s in Cisco Unified CME. **Note** This interactive Cisco CME setup tool is restricted to generating basic configuration files for Cisco Unified IP Phone 7910s, 7940s, and 7960s running SCCP protocol only.

Defaults

No Cisco Unified CME configuration for SCCP IP phones is present.

Command Modes

Global configuration

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.1(5)YD	1.0	This command was introduced on the Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(2)XT	2.0	This command was implemented on the Cisco 1750 and Cisco 1751.
12.2(8)T	2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	2.0	This command was implemented on the Cisco 2600XM and Cisco 2691
12.2(11)T	2.01	This command was implemented on the Cisco 1760.
12.2(15)ZJ	3.0	The **setup** keyword was added.
12.3(4)T	3.0	This command was integrated into Cisco IOS Release 12.3(4)T.

Usage Guidelines

This command enters the telephony-service configuration mode for configuring system wide parameters for SCCP IP phones in Cisco Unified CME

Use the **setup** keyword to start the interactive setup tool to automatically configure only Cisco Unified IP Phone 7910s, 7940s, and 7960s in Cisco Unified CME.

For alternate methods of automatically configuring Cisco Unified CME, including Cisco Unified IP Phone 7910s, 7940s, and 7960s and other Cisco Unified IP phones, see the *Cisco Unified CME Administrator Guide* at http://www.cisco.com/en/US/products/sw/voicesw/ps4625/products_documentation_roadmap09186a0080189132.html.

The **setup** keyword is not stored in the router nonvolatile random-access memory (NVRAM).

If you attempt to use the **setup** option for a system that already has a nonempty telephony-service configuration, the command is rejected. To use the **setup** option after an existing telephony-service configuration has been created, first remove the existing configuration using the **no telephony-service** command.

Table 55 shows a sample dialog with the Cisco CME setup tool and explains possible responses to the Cisco CME setup tool prompts.

Table 55 Cisco CME Setup Tool Dialog Prompts

Cisco CME Setup Tool Prompt	Description

`Do you want to setup DHCP service for your IP phones? [yes/no]:`
 	
If you respond yes, you see the following prompts:
 	
`IP network for telephony-service DHCP Pool:`
`Subnet mask for DHCP network :`
`TFTP Server IP address (Option 150) :`
`Default Router for DHCP Pool :`
	• **Yes**—Configures the Cisco Unified CME router to act as a Dynamic Host Configuration Protocol (DHCP) server, automatically providing IP addresses to your IP phones and provisioning the default gateway and TFTP IP addresses to be used by the phones. This method creates a single pool of IP addresses. If you need a pool for non-IP phones or if the Cisco router cannot act as the DHCP router, answer no and manually define the DHCP server.
	• **No**—Indicates that you have already configured DHCP or static IP addresses for the IP phones.

`Do you want to start telephony-service setup? [yes/no]:`	• **Yes—**Starts the interactive setup tool for configuring Cisco Unified IP Phone 7910s, 7940s, and 7960s. • **No**—Terminates the Cisco CME setup tool.

`Enter the IP source address for Cisco CallManager Express:`
 	
`Enter the Skinny Port for Cisco CallManager Express: [2000]:`
	IP address on which the router provides Cisco Unified CME services, usually the default gateway for the IP subnet that you are using for the IP phones, and the port for Skinny Client Control Protocol (SCCP) messages.

`How many IP phones do you want to configure : [0]:`	Enter the maximum number of IP phones that this Cisco Unified CME system will support. This number can be increased later, to the maximum allowed for this version and your router. **Note** The Cisco CME setup tool associates one number with each newly registering phone. If you want additional numbers on a phone, manually add them later.

Table 55 Cisco CME Setup Tool Dialog Prompts (continued)

<table>
<tr><th>Cisco CME Setup Tool Prompt</th><th>Description</th></tr>
<tr><td><pre>Do you want dual-line extensions
assigned to phones? [yes for dual-line
/ no for single-line]:</pre></td><td>Yes—Each newly registering IP phones is assigned a single number that is associated with a single phone button. The system generates a dual-line ephone-dn entry for each ephone-dn.
No—IP phones are linked directly to one or more PSTN trunk lines. Using keyswitch mode requires manual configuration in addition to using the Cisco CME setup tool. The system generates two ephone-dn entries for each ephone-dn, and they are both assigned to a single phone.</td></tr>
<tr><td><pre>What language do you want on IP phones?
      0 English
      1 French
      2 German
      3 Russian
      4 Spanish
      5 Italian
      6 Dutch
      7 Norwegian
      8 Portuguese
      9 Danish
      10 Swedish
 [0]:</pre></td><td>Language for IP phone displays, selected from the list.
Default is 0, English.</td></tr>
<tr><td><pre>Which Call Progress tone set do you
want on IP phones :
      0 United States
      1 France
      2 Germany
      3 Russia
      4 Spain
      5 Italy
      6 Netherlands
      7 Norway
      8 Portugal
      9 UK
    10 Denmark
    11 Switzerland
    12 Sweden
    13 Austria
    14 Canada
[0]:</pre></td><td>Locale for the tone set used to indicate call status or progress, selected from the list.
Default is 0, United States.</td></tr>
<tr><td><pre>What is the first extension number you
want to configure :[0]:</pre></td><td>First number in pool of extension numbers to be created for IP phones connected to the Cisco router to be configured.
Starting with this number, remaining extension numbers are automatically configured in a contiguous manner.
This number must be compatible with your telephone number plan and if you use Direct Inward Dialing (DID) service, with public switched telephone network (PSTN) numbering requirements.</td></tr>
</table>

Table 55 Cisco CME Setup Tool Dialog Prompts (continued)

Cisco CME Setup Tool Prompt	Description
`Do you have Direct-Inward-Dial service for all your phones? [yes/no]:`	• **Yes**—If you have trunk access to public telephone service by ISDN or VoIP for all extension numbers. The system creates an appropriate dial plan. • **No**—If you have simple analog phone lines only (for example, foreign exchange office [FXO] interfaces) or if you have trunk access for some lines but not all lines.
If you answer yes to the previous question, you see the following prompt: `Enter the full E.164 number for the first phone:`	Complete ten-digit telephone number, including area code, that corresponds to the first extension number.
`Do you want to forward calls to a voice message service? [yes/no]:`	• **Yes**—To forward calls to a single voice message service number when an IP phone is busy or does not answer. All phone extensions forward their calls to the same voice message service pilot number. • **No**—Not to forward calls to a single voice message service number. Answer **no** if you do not have a voice message system or if you want to customize call-forwarding behavior for each extension.
If you answer yes to the previous question, you see the following prompt: `Enter the extension or pilot number of the voice message service:`	Voice message service pilot number. • This step can be ignored during the setup dialog and manually configured later.
`Call forward No Answer Timeout: [18]:`	Timeout, in seconds, after which to forward calls to voice mail if they are not answered. • Default is 18.
`Do you wish to change any of the above information? [yes/no]:`	• **Yes**—Starts the dialog over again without implementing any of the answers that you previously gave. • **No**—Uses specified values to automatically build basic configuration for Cisco Unified IP Phone 7910s, 7940s, and 7960s in Cisco Unified CME.

Examples

The following example shows how to enter telephony-service configuration mode for manually configuring Cisco Unified CME. This example also includes the command for configuring the maximum number of phones to 12:

```
Router(config)# telephony-service
Router(config-telephony)# max-ephones 12
```

The following example shows how to start the Cisco CME setup tool:

```
Router(config)# telephony-service setup
```

template (voice register pool)

To apply a template to a SIP phone, use the **template** command in voice register pool configuration mode. To remove the template, use the **no** form of this command.

template *template-tag*

no template *template-tag*

Syntax Description

template-tag	The template tag that was created with the **voice register template** command in voice register global configuration mode. Range is 1 to 5.

Defaults

No default behavior or values

Command Modes

Voice register pool configuration

Command History

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

Usage Guidelines

Apply any one of five previously defined templates to a SIP phone. Only one template is applied to a SIP phone at one time.

Examples

The following example shows how to define templates 1 and 2 and apply template 1 to SIP phones 1, 2, and 3, and template 2 to SIP phone 4:

```
Router(config)#  voice register template 1
Router(config-register-temp)# anonymous block
Router(config-register-temp)# caller-id block
Router(config-register-temp)# voicemail 5001 timeout 15

Router(config)# voice register template 2
Router(config-register-temp)# anonymous block
Router(config-register-temp)# caller-id block
Router(config-register-temp)# no conference
Router(config-register-temp)# no transfer-attended
Router(config-register-temp)# voicemail 5005 timeout 15

Router(config)# voice register pool 1
Router(config-register-pool)# template 1

Router(config)# voice register pool 2
Router(config-register-pool)# template 1

Router(config)# voice register pool 3
Router(config-register-pool)# template 1

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

```
Router(config-register-pool)# template 2
```

Related Commands

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

tftp-path (voice register global)

To specify the directory to which the configuring files for SIP phones in Cisco Unified CME are written, use the **tftp-path** command in voice register global configuration mode. To return to the default, use the **no** form of this command.

tftp-path {**system:** | **flash:** | **slot0:** | **tftp://***URL*}

no tftp-path

Syntax Description

flash:	Router flash memory.
slot0:	Router slot 0 memory.
tftp://	External TFTP server.
URL	URL for external TFTP server.
system:	System memory (system:/cme/sipphone/).

Defaults

Default is system memory.

Command Modes

Voice register global configuration

Command History

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

Usage Guidelines

Use this command to define the location of the provisioning files that are generated by using the **create profile** command.

Examples

The following example shows how to set the path to an HTTP directory for an external TFTP server:

```
Router(config)# voice register global
Router(config-register-global)# tftp-path tftp://mycompany.com/files/
```

Related Commands

Command	Description
create profile (voice register global)	Generates the configuration profiles required for SIP phones.

tftp-server-credentials trustpoint

To specify the PKI trustpoint that signs the phone configuration files, use the **tftp-server-credentials trustpoint** command in telephony-service configuration mode. To return to the default, use the **no** form of this command.

tftp-server-credentials trustpoint *label*

no tftp-server-credentials trustpoint

Syntax Description

label	Name of a configured PKI trustpoint with a valid certificate.

Command Default

No trustpoint is defined for TFTP server communications.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used with Cisco Unified CME phone authentication.

Examples

The following example names the CA trustpoint, server12, as the trustpoint that signs the phone configuration files.

```
Router(config)# telephony-service
Router(config-telephony)# device-security-mode authenticated
Router(config-telephony)# secure-signaling trustpoint server25
Router(config-telephony)# tftp-server-credentials trustpoint server12
Router(config-telephony)# load-cfg-file slot0:Ringlist.xml alias Ringlist.xml sign create
Router(config-telephony)# exit
```

time-format

To select a 12-hour clock or a 24-hour clock for the time display format on Cisco IP phones in a Cisco CallManager Express (Cisco CME) system, use the **time-format** command in telephony-service configuration mode. To return to the default, use the **no** form of this command.

time-format {**12** | **24**}

no time-format

Syntax Description

12	Selects a 12-hour clock. This is the default.
24	Selects a 24-hour clock.

Defaults

12

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.2(2)XT	2.0	This command was introduced on the following platforms: Cisco 1750, Cisco 1751, Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(8)T	2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	2.0	This command was implemented on the Cisco 2600XM and Cisco 2691.
12.2(11)T	2.01	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760.

Examples

The following example selects a 24-hour clock for the time display on Cisco IP phones:

```
Router(config)# telephony-service
Router(config-telephony)# time-format 24
```

Related Commands

Command	Description
date-format	Selects a format to display the date on Cisco IP phones.
telephony-service	Enters telephony-service configuration mode.

time-format (voice register global)

To set the time display format on SIP phones in a Cisco Unified CME system, use the **time-format** command in voice register global configuration mode. To display the date in the default format, use the **no** form of this command.

time-format {**12** | **24**}

no date-format

Syntax Description

12	Sets time in a 12-hour (AM/PM) clock.
24	Sets time in a 24-hour clock.

Defaults

12-hour clock

Command Modes

Voice register global configuration

Command History

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

Examples

The following example shows how to set the time format to a 24-hour clock so that 11:00PM is displayed as 2300.

```
Router(config)# voice register global
Router(config-register-global)# time-format 24
```

Related Commands

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

timeout (ephone-hunt)

To define the number of seconds after which a call that is not answered is redirected to the next number in a hunt-group list in Cisco Unified CME, use the **timeout** command in ephone-hunt configuration mode. To return to the default, use the **no** form of this command.

timeout *seconds*[**,** *seconds*...]

no timeout *seconds*[**,** *seconds*...]

Syntax Description

seconds	Number of seconds. Range: 3 to 60000. You can enter a different value for each hop between ephone-dns in a hunt group. If you enter a single value, the value is applied to each hop between ephone-dns in a hunt group.

Command Default

The time period set by the **timeouts ringing** command, which has a default of 180 seconds if it is not set to another value.

Command Modes

Ephone-hunt configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.4(4)XC	Cisco Unified CME4.0	This command was modified to accept multiple arguments that correspond to the number of ephone-dns configured in the hunt group.
12.4(9)T	Cisco Unified CME 4.0	This command modified to accept multiple arguments that correspond to the number of ephone-dns configured in the hunt group was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Use this command to set no-answer timeouts for each hop in a hunt group. You can enter a different value for each hop between ephone-dns in a hunt group or you enter a single value to be applied to each hop between ephone-dns in a hunt group list.

If you configure this command and you also configure the **max-timeout** command for an ephone hunt group, the max-timeout command takes precedence over this command.

Examples

The following example defines a no-answer timeout of 10 seconds for each hop between ephone-dns in hunt group 25. If extension 1001 does not answer in 10 seconds, the call is sent to 1002. If 1002 does not answer in 10 seconds, the call is sent to 1003. If 1003 does not answer in 10 seconds, the call is sent to the final number.

```
ephone-hunt 25 sequential
```

```
 pilot 4200
 list 1001, 1002, 1003
 timeout 10
 final 4500
```

The following example shows a hunt-group configuration with separate timeouts, one for each ephone in the hunt-group. If the first extension (1001) does not answer in 7 seconds, the call is sent to the second extension (1002). If the call is not answered by the second extension in 9 seconds, the call is forwarded to the third extension (1003). Extension 1003 has 15 seconds to answer before the call is sent to the final number.

```
ephone-hunt 3 peer
 pilot 4200
 list 1001, 1002, 1003
 timeout 7, 9, 15
 final 4500
```

The following example shows the configuration for an ephone hunt group for which the max-timeout command is also configured. using this configuration, if the second number is busy, the third extension, 1003, has only 13 seconds to answer (20 - 7 = 13) because the value for max-timeout command is 20 seconds.

```
ephone-hunt 3 peer
 pilot 4200
 list 1001, 1002, 1003
 timeout 7, 9, 15
 max-timeout 20
 final 4500
```

Related Commands

Command	Description
final	Defines the last ephone-dn in an ephone hunt group.
hops	Defines the number of times that a call is redirected to the next ephone-dn in a peer ephone-hunt-group list before proceeding to the final ephone-dn.
list	Defines the ephone-dns that participate in an ephone hunt group.
max-redirect	Changes the current number of allowable redirects in a Cisco Unified CME system.
max-timeout	Sets the maximum combined timeout for the no-answer periods for all ephone-dns in an ephone-hunt list,
no-reg (ephone-hunt)	Specifies that the pilot number of an ephone hunt group should not register with the H.323 gatekeeper.
pilot	Defines the ephone-dn that callers dial to reach an ephone hunt group.
preference (ephone-hunt)	Sets preference order for the ephone-dn associated with an ephone-hunt-group pilot number.

timeout (voice hunt-group)

To define the number of seconds after which a call that is not answered is redirected to the next number in a hunt-group list, use the **timeout** command in voice hunt-group configuration mode. To return to the default timeout, use the **no** form of this command.

timeout *seconds*

no timeout *seconds*

Syntax Description

seconds	Number of seconds. Range is 3 to 60000. Default is 180.

Defaults

180 seconds

Command Modes

Voice hunt-group configuration

Command History

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

Examples

The following example shows how to define a no-answer timeout of 15 seconds to be applied to each hop between phones in peer hunt-group 25:

```
Router(config)# voice hunt-group 25 peer
Router(config-voice-hunt-group)# timeout 15
```

Related Commands

Command	Description
final (voice hunt-group)	Defines the last extension in a voice hunt group.
hops (voice hunt-group)	Defines the number of times that a call is redirected to the next directory number in a peer voice hunt-group list before proceeding to the final directory number.
list (voice hunt-group)	Defines the directory numbers that participate in a hunt group.

timeouts busy

To set the amount of time after which a call is disconnected from a busy signal, use the **timeouts busy** command in telephony-service configuration mode. To return to the default value, use the **no** form of this command.

timeouts busy *seconds*

no timeouts busy

Syntax Description

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

Defaults

10 seconds

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.2(8)T	2.0	This command was introduced.

Examples

The following example sets a busy timeout of 10 seconds:

```
Router(config)# telephony-service
Router(config-telephony)# timeouts busy 10
```

Related Commands

Command	Description
telephony-service	Enters telephony-service configuration mode.

timeouts interdigit (telephony-service)

To set the interdigit timeout value for all Cisco IP phones in a Cisco CallManager Express (Cisco CME) system, use the **timeouts interdigit** command in telephony-service configuration mode. To return to the default value, use the **no** form of this command.

timeouts interdigit *seconds*

no timeouts interdigit

Syntax Description

seconds	Interdigit timeout duration for Cisco IP phones, in seconds. Range is from 2 to 120. Default is 10.

Defaults

10 seconds

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.2(2)XB	1.0	This command was introduced on the Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(2)XT	2.0	This command was implemented on the Cisco 1750 and Cisco 1751.
12.2(8)T	2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	2.0	This command was implemented on the Cisco 2600XM and Cisco 2691.
12.2(11)T	2.01	This command was implemented on the Cisco 1760.

Usage Guidelines

The interdigit timeout timer is activated when the caller enters a digit and is restarted each time the caller enters subsequent digits until the destination address is identified. This command specifies how long, in seconds, the system waits after a caller enters an initial digit or a subsequent digit of a dialed string. If the configured timeout value is exceeded before the destination address is identified, a tone sounds and the call is terminated. The default is 10 seconds.

To disable the timeouts interdigit timer, set the *seconds* value to zero.

Examples

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

```
Router(config)# telephony-service
Router(config-telephony)# timeouts interdigit 5
```

In this example, 5 seconds is also the elapsed time after which an incompletely dialed number times out. For example, if you dial nine digits (408555013) instead of the required ten digits (4085550134), you hear a busy tone after 5 "timeout" seconds.

Related Commands

Command	Description
telephony-service	Enters telephony-service configuration mode.
timeouts interdigit (voice-port)	Configures the interdigit timeout value for a specified voice port.

timeouts ringing (telephony-service)

To set the timeout value for ringing in a Cisco CallManager Express (Cisco CME) system, use the **timeouts ringing** command in telephony-service configuration mode. To reset the timeout value to the default value, use the **no** form of this command.

timeouts ringing *seconds*

no timeouts ringing

Syntax Description

seconds	Duration, in seconds, for which the Cisco CME system allows ringing to continue if a call is not answered. Range is from 5 to 60000. Default is 180.

Defaults

180 seconds

Command Modes

Telephony-service configuration

Command History

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

Examples

The following example allows incoming calls to ring for 600 seconds:

```
Router(config)# telephony-service
Router(config-telephony)# timeouts ringing 600
```

Related Commands

Command	Description
telephony-service	Enters telephony-service configuration mode.

time-webedit (telephony-service)

To enable the system administrator to set time on the Cisco CallManager Express (Cisco CME) router through the web interface, use the **time-webedit** command in telephony-service configuration mode. To disable this feature, use the **no** form of this command.

time-webedit

no time-webedit

Syntax Description

This command has no arguments or keywords.

Defaults

Time-setting through the web interface is disabled.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.2(2)XT	2.0	This command was introduced on the Cisco 1750, Cisco 1751, Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(8)T	2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	2.0	This command was implemented on the Cisco 2600XM and Cisco 2691.
12.2(11)T	2.01	This command was implemented on the Cisco 1760.

Usage Guidelines

The **time-webedit** command allows a local administrator of the Cisco CME router to change and set time through the web-based graphical user interface (GUI).

Note Cisco discourages this method for setting network time. The router should be set up to automatically synchronize its router clock from a network-based clock source using Network Time Protocol (NTP). In the rare case that a network NTP clock source is not available, the **time-webedit** command can be used to allow manual setting and resetting of the router clock through the Cisco CME GUI.

Examples

The following example enables web editing of time:

```
Router(config)# telephony-service
Router(config-telephony)# time-webedit
```

Related Commands

Command	Description
dn-webedit	Enables adding of directory numbers through a web interface.
telephony-service	Enters telephony-service configuration mode.

time-zone

To set Cisco IP Phone 7970G time displays to the correct time zone, use the **time-zone** command in telephony-service configuration mode. To disable a time-zone setting configured with the **time-zone** command and return to the default time zone (Pacific Standard Time), use the **no** form of this command.

time-zone *number*

no time-zone

Syntax Description

number	Numeric code for a named time zone. The following are the selections. The numbers in parentheses indicate the offset from Coordinated Universal Time (UTC) in minutes. • **1**—Dateline Standard Time (-720) • **2**—Samoa Standard Time (-660) • **3**—Hawaiian Standard Time (-600) • **4**—Alaskan Standard/Daylight Time (-540) • **5**—Pacific Standard/Daylight Time (-480) • **6**—Mountain Standard/Daylight Time (-420) • **7**—United States (US) Mountain Standard Time (-420) • **8**—Central Standard/Daylight Time (-360) • **9**—Mexico Standard/Daylight Time (-360) • **10**—Canada Central Standard Time (-360) • **11**—SA Pacific Standard Time (-300) • **12**—Eastern Standard/Daylight Time (-300) • **13**—US Eastern Standard Time (-300) • **14**—Atlantic Standard/Daylight Time (-240) • **15**—South America (SA) Western Standard Time (-240) • **16**—Newfoundland Standard/Daylight Time (-210) • **17**—SA Standard/Daylight Time (-180) • **18**—SA Eastern Standard Time (-180) • **19**—Mid-Atlantic Standard/Daylight Time (-120) • **20**—Azores Standard/Daylight Time (-60) • **21**—UTC Standard/Daylight Time (+0) • **22**—Greenwich Standard Time (+0) • **23**—Western Europe Standard/Daylight Time (+60) • **24**—GTB (Athens, Istanbul, Minsk) Standard/Daylight Time (+60) • **25**—Egypt Standard/Daylight Time (+60) • **26**—Eastern Europe Standard/Daylight Time (+60)

Syntax Description		
	number continued	• **27**—Romance Standard/Daylight Time (+120) • **28**—Central Europe Standard/Daylight Time (+120) • **29**—South Africa Standard Time (+120) • **30**—Jerusalem Standard/Daylight Time (+120) • **31**—Saudi Arabia Standard Time (+180) • **32**—Russian Standard/Daylight Time (+180) • **33**—Iran Standard/Daylight Time (+210) • **34**—Caucasus Standard/Daylight Time (+240) • **35**—Arabian Standard Time (+240) • **36**—Afghanistan Standard Time (+270) • **37**—West Asia Standard Time (+300) • **38**—Ekaterinburg Standard Time (+300) • **39**—India Standard Time (+330) • **40**—Central Asia Standard Time (+360) • **41**—Southeast Asia Standard Time (+420) • **42**—China Standard/Daylight Time (+480) • **43**—Taipei Standard Time (+480) • **44**—Tokyo Standard Time (+540) • **45**—Central Australia Standard/Daylight Time (+570) • **46**—Australia Central Standard Time (+570) • **47**—East Australia Standard Time (+600) • **48**—Australia Eastern Standard/Daylight Time (+600) • **49**—West Pacific Standard Time (+600) • **50**—Tasmania Standard/Daylight Time (+600) • **51**—Central Pacific Standard Time (+660) • **52**—Fiji Standard Time (+720) • **53**—New Zealand Standard/Daylight Time (+720)

Defaults

The default is time-zone 5, Pacific Standard/Daylight Time (-480).

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.3(11)XL	3.2.1	This command was introduced.
12.3(14)T	3.3	This command was integrated into Cisco IOS Release 12.3(14)T.

Usage Guidelines

The clocks in the Cisco IP Phone 7970G units obtain Coordinated Universal Time (UTC) from their Cisco CME router's clocks. To display the correct local time, nearly all Cisco IP Phone 7970G units' time display must be offset with the **time-zone** command.

The **time-zone** command works with the vendorConfig section of the Sep*.conf.xml configuration file, which is read by the phone firmware when a Cisco IP Phone 7970G is booted up. For changes to the time-zone settings take effect, the Sep*.conf.xml file must be updated with the **create cnf-files** command and the Cisco IP Phone 7970G units must rebooted with the **reset** command.

Examples

The following example sets the Cisco IP Phone 7970 units to Fiji Standard Time:

```
Router(config)# telephony-service
Router(config-telephony)# time-zone 53
```

Related Commands

Command	Description
create cnf-files	Sets display and phone functionality for the Cisco IP Phone 7970 units using the vendorConfig parameters of the downloaded firmware's Sep*.conf.xml configuration file.
reset (telephony-service)	Performs a complete reboot of one or all phones associated with a Cisco CME router.

timezone (voice register global)

To set the time zone used for SIP phones in a Cisco Unified CME system, use the **timezone** command in voice register global configuration mode. To return to the default, use the **no** form of this command.

timezone *number*

no timezone

Syntax Description

number	Range is 1 to 53. Default is 5, Pacific Standard/Daylight Time

Defaults

5, Pacific Standard/Daylight Time

Command Modes

Voice register global configuration

Command History

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

Usage Guidelines

Table 56 lists the supported time zone numbers and the corresponding description.

Table 56 Time Zones

Number	Description	Offset in Minutes
1	Dateline Standard Time	-720
2	Samoa Standard Time	-660
3	Hawaiian Standard Time	-600
4	Alaskan Standard/Daylight Time	-540
5	Pacific Standard/Daylight Time	-480
6	Mountain Standard/Daylight Time	-420
7	US Mountain Standard Time	-420
8	Central Standard/Daylight Time	-360
9	Mexico Standard/Daylight Time	-360
10	Canada Central Standard Time	-360
11	SA Pacific Standard Time	-300
12	Eastern Standard/Daylight Time	-300
13	US Eastern Standard Time	-300
14	Atlantic Standard/Daylight Time	-240
15	SA Western Standard Time	-240

Table 56 Time Zones (continued)

Number	Description	Offset in Minutes
16	Newfoundland Standard/Daylight Time	-210
17	South America Standard/Daylight Time	-180
18	SA Eastern Standard Time	-180
19	Mid-Atlantic Standard/Daylight Time	-120
20	Azores Standard/Daylight Time	-60
21	GMT Standard/Daylight Time	+0
22	Greenwich Standard Time	+0
23	W. Europe Standard/Daylight Time	+60
24	GTB Standard/Daylight Time	+60
25	Egypt Standard/Daylight Time	+60
26	E. Europe Standard/Daylight Time	+60
27	Romance Standard/Daylight Time	+120
28	Central Europe Standard/Daylight Time	+120
29	South Africa Standard Time	+120
30	Jerusalem Standard/Daylight Time	+120
31	Saudi Arabia Standard Time	+180
32	Russian Standard/Daylight Time	+180
33	Iran Standard/Daylight Time	+210
34	Caucasus Standard/Daylight Time	+240
35	Arabian Standard Time	+240
36	Afghanistan Standard Time	+270
37	West Asia Standard Time	+300
38	Ekaterinburg Standard Time	+300
39	India Standard Time	+330
40	Central Asia Standard Time	+360
41	SE Asia Standard Time	+420
42	China Standard/Daylight Time	+480
43	Taipei Standard Time	+480
44	Tokyo Standard Time	+540
45	Cen. Australia Standard/Daylight Time	+570
46	AUS Central Standard Time	+570
47	E. Australia Standard Time	+600
48	AUS Eastern Standard/Daylight Time	+600
49	West Pacific Standard Time	+600
50	Tasmania Standard/Daylight Time	+600
51	Central Pacific Standard Time	+660

Table 56 Time Zones (continued)

Number	Description	Offset in Minutes
16	Newfoundland Standard/Daylight Time	-210
17	South America Standard/Daylight Time	-180
18	SA Eastern Standard Time	-180
19	Mid-Atlantic Standard/Daylight Time	-120
20	Azores Standard/Daylight Time	-60
21	GMT Standard/Daylight Time	+0
22	Greenwich Standard Time	+0
23	W. Europe Standard/Daylight Time	+60
24	GTB Standard/Daylight Time	+60
25	Egypt Standard/Daylight Time	+60
26	E. Europe Standard/Daylight Time	+60
27	Romance Standard/Daylight Time	+120
28	Central Europe Standard/Daylight Time	+120
29	South Africa Standard Time	+120
30	Jerusalem Standard/Daylight Time	+120
31	Saudi Arabia Standard Time	+180
32	Russian Standard/Daylight Time	+180
33	Iran Standard/Daylight Time	+210
34	Caucasus Standard/Daylight Time	+240
35	Arabian Standard Time	+240
36	Afghanistan Standard Time	+270
37	West Asia Standard Time	+300
38	Ekaterinburg Standard Time	+300
39	India Standard Time	+330
40	Central Asia Standard Time	+360
41	SE Asia Standard Time	+420
42	China Standard/Daylight Time	+480
43	Taipei Standard Time	+480
44	Tokyo Standard Time	+540
45	Cen. Australia Standard/Daylight Time	+570
46	AUS Central Standard Time	+570
47	E. Australia Standard Time	+600
48	AUS Eastern Standard/Daylight Time	+600
49	West Pacific Standard Time	+600
50	Tasmania Standard/Daylight Time	+600
51	Central Pacific Standard Time	+660

Table 56 Time Zones (continued)

Number	Description	Offset in Minutes
52	Fiji Standard Time	+720
53	New Zealand Standard/Daylight Time	+720

Examples

The following example shows how top set the time zone to 8, Central Standard Daylight Time:

```
Router(config)# voice register global
Router(config-register-global)# timezone 8
```

Related Commands

Command	Description
dst (voice register global)	Sets the time period for daylight saving time on SIP phones.
dst auto-adjust (voice register global)	Enables automatic adjustment of daylight saving time on SIP phones.
time-format (voice register global)	Selects a 12-hour clock or a 24-hour clock for the time display format on SIP phones in a Cisco CME system

transfer max-length

To prevent a phone user from dialing more than the specified number of digits when transferring calls, use the **transfer max-length** command in ephone-template configuration mode. To return to the default, use the **no** form of this command.

transfer max-length *digit-length*

no transfer max-length

Syntax Description

digit-length	Maximum number of digits that can be dialed when transferring a call. Range is from 3 to 16.

Command Default

Phone users can dial a maximum of 16 digits when transferring calls.

Command Modes

Ephone-template configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command can be used to eliminate toll charges on call transfers by limiting the number of digits that phone users can dial when transferring calls. For example, if you specify a maximum of eight digits, that allows users to dial a single external access digit and a seven-digit local number. In most locations this plan will limit transfers to non-toll destinations. Long-distance calls that typically require ten digits or more will not be allowed.

Examples

The following example limits transfers from ephone 6, extension 2977, to numbers of 8 digits or less.

```
ephone-template 2
 transfer max-length 8

ephone-dn 4
 number 2977

ephone 6
 button 1:4
 ephone-template 2
```

transfer-attended (voice register template)

To enable a soft key for attended transfer in a SIP phone template, use the **transfer-attended** command in voice register template configuration mode. To disable the soft key, use the **no** form of this command.

transfer-attended

no transfer-attended

Syntax Description

This command has no arguments or keywords.

Defaults

Enabled

Command Modes

Voice register template configuration

Command History

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

Usage Guidelines

This command enables a soft key for attended transfer in the specified template which can then be applied to SIP phones in Cisco CME. The attended transfer soft key is enabled by default. To disable the attended transfer soft key, use the **no transfer-attended** command. To apply the template to a SIP phone, use the **template** command in voice register pool configuration mode.

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

Examples

The following example shows how to disable attended transfer in template 1:

```
Router(config)# voice register template 1
Router(config-register-temp)# no transfer-attended
```

Related Commands

Command	Description
conference (voice register template)	Enables the soft key for conference in a SIP phone template.
template	Applies a template to a SIP phone.
transfer-blind (voice register template)	Enables a soft key for blind transfer in a SIP phone template.

transfer-blind (voice register template)

To enable a soft key for blind transfer in a SIP phone template, use the **transfer-blind** command in voice register template configuration mode. To disable the soft key, use the **no** form of this command.

transfer-blind

no transfer-blind

Syntax Description

This command has no arguments or keywords.

Defaults

Enabled

Command Modes

Voice register template configuration

Command History

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

Usage Guidelines

This command enables a soft key for blind transfer in the specified template which can then be applied to SIP phones in Cisco CME. The blind transfer soft key is enabled by default. To disable the blind transfer soft key, use the **no transfer-blind** command. To apply the template to a SIP phone, use the **template** command in voice register pool configuration mode.

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

Examples

The following example shows how to disable blind transfer in template 1:

```
Router(config)# voice register template 1
Router(config-register-temp)# no transfer-blind
```

Related Commands

Command	Description
conference (voice register template)	Enables the soft key for conference in a SIP phone template.
template	Applies a template to a SIP phone.
transfer-attended (voice register template)	Enables the soft key for attended transfer on SIP phones.

transfer-mode

To specify the type of call transfer for an individual IP phone extension that uses the ITU-T H.450.2 standard, use the **transfer-mode** command in ephone-dn configuration mode. To remove this specification, use the **no** form of this command.

transfer-mode {**blind** | **consult**}

no transfer-mode

Syntax Description

blind	Transfers calls without consultation using a single phone line.
consult	Transfers calls with consultation using a second phone line, if available.

Command Default

The ephone-dn uses the transfer-system value that was set systemwide.

Command Modes

Ephone-dn configuration

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.2(11)YT	2.1	This command was introduced.
12.2(15)T	2.1	This command was integrated into Cisco IOS Release 12.2(15)T.

Usage Guidelines

This command specifies the type of call transfer for an individual Cisco IP phone extension that is using the ITU-T H.450.2 protocol. It allows you to override the system default **transfer-system** setting (full-consult or full-blind) for that extension.

Call transfers that use H.450.2 can be blind or consultative. A blind transfer is one in which the transferring phone connects the caller to a destination extension before ringback begins. A consultative transfer is one in which the transferring party either connects the caller to a ringing phone (ringback heard) or speaks with the third party before connecting the caller to the third party.

You can specify blind or consultative transfer on a systemwide basis by using the **transfer-system** command. The systemwide setting can then be overridden for individual phone extensions by using the **transfer-mode** command. For example, in a Cisco CallManager Express (Cisco CME) network that is set up for consultative transfer, a specific extension with an auto-attendant that automatically transfers incoming calls to specific extension numbers can be set to use blind transfer, because auto-attendants do not use consultative transfer.

Use this command with Cisco IOS Telephony Services V2.1, Cisco CallManager Express 3.0, or a later version.

Examples

The following example sets blind mode for call transfers from this directory number:

```
Router(config)# ephone-dn 21354
Router(config-ephone-dn)# transfer-mode blind
```

Related Commands

Command	Description
ephone-dn	Enters ephone-dn configuration mode to set directory numbers and parameters for individual Cisco IP phone lines.
transfer-system	Specifies the call transfer method for all IP phones on a Cisco ITS router using the ITU-T H.450.2 standard.

transfer-park blocked

To prevent extensions on an ephone from parking calls, use the **transfer-park blocked** command in ephone or ephone-template configuration mode. To return to the default, use the **no** form of this command.

transfer-park blocked

no transfer-park blocked

Syntax Description

This command has no arguments or keywords.

Command Default

Transfer to park is allowed.

Command Modes

Ephone configuration
Ephone-template configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command prevents transfers to park that use the Trnsfer soft key and a call-park slot number, while allowing call-parks that use only the Park soft key. To prevent use of the Park soft key, use an ephone template to remove it from the phone.

An exception to this command is made for phones with dedicated park slots. If the **transfer-park blocked** command is used on an ephone that has a dedicated park slot, the phone is blocked from parking calls at park slots other than the dedicated park slot, but is still able to park calls at its own dedicated park slot. On an IP phone, the user presses the Trnsfer soft key and the call-park feature access code (FAC) to park a call at the phone's dedicated park slot. On an analog phone, the user presses hookflash and the call-park FAC.

When the **transfer-park blocked** command is used on an ephone that does not have a dedicated park slot, the phone is blocked from parking any calls.

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

Examples

The following example prevents ephone 25 and extensions 234, 235, and 236 from parking calls at any call-park slot.

```
ephone-dn 11
 number 234

ephone-dn 12
 number 235

ephone-dn 13
 number 236

ephone 25
 button 1:11 2:12 3:13
 transfer-park blocked
```

The following example uses an ephone template to prevent ephone 26 and extension 76589 from parking calls at any call-park slot.

```
ephone-dn 33
 number 76589

ephone-template 1
 transfer-park blocked

ephone 26
 button 1:33
 ephone-template 1
```

The following example sets up a dedicated park slot for the extensions on ephone 6 and blocks transfers to call park from extensions 2977, 2978, and 2979 on that phone. Those extensions can still park calls at the phone's dedicated park slot by using the Park soft key or Trnsfer and the call-park FAC.

```
ephone-dn 3
 number 2558
 name Park 2977
 park-slot reserved-for 2977 timeout 60 limit 3 recall alternate 3754

ephone-dn 4
 number 2977

ephone-dn 5
 number 2978

ephone-dn 6
 number 2979

ephone 6
 button 1:4 2:5 3:6
 transfer-park blocked
```

transfer-pattern (telephony-service)

To allow transfer of telephone calls from Cisco IP phones to phones other than Cisco IP phones, use the **transfer-pattern** command in telephony-service configuration mode. To disable these transfers, use the **no** form of this command.

transfer-pattern *transfer-pattern* [**blind**]

no transfer-pattern

Syntax Description

transfer-pattern	String of digits for permitted call transfers. Wildcards are allowed. A maximum of 32 transfer patterns can be entered, using a separate command for each one.
blind	(Optional) When H.450.2 consultative call transfer is used, this keyword forces transfers that match the pattern to be executed as blind transfers. Overrides settings made using the **transfer-system** and **transfer-mode** commands.

Defaults

Transfer of calls is enabled only to local Cisco IP phones.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.1(5)YD	1.0	This command was introduced on the Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(2)XT	2.0	This command was implemented on the Cisco 1750 and Cisco 1751.
12.2(8)T	2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	2.0	This command was implemented on the Cisco 2600XM and Cisco 2691.
12.2(11)T	2.01	This command was implemented on the Cisco 1760.
12.2(15)T	2.1	The **blind** keyword was added.

Usage Guidelines

This command allows you to transfer calls to "other" phones—that is, to non-IP phones and phones outside of your network. A call is then established between the transferred party and the new recipient. By default, all Cisco IP phone extension numbers are allowed as transfer targets.

The **blind** keyword is valid only for systems that use Cisco IOS Telephony Services V2.1, Cisco CallManager Express 3.0, or a later version and applies only to consultative transfers made using the H.450.2 standard. The **blind** keyword forces calls that are transferred to numbers that match the transfer pattern to be executed as blind or full-blind transfers, overriding any settings made using the **transfer-system** and **transfer-mode** commands.

When defining transfers to non-local numbers, it is important to note that transfer-pattern digit matching is performed before translation-rule operations. Therefore, you should specify in this command the digits actually entered by phone users before they are translated. For more information, see the "Translation Rules and Profiles" section in the *Cisco Unified CallManager Express System Administrator Guide* at http://www.cisco.com/en/US/products/sw/voicesw/ps4625/products_documentation_roadmap09186a0080189132.html.

Use of the .**T** control character for the *transfer-pattern* argument is not recommended. The .**T** control character indicates a variable-length dial string, which causes Cisco CME to wait for an interdigit timeout (default is 10 seconds) before transferring a call. To avoid the interdigit timeout, a matching transfer pattern should be used with the **transfer-pattern** command. For example, use the **transfer-pattern 9.........** command instead of the **transfer-pattern .T** command.

Examples

The following example sets a transfer pattern. A maximum of 32 transfer patterns can be entered. In this example, 55501.. (the two periods are wildcards) permits transfers to any number in the range from 555-0100 to 555-0199.

```
Router(config)# telephony-service
Router(config-telephony)# transfer-pattern 55501..
```

Related Commands

Command	Description
transfer-mode	Specifies the type of call transfer for an individual IP phone extension number that uses the ITU-T H.450.2 standard.
transfer-system	Specifies the call transfer method for all Cisco CME extensions that use the ITU-T H.450.2 standard.

transfer-pattern blocked

To prevent extensions on an ephone from transferring calls to patterns defined using the **transfer-pattern (telephony-service)** command, use the **transfer-pattern blocked** command in ephone or ephone-template configuration mode. To return to the default, use the **no** form of this command.

transfer-pattern blocked

no transfer-pattern blocked

Syntax Description

This command has no arguments or keywords.

Command Default

Call transfer is allowed to patterns specified in the **transfer-pattern (telephony-service)** command.

Command Modes

Ephone configuration
Ephone-template configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4.(9)T.

Usage Guidelines

By default, transfers to all numbers except for those on local ephones are automatically blocked. During configuration, you can allow transfers to non-local numbers using the **transfer-pattern (telephony-service)** command.

Use the **transfer-pattern blocked** command to prevent individual phones from transferring calls to non-local numbers that have been globally enabled for transfer.

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

Examples

The following example blocks transfers from ephone 6 (extension 2977) to the PSTN.

```
ephone-dn 4
 number 2977

ephone 6
 button 1:4
 transfer-pattern blocked
```

The following example uses an ephone template to block transfers from ephone 6 (extension 2977) and ephone 7 (extension 2978) to the PSTN.

```
ephone-template 2
 transfer-pattern blocked
```

```
ephone-dn 4
 number 2977
ephone-dn 5
 number 2978

ephone 6
 button 1:4
 ephone-template 2

ephone 7
 button 1:5
 ephone-template 2
```

Related Commands

Command	Description
transfer-pattern (telephony service)	Allows call transfers to numbers outside the Cisco Unified CME system.

transfer-system

To specify the call transfer method to be used by Cisco Unified IP phones in Cisco Unified CME, use the **transfer-system** command in telephony-service configuration mode. To disable the call transfer method, use the **no** form of this command.

transfer-system {**blind** | **full-blind** | **full-consult** [**dss**] | **local-consult**}

no transfer-system

Syntax Description

blind	Transfers calls without consultation using a single phone line and the Cisco proprietary method. This is the default for Cisco CME 3.4 and earlier versions.
full-blind	Transfers calls without consultation using H.450.2 standard methods.
full-consult	Transfers calls using H.450.2 with consultation using a second phone line, if available. The calls fall back to **full-blind** if a second line is not available. This is the default for Cisco Unified CME 4.0 and later versions.
dss	Transfers calls with consultation to idle monitor lines. All other call-transfer behavior is identical to full-consult.
local-consult	Transfers calls with local consultation using a second phone line, if available, or the calls fall back to blind if the target for consultation or transfer is not local. This mode is intended for use primarily in Voice over Frame Relay (VoFR) networks, because the Cisco VoFR call transfer protocol does not support an end-to-end transfer-with-consultation mechanism. Not supported if transfer-to destination is on the Cisco ATA, Cisco VG224, or a SCCP-controlled FXS port.

Command Default

For Cisco Unified CME 4.0 and later versions, the transfer mode is **full-consult**.
For Cisco CME 3.4 and earlier versions, the transfer mode is **blind**.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(11)YT	Cisco ITS 2.1	This command was introduced.
12.2(15)T	Cisco ITS 2.1	This command was integrated into Cisco IOS Release 12.2(15)T.
12.3(11)T	Cisco CME 3.2	The **dss** keyword was introduced.
12.4(4)XC	Cisco Unified CME 4.0	The command default was changed from **blind** to **full-consult**.
12.4(9)T	Cisco Unified CME 4.0	This command with the default of **full-consult** was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Direct station select is a functionality that allows a multibutton phone user to transfer calls to an idle monitor line by pressing the Transfer key and the appropriate monitor button. The **dss** keyword permits consultative call transfer to monitored lines.

Call transfers can be blind or consultative. A blind transfer is one in which the transferring extension connects the caller to a destination extension before ringback begins. A consultative transfer is one in which the transferring party either connects the caller to a ringing phone (ringback heard) or speaks with the third party before connecting the caller to the third party.

The **transfer-system** command specifies whether the H.450.2 standard or a Cisco proprietary method will be used to communicate call transfer information across the network. When you specify use of the H.450.2 consultative or blind mode of transfer globally by using the **transfer-system** command (or by using the default), you can override this mode for individual ephones by using the **transfer-mode** command. For example, in a system that is set up for consultative transfer, a specific extension with an auto-attendant that automatically transfers incoming calls to specific extension numbers can be set to use blind transfer, because auto-attendants do not use consultative transfer.

Prior to Cisco Unified CME 4.0, the default for this command specified the Cisco proprietary method. In Cisco Unified CME 4.0, the default was changed to specify the H.450.2 standard as the transfer method. Consult Table 57 for configuration recommendations for different versions of Cisco Unified CME.

Table 57 Transfer Method Recommendations

Cisco Product	transfer-system Command Default	transfer-system Command to Use	Transfer Method Recommendation
Cisco Unified CME 4.0 and later versions	**full-consult**	**full-consult** or **full-blind**	Use H.450.2 for call transfer. Because this is the default for this version, you do not need to use the **transfer-system** command unless you want to use the **full-blind** or **dss** keyword. Optionally, you can use the proprietary Cisco method by using the **transfer-system** command with the **blind** or **local-consult** keyword.
Cisco CME 3.0 to 3.3	**blind**	**full-consult** or **full-blind**	Use H.450.2 for call transfer. You must explicitly configure the **transfer-system** command with the **full-consult** or **full-blind** keyword because H.450.2 is not the default for this version. Optionally, you can use the proprietary Cisco method by using the **transfer-system** command with the **blind** or **local-consult** keyword.
Cisco ITS 2.1 to 3.0	**blind**	**blind** or **local-consult**	Use the Cisco proprietary method. Because this is the default for this version, you do not need to use the **transfer-system** command unless you want to use the **local-consult** keyword. Optionally, you can use the H.450.2 standard for call transfer by using **transfer-system** command with the **full-consult** or **full-blind** keyword. You must also configure the router with a Tcl script that is contained in the file called app-h450-transfer.x.x.x.x.zip. This file is posted on the Cisco Unified CME software download website at http://www.cisco.com/cgi-bin/tablebuild.pl/ip-iostsp.

Examples

The following example sets full consultation as the call transfer method:

```
Router(config)# telephony-service
Router(config-telephony)# transfer-system full-consult
```

Related Commands

Command	Description
transfer-mode	Specifies the type of call transfer for an individual IP phone extension that uses the H.450.2 standard.

translate (ephone-dn)

To apply a translation rule in order to manipulate the digits that are dialed by users of Cisco Unified IP phones, use the **translate** command in ephone-dn or ephone-dn-template configuration mode. To disable the translation rule, use the **no** form of this command.

translate {**called** | **calling**} *translation-rule-tag*

no translate {**called** | **calling**}

Syntax Description

called	Translate the called number.
calling	Translate the calling number.
translation-rule-tag	Unique sequence number by which the rule set is referenced. This number is arbitrarily chosen. Range is from 1 to 2147483647. There is no default value.

Command Default

No translation rule is applied.

Command Modes

Ephone-dn configuration
Ephone-dn-template configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(2)XT	Cisco ITS 2.0	This command was introduced on the Cisco 1750, Cisco 1751, Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	Cisco ITS 2.0	This command was implemented on the Cisco 2600XM and Cisco 2691.
12.2(11)T	Cisco ITS 2.01	This command was implemented on the Cisco 1760.
12.4(4)XC	Cisco Unified CME 4.0	This command was made available in ephone-dn-template configuration mode.
12.4(9)T	Cisco Unified CME 4.0	This command in ephone-dn-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command allows you to select a preconfigured translation rule to modify the number dialed by a specific extension (Cisco Unified IP phone destination number, or ephone-dn). A translation rule is a general-purpose digit-manipulation mechanism that performs operations such as automatically adding telephone area and prefix codes to dialed numbers. The translation rules are applied to the voice ports created by the ephone-dn. The **called** keyword translates the called number, and the **calling** keyword translates the calling number.

The translation rule mechanism inserts a delay into the dialing process when digits are entered that do not explicitly match any of the defined translation rules. This delay is set by the interdigit timeout. The translation-rule mechanism uses the delay to ensure that it has acquired all of the digits from the phone user before making a final decision that there is no translation-rule match available (and therefore no translation operation to perform). To avoid this delay, it is recommended that you include a dummy translation rule to act as a pass-through rule for digit strings that do not require translation. For example, a rule like "^5 5" that maps a leading 5 digit into a 5 would be used to prevent the translation rule delay being applied to local extension numbers that started with a 5.

Note For this command to take effect, appropriate translation rules must have been created at the VoIP configuration level. Use the **show voice translation-rule** command to view the translation rules that you have defined. For in formation, see the *Dial Peer Configuration on Voice Gateway Routers*.

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

Examples

The following example applies translation rule 20 to numbers called by extension 46839:

```
Router(config)# translation-rule 20
Router(config-translate)# rule 0 1234 2345 abbreviated
Router(config-translate)# exit
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 46839
Router(config-ephone-dn)# translate called 20
```

The following example uses an ephone-dn-template to apply translation rule 20 to numbers called by extension 46839:

```
Router(config)# translation-rule 20
Router(config-translate)# rule 0 1234 2345 abbreviated
Router(config-translate)# exit
Router(config)# ephone-dn-template 1
Router(config-ephone-dn-template)# translate called 20
Router(config-ephone-dn-template)# exit
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 46839
Router(config-ephone-dn)# ephone-dn-template 1
```

Related Commands

Command	Description
rule	Defines a translation rule.
translation-rule	Creates a translation identifier and enters translation-rule configuration mode.

translate-outgoing (voice register pool)

To allow an explicit setting of translation rules on the VoIP dial peer in order to modify a phone number dialed by any Cisco IP phone user, use the **translate-outgoing** command in voice register pool configuration mode. To disable translation rules, use the **no** form of this command.

translate-outgoing {**called** | **calling**} *rule-tag*

no translate-outgoing {**called** | **calling**}

Syntax Description

called	Called party requires translation.
calling	Calling party requires translation.
rule-tag	The *rule-tag* is an arbitrarily chosen number by which the rule set is referenced. The range is from 1 to 2147483.

Command Default

None

Command Modes

Voice register pool configuration

Command History

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

Usage Guidelines

Translation rules are a powerful general-purpose number-manipulation mechanism that perform operations such as automatically adding telephone area and prefix codes to dialed numbers. The translation rules are applied to VoIP dial peers created by Cisco Unified Session Initiation Protocol (SIP) Survivable Remote Site Telephony (SRST) or Cisco Unified CallManager Express (Cisco Unified CME)

.

During registration, a dial peer is created, and that dial peer includes a default translation rule. The **translate-outgoing** command allows you to change the translation rule, if desired. The **translate-outgoing** command allows you to select a preconfigured number translation rule to modify the number dialed by a specific extension.

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

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

Examples

Cisco Unified CME

The following is partial sample output from the **show running-config** command showing that called-party 1 requires translation.

```
voice register pool 1
 id mac 0030.94C2.A22A
 preference 5
 cor incoming call91 1 91011
 translate-outgoing called 1
```

Cisco Unified SIP SRST

The following is partial sample output from the **show running-config** command showing that called-party 1 requires translation.

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

Related Commands

Command	Description
alias (voice register pool)	Allows Cisco SIP IP phones to handle inbound PSTN calls to telephone numbers that are unavailable when the main proxy is not available.
id (voice register pool)	Explicitly identifies a locally available individual Cisco SIP IP phone, or when running Cisco Unified SIP SRST, set of Cisco SIP IP phones.
translate-outgoing (dial-peer)	Applies a translation rule to manipulate dialed digits on an outbound POTS or VoIP call leg.
voice register pool	Enters voice register pool configuration mode for SIP phones.

translation-profile

To assign a translation profile for incoming or outgoing call legs on a Cisco Unified IP phone, use the **translation-profile** command in ephone-dn or ephone-dn-template configuration mode. To delete the translation profile from the voice port, use the **no** form of this command.

translation-profile {**incoming** | **outgoing**} *name*

no translation-profile {**incoming** | **outgoing**} *name*

Syntax Description

incoming	Specifies that this translation profile handles incoming calls.
outgoing	Specifies that this translation profile handles outgoing calls.
name	Name of the translation profile.

Command Default

No default behavior or values

Command Modes

Ephone-dn configuration
Ephone-dn-template configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)T	Cisco CME 3.2	This command was introduced.
12.4(4)XC	Cisco Unified CME 4.0	This command was made available in ephone-dn-template configuration mode.
12.4(9)T	Cisco Unified CME 4.0	This command in ephone-dn-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Use the **translation-profile** command to assign a global predefined translation profile to an incoming or outgoing call leg.

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

Examples

The following example assigns the translation profile named call_in to handle translation of incoming calls and a translation profile named call_out to handle outgoing calls:

```
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 2555
Router(config-ephone-dn)# translation-profile incoming call_in
Router(config-ephone-dn)# translation-profile outgoing call_out
```

The following example uses an ephone-dn-template to assign the translation profile named call_in to handle translation of incoming calls and the translation profile named call_out to handle outgoing calls:

```
Router(config)# ephone-dn-template 10
Router(config-ephone-dn-template)# translation-profile incoming call_in
Router(config-ephone-dn-template)# translation-profile outgoing call_out
Router(config-ephone-dn-template)# exit
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 2555
Router(config-ephone-dn)# ephone-dn-template 10
```

Related Commands

Command	Description
show voice translation-profile	Displays the configuration of a translation profile.
translate	Applies a translation rule to modify the phone number dialed or received by any Cisco Unified IP phone user.
translation-rule	Creates a translation name and enters translation-rule configuration mode.
voice translation-profile	Defines a translation profile for voice calls.
voice translation-rule	Defines a translation rule for voice calls.

translation-profile incoming

To assign a translation profile for incoming call legs on a SIP phone, use the **translation-profile incoming** command in voice-register-dn configuration mode. To delete the translation profile from the directory number, use the **no** form of this command.

translation-profile incoming *name*

no translation-profile incoming

Syntax Description

name	Name of the translation profile to apply to incoming calls to this directory number. This is the *name* argument that was created for the profile with the **voice translation-profile** command.

Command Default

No translation profile is assigned to the directory number.

Command Modes

Voice-register-dn configuration

Command History

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

Usage Guidelines

Use this command to assign a predefined translation profile to incoming call legs on the specified directory number. The translation profile that you assign is created by using the **voice translation-profile** command.

Examples

The following example shows that the translation profile named call_in is assigned to handle translation of incoming calls to directory number 1:

```
Router(config)# voice register dn 1
Router(config-register-dn)# number 2555
Router(config-register-dn)# translation-profile incoming call_in
```

Related Commands

Command	Description
show voice translation-profile	Displays the configuration of a translation profile.
translate (translation profiles)	Associates a translation rule with a voice translation profile.
voice translation-profile	Defines a translation profile for voice calls.
voice translation-rule	Defines a translation rule for voice calls.

trunk

To associate an ephone-dn with a foreign exchange office (FXO) port, use the **trunk** command in ephone-dn configuration mode. To disassociate the ephone-dn from the trunk number, use the **no** form of this command.

trunk *digit-string* [**timeout** *seconds*] [**transfer-timeout** *seconds*] [**monitor-port** *port*]

no trunk

Syntax Description

digit-string	The number of the trunk line.
timeout *seconds*	(Optional) Interdigit timeout between dialed digits, in seconds. Range is 3 to 30. Default is 3.
transfer-timeout *seconds*	(Optional) Number of seconds that Cisco Unified CME waits for the transfer-to party to answer a call after which the call is recalled to the phone that initiated the transfer. This keyword is supported for dual-line ephone-dns only. Range is 5 to 60000. Default is disabled.
monitor-port *port*	(Optional) Enables a button lamp or icon that shows that the specified port is in use. *Port* argument is platform-dependent; type **?** to display syntax.

Command Default

Ephone-dns are not associated with FXO ports.

Command Modes

Ephone-dn configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(11)T	Cisco CME 3.2	This command was introduced.
12.4(4)XC	Cisco Unified CME 4.0	The **monitor-port** and **transfer-timeout** keywords were added and support for dual-line ephone-dns was added.
12.4(9)T	Cisco Unified CME 4.0	The enhancements in Cisco IOS Release 12.4(4)XC were integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command configures ephone-dns to support FXO lines that allow phones to have private lines connected directly to the PSTN. To bind the ephone-dn to the FXO port, use the destination pattern configured for the FXO line's POTS dial peer for the *digit-string* argument.

The **timeout** *seconds* argument controls the interdigit delay period, during which digits are collected from the user, and the delay before the connection to the FXO port is established. The argument controls the amount of time that Cisco Unified CME waits to collect digits for the dialed number, so that the digits can be included in the redial buffer and the Placed Calls directory of the phone. Digits that are entered after the timeout period are not included in the redial buffer or in the Placed Calls directory on the phone. The timeout parameter does not affect the time used to cut through the connection from the phone's trunk button to the FXO port. The phone user must either enter the pound (#) key or wait for this interdigit timeout to complete digit collection.

The phone user also has the option to use the phone's on-hook dialing feature so that the phone itself performs complete dial-string digit collection before signaling off-hook to the Cisco Unified CME. In this case all digits will be included in the Redial and Placed Calls Directory.

The **monitor-port** keyword enables direct status monitoring of the FXO port on the line button of the IP phone. The line button indicator, either a lamp or an icon depending on the phone, shows the in-use status of the FXO port during the duration of the call.

The **transfer-timeout** argument enables a transferred call to be automatically recalled if the transfer target does not answer after the specified number of seconds. The call is withdrawn from the transfer-to phone and the call resumes ringing on the phone that initiated the transfer.

The **monitor-port** and **transfer-timeout** keywords are not supported on ephone-dns for analog ports on the Cisco VG 224.

For dual-line ephone-dns, the second channel cannot receive incoming calls when the **trunk** command is configured.

Examples

The following example shows the configuration for two phones that each have a private FXO line button and a shared-line button.

The shared line's voice ports and dial peers are as follows:

```
Router(config)# voice-port 1/0/1
Router(config-voice-port)# connection plar-opx 1000

Router(config)# dial-peer voice 101 pots
Router(config-dial-peer)# destination-pattern 9
Router(config-dial-peer)# port 1/0/1
```

The private lines' voice ports and dial peers are as follows:

```
Router(config)# voice-port 1/1/0
Router(config-voice-port)# connection plar-opx 5550111

Router(config)# dial-peer voice 110 pots
Router(config-dial-peer)# destination-pattern 80
Router(config-dial-peer)# port 1/1/0

Router(config)# voice-port 1/1/1
Router(config-voice-port)# connection plar-opx 5550112

Router(config)# dial-peer voice 111 pots
Router(config-dial-peer)# destination-pattern 81
Router(config-dial-peer)# port 1/1/1
```

The following is the configuration for the shared and private ephone-dns:

```
Router(config)# ephone-dn 1
Router(config-ephone-dn)# number 1000
Router(config-ephone-dn)# name Line1
Router(config-ephone-dn)# no huntstop

Router(config)# ephone-dn 2
Router(config-ephone-dn)# number 5550111
Router(config-ephone-dn)# name Private line
Router(config-ephone-dn)# trunk 80
```

```
Router(config)# ephone-dn 3
Router(config-ephone-dn)# number 5550112
Router(config-ephone-dn)# name Private line
Router(config-ephone-dn)# trunk 81
```

The following is the configuration for ephones with button 1 as a shared line and button 2 a private line:

```
Router(config)# ephone 1
Router(config-ephone)# mac-address 1111.1111.1101
Router(config-ephone)# button 1:1 2:2

Router(config)# ephone 2
Router(config-ephone)# mac-address 1111.1111.1102
Router(config-ephone)# button 1:1 2:3
```

The following example shows that transferred calls are recalled after 30 seconds if the destination party does not answer and status monitoring is enabled for FXO port 1/1/1.

```
Router(config)# ephone-dn 5
Router(config-ephone-dn)# trunk 801 timeout 5 transfer-timeout 30 monitor-port 1/1/1
```

Related Commands

Command	Description
destination-number	Specifies a connection mode for a voice port.

trustpoint (credentials)

To specify the name of the trustpoint to be associated with a Cisco Unified CME CTL provider certificate or with the Cisco Unified SRST router certificate, use the **trustpoint** command in credentials configuration mode. To change the specified trustpoint, use the **no** form of this command.

trustpoint *trustpoint-name*

no trustpoint

Syntax Description

trustpoint-name	Name of the trustpoint to be associated with the Cisco Unified CME CTL provider certificate or the Cisco Unified SRST device certificate.

Command Default

No default behavior or values.

Command Modes

Credentials configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.3(14)T	Cisco SRST 3.3	This command was introduced for Cisco SRST.
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced for Cisco Unified CME.
12.4(9)T	Cisco Unified CME 4.0	This command for Cisco Unified CME was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Cisco Unified CME

This command is used with Cisco Unified CME phone authentication to define the trustpoint for the CTL provider. This trustpoint will be used for TLS sessions with the CTL client.

Cisco Unified SRST

The name of the trustpoint must be consistent with the trustpoint name of the Cisco Unified SRST router.

Examples

Cisco Unified CME

The following example sets up a CTL provider on the Cisco Unified CME router with the IP address 172.19.245.1.

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

Cisco Unified SRST

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

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

Related Commands

Command	Description
ctl-service admin	Specifies a user name and password to authenticate the CTL client during the CTL protocol.
debug credentials	Sets debugging on the credentials service.
ip source-address (credentials)	Enables the router to receive messages through the specified IP address and port.
show credentials	Displays the credentials settings.

trustpoint-label

To specify the PKI trustpoint label to be used for the TLS connection between the CAPF server and the phone, use the **trustpoint-label** command in CAPF-server configuration mode. To return to the default, use the **no** form of this command.

trustpoint-label *label*

no trustpoint-label

Syntax Description

label	Trustpoint name for the CAPF server.

Command Default

No trustpoint label is specified for TLS connections.

Command Modes

CAPF-server configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

This command is used with Cisco Unified CME phone authentication to provide a PKI trustpoint name for the CAPF server. This trustpoint label is used for the TLS connection between the CAPF server and the phone.

Examples

The following example defines the CAPF server trustpoint name as server25.

```
Router(config)# capf-server
Router(config-capf-server)# source address 10.10.10.1
Router(config-capf-server)# trustpoint-label server25
Router(config-capf-server)# cert-oper upgrade all
Router(config-capf-server)# cert-enroll-trustpoint server12 password 0 x8oWiet
Router(config-capf-server)# auth-mode auth-string
Router(config-capf-server)# auth-string generate all
Router(config-capf-server)# port 3000
Router(config-capf-server)# keygen-retry 5
Router(config-capf-server)# keygen-timeout 45
Router(config-capf-server)# phone-key-size 2048
```

type

To define a phone type or to define one or two add-on phone modules for a Cisco Unified IP phone, use the **type** command in ephone or ephone-template configuration mode. To remove a definition, use the **no** form of this command.

type *phone-type* [**addon 1** *module-type* [**2** *module-type*]]

no type *phone-type* [**addon 1** *module-type* [**2** *module-type*]]

Syntax Description

phone-type	Type of IP phone that is being defined or the type of IP phone to which a module is being added. Valid entries are as follows: • **7902**—Cisco Unified IP Phone 7902G. • **7905**—Cisco Unified IP Phone 7905G. • **7910**—Cisco Unified IP Phones 7910 and 7910G. • **7911**—Cisco Unified IP Phone 7911G. • **7912**—Cisco Unified IP Phone 7912G. • **7920**—Cisco Unified IP Phone 7920. • **7921**—Cisco Unified Wireless IP Phone 7921G. • **7931**—Cisco Unified IP Phone 7931G. • **7935**—Cisco Unified IP Conference Station 7935. • **7936**—Cisco Unified IP Conference Station 7936. • **7940**—Cisco Unified IP Phone 7940G. • **7941**—Cisco Unified IP Phone 7941G. • **7941GE**—Cisco Unified IP Phone 7941G-GE. • **7960**—Cisco Unified IP Phone 7960G. • **7961**—Cisco Unified IP Phone 7961G. • **7961GE**—Cisco Unified IP Phone 7961G-GE. • **7970**—Cisco Unified IP Phone 7970G. • **7971**—Cisco Unified IP Phone 7971G-GE. • **anl**—Analog. • **ata**—Cisco ATA-186 or Cisco ATA-188. • **CIPC**—Cisco IP Communicator.
addon 1 *module-type*	(Optional) Tells the router that a module is being added to this Cisco Unified IP phone and the type of module. The valid entry for *module-type* follows: • **7914**—Cisco Unified IP Phone 7914 Expansion Module.
2 *module-type*	(Optional) Tells the router that a second module is being added to this Cisco Unified IP phone and the type of module. The valid entry for *module-type* follows: • **7914**—Cisco Unified IP Phone 7914 Expansion Module.

Command Default

No phone type or add-on module is defined.

Command Modes

Ephone configuration
Ephone-template configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(11)YT	Cisco ITS 2.1	This command was introduced.
12.2(15)T	Cisco ITS 2.1	This command was integrated into Cisco IOS Release 12.2(15)T.
12.2(15)ZJ	Cisco CME 3.0	The following keywords were added to this command: **7902**, **7905**, and **7912**.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.3(7)T	Cisco CME 3.1	The **7920** and **7936** keywords were added.
12.3(11)XL	Cisco CME 3.2.1	The **7970** keyword was added.
12.3(14)T	Cisco CME 3.3	The **7971** keyword was added, and this command was integrated into Cisco IOS Release 12.3(14)T.
12.4(4)XC	Cisco Unified CME 4.0	The **7911**, **7941**, **7941GE**, **7961**, and **7961GE** keywords were added. This command was made available in ephone-template configuration mode.
12.4(9)T	Cisco Unified CME 4.0	This command with the **7911**, **7941**, **7941GE**, **7961**, and **7961GE** keywords and this command in ephone-template configuration mode was integrated into Cisco IOS Release 12.4(9)T.
12.4(6)XE	Cisco Unified CME 4.0(2)	The **7931** keyword was added.
12.4(4)XC4	Cisco Unified CME 4.0(3)	The **7931** keyword was added.
12.4(11)T	Cisco Unified CME 4.0(3)	The **7931** keyword was integrated into Cisco IOS Release 12.4(11)T.
12.4(11)XJ2	Cisco Unified CME 4.1	The **7921** keyword was added.
12.4(15)T	Cisco Unified CME 4.1	This command was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

For Cisco Unified CME 4.0 and later versions, the only *phone-type* to which you can apply an add-on module are **7960**, **7961**, **7961GE**, and **7970**. For Cisco Unified CME 3.4 and earlier versions, the only *phone-type* to which you can apply an add-on module is **7960**.

This command must be followed by a phone reboot using the **reset** command.

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

Examples

The following example defines the IP phone with phone-tag 10 as a Cisco Unified IP Phone 7960G with 2 attached Cisco Unified IP Phone 7914 Expansion Modules:

```
Router(config)# ephone 10
Router(config-ephone)# type 7960 addon 1 7914 2 7914
```

The following example defines the IP phone with phone-tag 4 as a Cisco ATA device:

```
Router(config)# ephone 4
Router(config-ephone)# mac 1234.87655.234
Router(config-ephone)# type ata
```

Related Commands

Command	Description
reset (ephone)	Performs a complete reboot of one phone associated with a Cisco Unified CME router.
reset (telephony-service)	Performs a complete reboot of one or all phones associated with a Cisco Unified CME router.

type (voice register dialplan)

To specify a phone type for a SIP dial plan, use the **type** command in voice register dialplan configuration mode. To remove a phone type, use the **no** form of this command.

type *phone-type*

no type

Syntax Description

phone-type	Type of SIP phone for which the dial plan is used. Values are: • **7905-7912**—Cisco Unified IP Phone 7905, 7905G, 7912, or 7912G. • **7940-7960-others**—Cisco Unified IP Phone 7911, 7940, 7940G, 7941, 7941GE, 7960, 7960G, 7961, 7961GE, 7970, or 7971.

Command Default

The phone type is not defined.

Command Modes

Voice register dialplan configuration

Command History

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

Usage Guidelines

This command specifies the type of SIP phone for which the dial plan is defined. You must use this command before defining dial patterns with the **pattern** command or selecting a dial pattern file in flash with the **filename** command.

The phone type specified with this command must match the phone type specified with the **type** command in voice register pool mode. If the dial plan type does not match the type assigned to the phone, the dial-plan configuration file is not generated.

Examples

The following example shows a SIP dial plan being defined for a Cisco Unified IP Phone 7905 or Cisco Unified IP Phone 7912:

```
Router(config)# voice register dialplan 10
Router(config-register-dialplan)# type 7905-7912
Router(config-register-dialplan)# pattern 52...
Router(config-register-dialplan)# pattern 91.......
```

Related Commands

Command	Description
dialplan	Assigns a dial plan to a SIP phone.
filename	Specifies a custom XML file that contains the dial patterns to use for the SIP dial plan.
pattern (voice register dialplan)	Defines a dial pattern for a SIP dial plan.
show voice register dialplan	Displays configuration information for a specific SIP dial plan.
type (voice register pool)	Defines a phone type for a SIP phone.

type (voice register pool)

To define a phone type for a SIP phone, use the **type** command in voice register pool configuration mode. To remove a phone type, use the **no** form of this command.

type *phone-type*

no type *phone-type*

Syntax Description

phone-type	Type of SIP phone that is being defined. Valid entries are as follows: • **3951**—Cisco Unified IP Phone 3951 • **7905**—Cisco Unified IP Phone 7905 and 7905G. • **7911**—Cisco Unified IP Phone 7911G. • **7912**—Cisco Unified IP Phone 7912 and 7912G. • **7940**—Cisco Unified IP Phone 7940 and 7940G. • **7941**—Cisco Unified IP Phone 7941G. • **7941GE**—Cisco Unified IP Phone 7941GE. • **7960**—Cisco Unified IP Phone 7960 and 7960G. • **7961**—Cisco Unified IP Phone 7961G. • **7961GE**—Cisco Unified IP Phone 7961GE. • **7970**—Cisco Unified IP Phone 7970G. • **7971**—Cisco Unified IP Phone 7971GE. • **ATA**—Cisco ATA-186 or Cisco ATA-188. • **P100**—PingTel Xpressa 100. • **P600**—Polycom SoundPoint 600.

Command Default

No phone type is defined.

Command Modes

Voice register pool configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.
12.4(11)XJ	Cisco Unified CME 4.1	The **3951**, **7911**, **7941**, **7941GE**, **7961**, **7961GE**, **7970**, and **7971** keywords were added.
12.4(15)T	Cisco Unified CME 4.1	The **3951**, **7911**, **7941**, **7941GE**, **7961**, **7961GE**, **7970** and **7971** keywords were added and this command was integrated into Cisco IOS Release 12.4(6th)T.

Usage Guidelines

After you use this command, reboot the phone with the **restart** or **reset** command.

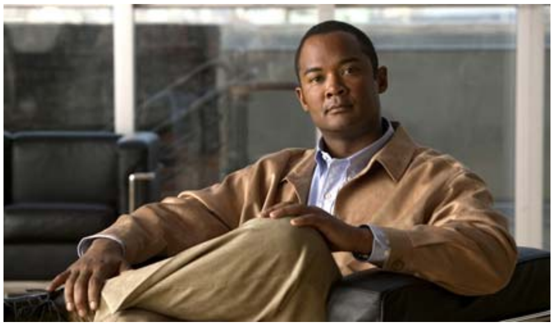
Examples

The following example shows how to define the SIP phone with phone-tag 10 as a Cisco Unified IP Phone 7960 or Cisco Unified IP Phone 7960G:

```
Router(config)# voice register pool 10
Router(config-register-pool)# type 7960
```

Related Commands

Command	Description
load (voice register global)	Associates a type of Cisco Unified IP phone with a phone firmware file.
reset (voice register global)	Performs a complete reboot of all SIP phones associated with a Cisco CME router.
reset (voice register pool)	Performs a complete reboot of one SIP phone associated with a Cisco CME router.
restart (voice register)	Perform a fast reset of one or all SIP phones associated with a Cisco Unified CME router.



Cisco Unified CME Commands: U

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This chapter contains commands to configure and maintain Cisco Unified Communications Manager Express (formally known as Cisco Unified CallManager Express). The commands are presented in alphabetical order. Some commands required for configuring Cisco Unified Communications Manager Express (Cisco Unified CME) may be found in other Cisco IOS command references. Use the command reference master index or search online to find these commands.

upgrade (voice register global)

To generate a OS79XX.TXT file for firmware upgrades, use the **upgrade** command in voice register global configuration mode. To return to the default, use the **no** form of this command.

upgrade

no upgrade

Syntax Description

This command has no arguments or keywords.

Defaults

No OS79XX.TXT file generated.

Command Modes

Voice register global configuration

Command History

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

Usage Guidelines

The upgrade command performs the TFTP server alias binding, which can be verified with the **show voice register tftp-bind** command.

Examples

The following example shows the use of the **upgrade** command to upgrade Cisco SIP phone firmware from SIP [456].x to SIP [567].y with comments:

```
Router(config)# voice register global
Router(config-register-global)# load 7960-7940 P00x... !Do not use file extension.
Router(config-register-global)# upgrade                !Generates OS79XX.txt file.
Router(config-register-global)# load 7960-7940 POSx... !Do not use file extension. This
                                                       !command is only required if you
                                                       !are upgrading to 7.y.
Router(config-register-global)# create profile  !Generates SIPDefault.cnf and other files.
Router(config-register-global)# reset
Router(config-register-global)# no upgrade  !Returns command to default condition.
```

The P00x... and P0Sx... firmware filenames are required because the content in OS79XX.TXT is different from image_version tag in SIPDefault.cnf.

url (telephony-service)

To provision uniform resource locators (URLs) for Cisco Unified IP phones connected to the Cisco Unified CME router, use the **url** command in telephony-service configuration mode. To remove a URL association, use the **no** form of this command.

url {**authentication** | **directories** | **information** | **messages** | **proxy-server** | **services**} *url*

no url {**authentication** | **directories** | **information** | **messages** | **proxy-server** | **services**}

Syntax Description

authentication	Uses the information at the specified URL to validate requests made to the phone web server.
directories	Uses the information at the specified URL for the Directories button display.
information	Uses the information at the specified URL for the Information button display. This button may be labeled "i" or "?". **Note** Cisco Unified CME does not support the use of this URL.
messages	Uses the information at the specified URL for the Messages button display.
proxy-server	Specifies the host and port used to enable proxy HTTP requests for access to nonlocal host addresses from the phone HTTP client.
services	Uses the information at the specified URL for the Services button display.
url	URL as defined in RFC 2396.

Command Default

The router automatically uses the local directory service.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(2)XT	Cisco ITS 2.0	This command was introduced.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T.

Usage Guidelines

Cisco Unified IP Phones can support URLs in association with the four programmable feature buttons on those IP phones: Directories, Information, Messages, and Services. The fifth button, Settings, is managed entirely by the phone. Operation of these services is determined by the Cisco Unified IP phone capabilities and the content of the referenced URL.

This command provisions URLs through the configuration file supplied by the Cisco Unified CME router to the Cisco Unified IP phones during phone registration.

Note Cisco Unified CME does not support provisioning an information URL to access help using the i or ? buttons on a phone.

To use a Cisco Unified CallManager directory as an external directory source for Cisco Unified CME phones, the Cisco Unified CallManager must be made aware of the phones. You must list the MAC addresses of the Cisco Unified CME phones in the Cisco Unified CallManager and reset the phones from the Cisco Unified CallManager. It is not necessary for you to assign ephone-dns to the phones or for the phones to register with Cisco Unified CallManager.

Note Provisioning of the directory URL to select an external directory resource disables the Cisco Unified CME local directory service.

You can disable the local directory by using the **no service local-directory** command.

The **url (telephony-service)** command must be followed by a complete phone reboot using the **reset** command.

Examples

The following example provisions the Directories and Services buttons. Note that the Messages button is configured by the **voicemail** command. The Messages button acts like a speed-dial key to retrieve messages from a specified telephone number.

```
Router(config)# telephony-service
Router(config-telephony)# url directories http://1.4.212.11/localdirectory
Router(config-telephony)# url services http://1.4.212.4/CCMUser/123456/urltest.html
```

Related Commands

Command	Description
reset (ephone)	Performs a complete reboot of one phone associated with a Cisco Unified CME router.
reset (telephony-service)	Performs a complete reboot of one or all phones associated with a Cisco Unified CME router.
service local-directory	Enables the availability of the local directory service on IP phones.
voicemail	Defines the telephone number that is speed-dialed when the Messages button on a Cisco Unified IP phone is pressed.

url (voice register global)

To provision uniform resource locators (URLs) for feature buttons on Cisco SIP IP phones connected to a Cisco CallManager Express (Cisco CME) router, use the **url** command in voice register global configuration mode. To remove a URL association, use the **no** form of this command.

url {**director**y | **service**} *url*

no url {**directory** | **service**}

Syntax Description

directory	Uses the information at the specified URL for the Directories button display.
service	Uses the information at the specified URL for the Services button display.
url	URL as defined in RFC 2396.

Defaults

The router automatically uses the local directory service.

Command Modes

Voice register global configuration

Command History

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

Usage Guidelines

The Cisco IP Phones 7940 and 7940G and Cisco IP Phones 7960 and 7960G can support two URLs in association with two programmable feature buttons: Directories and Services. Operation of these services is determined by the Cisco IP phone capabilities and the content of the specified URL. The Settings button is managed entirely by the phone. The Messages button is configured by the **voicemail** command.

The purpose of the **url** command is to provision the URLs through the configuration file supplied by the Cisco CallManager Express router to the SIP phones during phone registration.

You can disable the local directory by specifying the string “none” instead of a URL with the **directory** keyword, as shown in the following example:

```
Router(config)# voice register global
Router(config-register-global)# url directory none
```

Note Provisioning the directory URL to select an external directory resource disables Cisco CallManager Express local directory service.

After you configure this command, restart the phone by using the **reset** command.

Examples

The following example shows how to provision the Directories and Services buttons:

```
Router(config)# voice register global
Router(config-register-global)# url directory http://1.4.212.11/localdirectory
```

```
Router(config-register-global)# url service http://1.4.212.4/CCMUser/123456/urltest.html
```

Related Commands

Command	Description
reset (voice register pool)	Performs a complete reboot of one phone associated with a Cisco CME router.
reset (voice register global)	Performs a complete reboot of all SIP phones associated with a Cisco CME router.
telephony-service	Enters telephony-service configuration mode.
voicemail (voice register template)	Defines the telephone number that is speed-dialed when the Messages button on a Cisco IP phone is pressed.

url idle

To specify a file to display on an IP phone that is not in use, use the **url idle** command in telephony-service configuration mode. To disable display of the file, use the **no** form of this command.

url idle *url* **idle-timeout** *seconds*

no url idle

Syntax Description

url	URL as defined in RFC 2396.
idle-timeout *seconds*	Time interval between display refreshes, in seconds. Range is from 0 to 300.

Defaults

No file is specified for display on idle phones.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.2(11)YT	2.1	This command was introduced.
12.2(15)T	2.1	This command was integrated into Cisco IOS Release 12.2(15)T.

Usage Guidelines

Use this command with Cisco IOS Telephony Services V2.1, Cisco Call Manager Express 3.0, or a later version.

The file that is displayed must be encoded in eXtensible Markup Language (XML) using the Cisco XML document type definition (DTD). For more information about Cisco DTD formats, refer to *Cisco IP Phone Services Application Development Notes*.

This command must be followed by a complete phone reboot using the **reset** command.

Examples

The following example specifies that the file logo.xml should be displayed on IP phones when they are not being used and that the display should be refreshed every 12 seconds:

```
Router(config)# telephony-service
Router(config-telephony)# url idle http://mycompany.com/files/logo.xml idle-timeout 12
```

Related Commands

Command	Description
reset (ephone)	Performs a complete reboot of one phone associated with a Cisco CME router.
reset (telephony-service)	Performs a complete reboot of one or all phones associated with a Cisco CME router.
telephony-service	Enters telephony-service configuration mode.

user-locale (ephone-template)

To specify a language tag identifier in an ephone template, use the **user-locale** command in ephone-template configuration mode. To use the default user locale, use the **no** form of this command.

user-locale *language-tag*

no user-locale

Syntax Description

language-tag	Language tag identifier that was assigned to an alternative user locale using the **user-locale (telephony-service)** command.

Command Default

The default user locale (user-locale 0) is used.

Command Modes

Ephone-template configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

To apply user-locale language tag identifiers to individual ephones, you must specify per-phone configuration files using the **cnf-file perphone** command and define the language tags using the **user-locale (telephony-service)** command.

After creating an ephone template that contains a language tag identifier, use the **ephone-template (ephone)** command to apply the template to individual ephones.

Examples

The following example defines three alternative locales: JP (Japan), FR (France), and ES (Spain). The default is US for all phones that do not have the alternatives applied using ephone templates. In this example, ephone 11 uses JP for its locales, ephone 12 uses FR, ephone 13 uses ES, and ephone 14 uses the default, US.

```
telephony-service
 cnf-file location flash:
 cnf-file perphone
 user-locale 1 JP
 user-locale 2 FR
 user-locale 3 ES
 network-locale 1 JP
 network-locale 2 FR
 network-locale 3 ES
 create cnf-files

ephone-template 1
 user-locale 1
```

```
 network-locale 1

ephone-template 2
 user-locale 2
 network-locale 2

ephone-template 3
 user-locale 3
 network-locale 3

ephone 11
 button 1:25
 ephone-template 1

ephone 12
 button 1:26
 ephone-template 2

ephone 13
 button 1:27
 ephone-template 3

ephone 14
 button 1:28
```

Related Commands

Command	Description
cnf-file	Specifies the type of configuration files that phones use.
ephone-template (ephone)	Applies an ephone template to an ephone.
user-locale (telephony-service)	Sets the language for displays on supported phone types.

user-locale (telephony-service)

To set the language for displays on supported phone types, use the **user-locale** command in telephony-service configuration mode. To disable the selected setting, use the **no** form of this command.

user-locale [*user-locale-tag* [*user-defined-code*]] *language-code*

no user-locale [user-*locale-tag*] *language-code*

Syntax Description

user-locale-tag	(Optional) Assigns a locale identifier to the specified language code. Range is 0 to 4. Default is 0.
user-defined-code	(Optional) Assigns one of the user-defined codes to the specified language code. Valid codes are **U1**, **U2**, **U3**, **U4**, and **U5**. There is no default.
language-code	Language files for the following ISO 3166 codes are installed in system storage for supported phone types: • **DE**—German • **DK**—Danish • **ES**—Spanish • **FR**—French • **IT**—Italian • **JP**—Japanese • **NL**—Dutch • **NO**—Norwegian • **PT**—Portuguese • **RU**—Russian • **SE**—Swedish • **US**—United States **Note** You can also assign any valid ISO 639 code that is not listed above to a user-defined code (U1 to U5), but you must first copy the appropriate XML language files into slot 0, flash, or TFTP memory and use the **cnf-files perphone** command to specify the use of per-phone configuration files.

Command Default

The default code is **US** (United States).

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(11)YT	Cisco ITS 2.1	This command was introduced.
12.2(15)T	Cisco ITS 2.1	This command was integrated into Cisco IOS Release 12.2(15)T.
12.2(15)ZJ	Cisco CME 3.0	The following keywords were added: **DK**, **NL**, **NO**, **PT**, **RU**, and **SE**.
12.3(4)T	Cisco CME 3.0	The keywords added for Cisco CME 3.0 were integrated into Cisco IOS Release 12.3(4)T.
12.4(4)XC	Cisco Unified CME 4.0	The *user-locale-tag* and *user-defined-code* arguments were added.
12.4(9)T	Cisco Unified CME 4.0	The *user-locale-tag* and *user-defined-code* arguments were integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

The **show telephony-service tftp-bindings** command displays the locale that has been set using this command. This locale is currently associated with the dictionary and language files.

This command must be followed by a complete phone reboot using the **reset** command.

Locale tag 0 always holds the default language, which is used for all phones that are not assigned alternative user locales or user-defined user locales. The system default is US, but you can define an different code to be the default, as shown in the "Examples" section.

Alternative User Locales

The *user-locale-tag* argument allows you to specify up to five alternative user locales in Cisco Unified CME 4.0 or a later release. For example, a company can specify French for phones A, B, and C; German for phones D, E, and F; and United States for phones G, H, and I.

Each of the five user locales that you can use in a multi-locale system is identified with the *locale-tag* argument. The identifier 0 always holds the default locale, although you can define this default to be any language code that is supported in the system and is listed in the CLI help for the command. For example, if you define user locale 0 to be JP (Japanese), the default user locale for the router is JP. If you do not specify a locale for identifier 0, the default is US (United States).

To apply alternative user locales to different phones, you must use the **cnf-files** command to specify per-phone configuration files. When you use per-phone configuration files, a phone's configuration file automatically uses the default locales in user locale 0 and network locale 0. You can override this default for individual ephones by assigning locale tags to the alternative language codes that you want to use and then creating ephone templates to assign the locale tag to individual ephones. For example, you can assign locale tag 2 to the language code RU (Russian).

Use the **user-locale** command in ephone-template mode to apply a locale tag to an ephone template. Use the **ephone-template** command in ephone configuration mode to apply the template to the phones that should use the alternative language. For an example, see the Alternative User Locale Example.

User-Defined User Locales

XML files for user locales and network locales that are not currently provided in the system must be downloaded in order to use this feature. In Cisco Unified CME 4.0 and later versions, you can install the files to support a particular user and network locale in slot 0, external TFTP, or flash memory. Note that these files cannot be installed in the system storage location. The user locales and network locales that are stored in this fashion can then be used as default or alternative entries for all or some phones.

For example, if you have a site at which the phones should use the displays and tones for Traditional Chinese, which is not one of the standard languages, you must download the XML files for Traditional Chinese to use this user-defined locale on a phone.

For an example, see the User-Defined User Locale Example.

Examples

The following example sets the default language tag for the IP phone display to French:

```
telephony-service
 user-locale FR
```

The following example sets the default language tag for the IP phone display to French. It shows another way to change the default:

```
telephony-service
 user-locale 0 FR
```

The following example sets the alternative language tag 1 to German:

```
telephony-service
 user-locale 1 DE
```

Alternative User Locale Example

The following example defines three alternative locales: JP (Japan), FR (France), and ES (Spain). The default is US for all phones that do not have the alternatives applied using ephone templates. In this example, ephone 11 uses JP for its locales, ephone 12 uses FR, ephone 13 uses ES, and ephone 14 uses the default, US.

```
telephony-service
 cnf-file location flash:
 cnf-file perphone
 user-locale 1 JP
 user-locale 2 FR
 user-locale 3 ES
 network-locale 1 JP
 network-locale 2 FR
 network-locale 3 ES
 create cnf-files

ephone-template 1
 user-locale 1
 network-locale 1

ephone-template 2
 user-locale 2
 network-locale 2

ephone-template 3
 user-locale 3
 network-locale 3

ephone 11
 button 1:25
 ephone-template 1

ephone 12
 button 1:26
 ephone-template 2
```

```
ephone 13
 button 1:27
 ephone-template 3

ephone 14
 button 1:28
```

User-Defined User Locale Example

The following example applies locale tag 4 to the user-defined code U1, which is defined as ZH. ZH is the code that represents Traditional Chinese in ISO 639, the Language Code Reference. Because the code for Traditional Chinese is not one of those that is provided in the system, the user must download the appropriate XML files to support this language.

In addition to the user-defined code, the example defines three alternative locales: JP (Japan), FR (France), and ES (Spain). The default is US for all phones that do not have the alternatives applied using ephone templates. In this example, ephone 11 uses JP for its locales; ephone 12 uses FR; ephone 13 uses ES; ephone 14 uses the default, US; and ephone 15 uses the user-defined language, ZH (Traditional Chinese).

```
telephony-service
 cnf-file location flash:
 cnf-file perphone
 user-locale 1 JP
 user-locale 2 FR
 user-locale 3 ES
 user-locale 4 U1 ZH
 network-locale 1 JP
 network-locale 2 FR
 network-locale 3 ES
 network-locale 4 U1 ZH
 create cnf-files

ephone-template 1
 user-locale 1
 network-locale 1

ephone-template 2
 user-locale 2
 network-locale 2

ephone-template 3
 user-locale 3
 network-locale 3

ephone-template 4
 user-locale 4
 network-locale 4

ephone 11
 button 1:25
 ephone-template 1

ephone 12
 button 1:26
 ephone-template 2

ephone 13
 button 1:27
 ephone-template 3
```

```
ephone 14
 button 1:28

ephone 15
 button 1:29
 ephone-template 4
```

Related Commands

Command	Description
cnf-files	Specifies the type of phone configuration files to be created.
ephone-template (ephone)	Applies an ephone template to an ephone.
network-locale (telephony-service)	Selects a code for a geographically specific set of tones and cadences on supported phone types.
reset (ephone)	Performs a complete reboot of one phone associated with a Cisco Unified CME router.
reset (telephony-service)	Performs a complete reboot of one or all phones associated with a Cisco Unified CME router.
show telephony-service tftp-bindings	Displays the current configuration files that are accessible to IP phones.
user-locale (ephone-template)	Applies a user locale tag to an ephone template.

username (ephone)

To assign a login account username and password to a phone user so that the user can log in to the Cisco IOS Telephony Service router through a web browser, use the **username** command in ephone configuration mode. To disable the username and password, use the **no** form of this command.

username *username* **password** *password*

no username *username* **password** *password*

Syntax Description

username	Username of the local Cisco IP phone user. Default is Admin.
password	Enables password for the Cisco IP phone user.
password	Password string.

Defaults

The default username for the administrator is Admin.

Command Modes

Ephone configuration

Command History

Release	Modification
12.2(2)XT	This command was introduced on the Cisco 1750, Cisco 1751, Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on Cisco 3725 and Cisco 3745 routers.
12.2(8)T1	This command was implemented on Cisco 2600-XM and Cisco 2691 routers.
12.2(11)T	This command was implemented on the Cisco 1760.

Usage Guidelines

This command assigns a login account username and password for a phone user and establishes a login account for each Cisco IP phone (ephone). This configuration can be completed only by the local system administrator of the Cisco IOS Telephony Service router.

You must also create a login account to allow Telephone Application Programming Interface (TAPI)-aware PC applications to register with the Cisco IOS Telephony Service router and exercise remote-control operation of a Cisco IP phone.

This configuration permits the phone user to log in to the Cisco IOS Telephony Service to view and change attributes associated only with the user's IP phone.

Examples

The following example shows how to set the username and password:

```
Router(config)# ephone 1
Router(config-ephone)# username smith password 9golf
```

Related Commands

Command	Description
admin-password	Sets a password for the local system administrator of the Cisco IOS Telephony Service.
admin-username	Sets the username for the local system administrator of the Cisco IOS Telephony Service router.

username (voice logout-profile)

To create an authentication credential be used used by a TAPI phone device and certain other applications to log into a Cisco Unified CME, use the username command in voice logout-profile configuration mode. To remove the credential, use the **no** form of this command.

username *username* **password** *password*

no username

Syntax Description

username	Alphanumeric string to be used by a TAPI phone device to log into Cisco Unified CME.
password	Password to be used with this user name for authentication purposes.
password	Alphanumeric string.

Command Default

Credential does not exist.

Command Modes

Voice logout-profile configuration (voice-logout-profile)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XW1	Cisco Unified CME 4.2	This command was introduced.

Usage Guidelines

This command in voice logout-profile configuration mode creates a credential to be used by a TAPI phone device and certain other applications to log into a Cisco Unified CME via an IP phone that is enabled for extension mobility and configured with the logout profile that includes this command.

Examples

The following example shows the configuration for a logout profile that defines the default appearance for a Cisco Unified IP phone that is enabled for extension mobility, including the username (23C2-8) and password (43214) to be used by a TAPI phone device to log into Cisco Unified CME.

```
pin 9999
user 23C2-8 password 43214
number 3001 type silent-ring
number 3002 type beep-ring
number 3003 type feature-ring
number 3004 type monitor-ring
number 3005,3006 type overlay
number 3007,3008 type cw-overly
speed-dial 1 2000
speed-dial 2 2001 blf
```

Related Commands

Command	Description
logout-profile	Enables Cisco Unified IP phone for extension mobility and assigns a logout profile to this phone.

username (voice register pool)

To assign an authentication credential to a phone user so that the SIP phone can register in Cisco CallManager Express (Cisco CME), use the **username** command in voice register pool configuration mode. To disable a username and password, use the **no** form of this command.

username *username* [**password** *password*]

no username *username* [**password** *password*]

Syntax Description

username	Username of the local Cisco IP phone user. Default: Admin.
password	Enables password for the Cisco IP phone user.
password	Password string.

Defaults

Disabled.

Command Modes

Voice register pool configuration

Command History

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

Usage Guidelines

Creates an authentication credential for SIP IP phone registration. This command is required if authentication is enabled with the **authenticate** command.

You must configure the voice register pool before configuring an authentication credential.

All lines in a phone share the same credential.

This configuration also permits the phone user to log in to Cisco Unified CME to view and change attributes associated only with the user's SIP IP phone.

Note This command is not for SIP proxy registration. The password will not be encrypted.

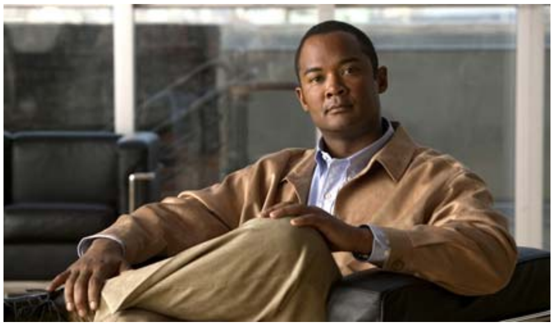
Examples

The following example shows how to set the username and password:

```
Router(config)# voice register pool 1
Router(config-register-pool)# username smith password 9golf
```

Related Commands

Command	Description
authenticate (voice register global)	Enables authentication for registration requests in which the MAC address cannot be identified by using other methods



Cisco Unified CME Commands: V

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This chapter contains commands to configure and maintain Cisco Unified Communications Manager Express (formally known as Cisco Unified CallManager Express). The commands are presented in alphabetical order. Some commands required for configuring Cisco Unified Communications Manager Express (Cisco Unified CME) may be found in other Cisco IOS command references. Use the command reference master index or search online to find these commands.

vad (voice register pool)

To enable voice activity detection (VAD) on a VoIP dial peer, use the **vad** command in voice register pool configuration mode. To disable VAD, use the **no** form of this command.

vad

no vad

Syntax Description

This command has no arguments or keywords.

Command Default

Enabled

Command Modes

Voice register pool configuration

Command History

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

Usage Guidelines

VAD detects periods of silence in the voice signal and temporarily discontinues transmission of the signal during these periods to save bandwidth. Because VAD is enabled by default, there is no comfort noise during periods of silence. As a result, the call may seem to be disconnected and you may prefer to set **no vad** on the SIP phone pool.

Examples

The following example shows how to disable VAD for pool 1:

```
Router(config)# voice register pool 1
Router(config-register-pool)# no vad
```

vad (voice register template)

To enable voice activity detection (VAD) on SIP phones, use the **vad** command in voice register template configuration mode. To return to the default, use the **no** form of this command.

vad

no vad

Syntax Description

This command has no arguments or keywords.

Defaults

Disabled

Command Modes

Voice register template configuration

Command History

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

Usage Guidelines

VAD detects periods of silence in the voice signal and temporarily discontinues transmission of the signal during these periods to save bandwidth. To apply the template to a SIP phone, use the **template** command in voice register pool configuration mode.

Examples

The following example shows how to enable VAD:

```
Router(config)# voice register template 1
Router(config-register-temp)# vad
```

Related Commands

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

video (telephony-service)

To enter video configuration mode for Cisco Unified CallManager Express (Cisco Unified CME), use the **video** command in telephony-service configuration mode. To reset global video parameters, use the **no** form of this command.

video

no video

Syntax Description

This command has no arguments or keywords.

Command Default

Global video parameters are configured.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Use the **video** command to enter telephony-service video configuration mode and set video parameters for all applicable Cisco Unified IP phones associated with a Cisco Unified CME router.

Examples

The following example shows how to enter video configuration mode for a Cisco Unified CME router. You must enter video configuration mode to set video parameters, such as maximum bit rate.

```
Router(config)# telephony-service
Router(config-telephony)# video
Router(config-tele-video)# maximum bit-rate 256
```

Related Commands

Command	Description
service phone	Configures all Cisco IP phones associated with a Cisco Unified CME router.
show call active video	Displays call information for SCCP video calls in progress.
show call history video	Displays call history information for SCCP video calls.
video (call-manager-feedback)	Enters video configuration mode to set video parameters for all applicable Cisco IP phones in a Cisco Unified SRST system.

vm-device-id (ephone)

To define a voice-messaging identification string, use the **vm-device-id** command in ephone configuration mode. To disable this feature, use the **no** form of this command.

vm-device-id *id-string*

no vm-device-id

Syntax Description

id-string	Voice-messaging device port identification (ID) string; for example, CiscoUM-VI1 for the first port and CiscoUM-VI2 for the second port. Note that the first two characters after the hyphen must be the uppercase letters V and I.

Defaults

No voice-mail identification string is defined.

Command Modes

Ephone configuration

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.2(2)XT	2.0	This command was introduced on the Cisco 1750, Cisco 1751, Cisco 2600 series, Cisco 360 series, and Cisco IAD2420 series.
12.2(8)T	2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	2.0	This command was implemented on the Cisco 2600XM and Cisco 2691.
12.2(11)T	2.01	This command was implemented on the Cisco 1760.

Usage Guidelines

Use this command to define a voice-messaging device ID string. A voice-messaging port registers with a device ID instead of a MAC address. To distinguish among different voice-messaging ports, the value of the voice-messaging device ID is used. The voice-messaging device ID is configured to a Cisco IP phone port, which maps to a corresponding voice-messaging port.

Examples

The following example shows how to set the voice-messaging device ID to CiscoUM-VI1:

```
Router(config) ephone 1
Router(config-ephone) vm-device-id CiscoUM-VI1
```

Related Commands

Command	Description
voicemail (telephony-service)	Configures the telephone number that is speed-dialed when the Messages button on a Cisco IP phone is pressed.

vm-integration

To enter voice-mail integration configuration mode and enable voice-mail integration with dual tone multifrequency (DTMF) and analog voice-mail systems, use the **vm-integration** command in global configuration mode. To disable voice-mail integration, use the **no** form of this command.

vm-integration

no vm-integration

Syntax Description

This command has no arguments or keywords.

Command Default

No voice-mail integration is defined.

Command Modes

Global configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(11)YT	Cisco SRST 2.1	This command was introduced for Cisco Survivable Remote Site Telephony (SRST).
12.2(2)XT	Cisco CME 2.0	This command was introduced on the following platforms: Cisco 1750, Cisco 1751, Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(8)T	Cisco CME 2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	Cisco CME 2.0	This command was implemented on the Cisco 2600XM and Cisco 2691.
12.2(11)T	Cisco CME 2.01	This command was integrated into Cisco IOS Release 12.2(11)T and implemented on the Cisco 1760.

Usage Guidelines

The **vm-integration** command is used to enter voice-mail integration configuration mode. Use voice-mail integration configuration mode to integrate a Cisco Unified CallManager Express (Cisco Unified CME) system with an analog voice-mail system.

Examples

The following example shows how to enter the voice-mail integration configuration mode:

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

Related Commands

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

voice hunt-group

To define a hunt group for SIP phones in a Cisco CallManager Express (Cisco CME) system, use the **voice hunt-group** command in global configuration mode. To delete a hunt group, use the **no** form of this command.

voice hunt-group *hunt-tag* [**longest-idle** | **parallel** | **peer** | **sequential**]

no voice hunt-group *hunt-tag*

Syntax Description

hunt-tag	Unique sequence number that identifies the hunt group. Range is 1 to 100
longest idle	Hunt group in which calls go to the directory number that has been idle the longest.
parallel	Hunt group in which calls simultaneously ring multiple phones.
peer	Hunt group in which the first extension to ring is selected round-robin from the list. Ringing proceeds in a circular manner, left to right, for the number of hops specified when the hunt group is defined. The round-robin selection starts with the number left of the number that answered when the hunt-group was last called.
sequential	Hunt group in which extensions ring in the order in which they are listed, left to right, when the hunt group was defined.

Defaults

No default behavior or values

Command Modes

Global configuration

Command History

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

Usage Guidelines

This command enters voice hunt-group configuration mode to define a hunt group. A hunt group is a list of phone numbers that take turns receiving incoming calls for a pilot number. The pilot dial peer contains a hunt-group list. This is the list of destination numbers to try based on a desired selection order. The list of numbers in the hunt group is defined by using the **list** command. If a number in the list is busy or does not answer, the call is redirected to the next number in the list. The last number tried is the final number, which is defined by using the **final** command.

If the number of times that a call is redirected to a new number exceeds 5, the **max-redirect** command must be used to increase the allowable number of redirects in the Cisco CallManager Express system.

To configure a new hunt group, you must specify the **longest-idle**, **peer**, or **sequential** keyword. To change an existing hunt group configuration, the keyword is not required. To change the type of hunt group, for instance from peer to sequential or sequential to peer, you must remove the existing hunt group first by using the **no** form of the command and then recreate it.

The **parallel** keyword creates a dial peer to allow an incoming call to ring multiple phones simultaneously. The use of parallel hunt groups is also referred to as application-level forking because it enables the forking of a call to multiple destinations. A pilot dial peer cannot be used as a voice hunt group and a hunt group at the same time.

Examples

The following example shows how to define a longest-idle hunt group 1 with a pilot number 7501, a final number 8000, and 9 numbers in the list. After a call is redirected six times (makes six hops), it is redirected to the final number 8000.

```
Router(config)# voice hunt-group 1 longest-idle
Router(config-voice-hunt-group)# pilot 7501
Router(config-voice-hunt-group)# list 7001, 7002, 7023, 7028, 7045, 7062, 7067, 7072, 7079
Router(config-voice-hunt-group)# final 8000
Router(config-voice-hunt-group)# hops 6
Router(config-voice-hunt-group)# timeout 20
Router(config-voice-hunt-group)# exit
```

The following example shows how to define a peer hunt group number 2. Callers dial the pilot number 5610 to reach the hunt group. The first extension to ring the first time that this hunt group is called is 5601. If 5601 does not answer, the hunt proceeds from left to right, beginning with the extension directly to the right, for four hops. If none of those extensions answers before the hops limit is reached, the call is forwarded to extension 6000, which is the number for the voice-mail service.

The second time someone calls the hunt group, the first extension to ring is 5602 if 5601 was answered during the previous call.

```
Router(config)# voice hunt-group 2 peer
Router(config-voice-hunt-group)# pilot 5610
Router(config-voice-hunt-group)# list 5601, 5602, 5617, 5633
Router(config-voice-hunt-group)# final 6000
Router(config-voice-hunt-group)# hops 4
Router(config-voice-hunt-group)# timeout 30
Router(config-voice-hunt-group)# exit
```

The following example shows how to define a sequential hunt group number 3. When callers dial extension 5601, the first phone to ring is 5001, then 5002, 5017, and 5028. If none of those extensions answers, the call is forwarded to extension 6000, which is the number for the voice-mail service.

```
Router(config)# voice hunt-group 3 sequential
Router(config-voice-hunt-group)# pilot 5601
Router(config-voice-hunt-group)# list 5001, 5002, 5017, 5028
Router(config-voice-hunt-group)# final 6000
Router(config-voice-hunt-group)# timeout 30
Router(config-voice-hunt-group)# exit
```

The following example shows how to define a voice hunt group. When callers dial extension 1000, extension 1001, 1002, and so forth ring simultaneously. The first extension to answer is connected. All other call legs are disconnected. If none of the extensions answers, the call is forwarded to extension 2000, which is the number for the voice-mail service.

```
Router(config)# voice hunt-group 4 parallel
Router(config-voice-hunt-group)# pilot 1000
Router(config-voice-hunt-group)# list 1001, 1002, 1003, 1004
Router(config-voice-hunt-group)# final 2000
Router(config-voice-hunt-group)# timeout 20
Router(config-voice-hunt-group)# exit
```

Related Commands

Command	Description
final (voice hunt-group)	Defines the last extension in a voice hunt group.
hops (voice hunt-group)	Defines the number of times that a call is redirected to the next directory number in a peer voice hunt-group list before proceeding to the final directory number.
list (voice hunt-group)	Defines the directory numbers that participate in a directory number hunt group.
pilot (voice hunt-group)	Defines the voice dn that callers dial to reach a Cisco CallManager Express (Cisco CME) voice hunt group.
timeout (voice hunt-group)	Sets the number of seconds after which a call that is not answered is redirected to the next number in the hunt-group list and defines the last directory number in the hunt group.

voice logout-profile

To enter voice logout-profile configuration mode for the purpose of creating a logout profile to define the default appearance for a Cisco Unified IP phone enabled for extension mobility, use the **voice logout-profile** command in global configuration mode. To delete an logout profile, use the **no** form of this command.

voice logout-profile *profile-tag*

no voice logout-profile *profile-tag*

Syntax Description

profile-tag	Unique number that identifies this profile during configuration tasks. Range: 1 to maximum number supported phones, where maximum is platform dependent.

Command Default

No logout profile is created.

Command Modes

Global configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XW1	Cisco Unified CME 4.2	This command was introduced.

Usage Guidelines

Use this command to create a logout profile containing a set of commands that define the default appearance for an IP phone that is registered in Cisco Unified CME and enabled for extension mobility, when the IP phone boots and when no phone user is logged into the phone.

Type **?** in voice profile configuration mode to see the commands that are available in this mode and that can be included in a logout profile. The following example shows help for voice logout-profile configuration mode at the time that this document was written:

```
Router(config-logout-profile)#?

Logout profile configuration commands:
 number        Create ip-phone line definition
 pin
 reset         Reset all phones associated with the profile being configured
 speed-dial    Define ip-phone speed-dial number
 username      Create authentication credential for TSP
```

All directory numbers to be included in a default logout profile or voice-user profile must be already configured in Cisco Unified CME.

After creating logout profile, assign the profile to one or more supported Cisco Unified IP phones by using the **logout-profile** command in ephone configuration mode to enable the IP phone for extension mobility.

The same logout profile can be assigned to more than one IP phone to create the appearance of shared lines. All IP phones on which the logout profile is downloaded will have the same directory numbers associated with the same buttons.

You cannot assign more than one logout profile to a particular IP phone. If you assign a second logout profile to a phone to which a logout profile is already applied, the second profile will overwrite the first profile configuration after you use the **reset** command.

After creating or modifying a profile, use the **reset** (voice user-profile) command to reset all phones associated with the profile being configured to propagate changes made to this profile.

Examples

The following example shows the configuration for two logout profiles and the three different IP phones. to which the profiles are assigned. All three phones are enabled for extension mobility. Two of the phones share logout profile 1, while the third phone is assigned logout profile 2. The logout profiles assigned to each phone are downloaded when these phones boot and when no phone user is logged into the phone.

```
voice logout-profile 1
 pin 12345
 user me password pass123
 number 2001 type silent-ring
 number 2002 type beep-ring
 number 2003 type feature-ring
 number 2004 type monitor-ring
 number 2005,2006 type overlay
 number 2007,2008 type cw-overly
 speed-dial 1 3001
 speed-dial 2 3002 blf
!
voice logout-profile 2
 speed-dial 1 9911
 speed-dial 2 2000
!
!
!
ephone 1
 mac-address 000D.EDAB.3566
 type 7960
 logout-profile 1

ephone 2
 mac-address 0012.DA8A.C43D
 type 7970
 logout-profile 1

ephone 3
 mac-address 1200.80FC.9B01
 type 7911
 logout-profile 2
```

Related Commands

Command	Description
logout-profile	Enables Cisco Unified IP phone for extension mobility and assigns a logout profile to this phone.
reset (voice logout-profile and voice user-profile)	Performs a complete reboot of all IP phones to which a particular logout profile or user profile is downloaded.

voice register dialplan

To enter voice register dialplan configuration mode to define a dial plan for SIP phones, use the **voice register dialplan** command in global configuration mode. To remove the dialplan, use the **no** form of this command.

voice register dialplan *dialplan-tag*

no voice register dialplan *dialplan-tag*

Syntax Description

dialplan-tag	Number that identifies the dial plan. Range: 1 to 24.

Command Default

No dial plan is defined.

Command Modes

Global configuration

Command History

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

Usage Guidelines

A dial plan allows a SIP phone to determine when enough digits are collected for call processing to take place. You define a dial plan using this command and then apply the dial plan to a SIP phone by using the **dialplan** command.

Dial plans allow SIP phones to perform pattern recognition as user input is collected. After a defined pattern is recognized, a SIP INVITE message is automatically sent to Cisco Unified CME and the user does not have to press the Dial key or wait for the interdigit timeout.

This command creates a dial plan file that is downloaded to the phone when the phone is reset or restarted.

Examples

The following example shows how to create dial plan 10 for a Cisco Unified IP Phone 7905:

```
Router(config)# voice register dialplan 10
Router(config-register-dialplan)# type 7905-7912
Router(config-register-dialplan)# pattern 52...
Router(config-register-dialplan)# pattern 91.......
```

Related Commands

Command	Description
dialplan	Assigns a dial plan to a SIP phone.
filename	Specifies a custom XML configuration file that contains the dial patterns to use for a SIP dial plan.
pattern (voice register dialplan)	Defines a dial pattern for a SIP dial plan.
show voice register dialplan	Displays all configuration information for a specific SIP dial plan.
type (voice register dialplan)	Defines a phone type for a SIP dial plan.

voice register dn

To enter voice register dn configuration mode to define an extension for a SIP phone line, intercom line, voice-mail port, or a message-waiting indicator (MWI), use the **voice register dn** command in global configuration mode. To remove the directory number, use the **no** form of this command.

voice register dn *dn-tag*

no voice register dn *dn-tag*

Syntax Description

dn-tag	Unique sequence number that identifies a particular directory number during configuration tasks. Range is 1 to 150, or the maximum defined by the **max-dn** command.

Defaults

No default behavior or values

Command Modes

Global configuration

Command History

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

Usage Guidelines

Use this command to create directory numbers in a SIP Cisco CallManager Express (Cisco CME) system. In voice register dn configuration mode, you assign an extension number by using the **number** command, a name to appear in the local directory by using the **name** command, and other provisioning parameters by using various commands.

Before using this command, set the maximum number of directory numbers to appear in your system by using the **max-dn** command in voice register global configuration mode.

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

Examples

The following example shows how to enter voice register dn configuration mode for directory number 4 and forward calls to extension 8888 when extension 1001 does not answer:

```
Router(config)# voice register dn 4
Router(config-register-dn)# number 1001
Router(config-register-dn)# call-forward phone noan 8888
Router(config-register-dn)# call-forward b2bua all 5454
Router(config-register-dn)# call-forward b2bua busy 5705
Router(config-register-dn)# call-forward b2bua mbox 5550
Router(config-register-dn)# call-forward b2bua noan 5050 timeout 20
Router(config-register-dn)# after-hour exempt
```

Related Commands

Command	Description
max-dn (voice register global)	Sets the maximum number of SIP phone directory numbers (extensions) supported by a Cisco CME router.
mode (voice register global)	Enables the mode for provisioning SIP phones in a Cisco CallManager Express (Cisco CME) system.
number (voice register pool)	Configures a valid number for a SIP phone.

voice register global

To enter voice register global configuration mode in order to set global parameters for all supported Cisco SIP phones in a Cisco Unified CallManager Express (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 remove the configuration, use the **no** form of this command.

voice register global

no voice register global

Syntax Description

This command has no arguments or keywords.

Defaults

No default behavior or values

Command Modes

Global configuration

Command History

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

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.

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/Y
```

```
Time-zone is 5
Hold-alert is disabled
Mwi stutter is disabled
Mwi registration for full E.164 is disabled
Dst auto adjust is enabled
 start at Apr week 1 day Sun time 02:00
 stop  at Oct week 8 day Sun time 02:00
```

Related Commands

Command	Description
allow connections sip to sip	Allows connections between SIP endpoints in a Cisco multiservice IP-to-IP gateway.
application (voice register global)	Selects the session-level application for all dial peers associated with SIP phones.
mode (voice register global)	Enables the mode for provisioning SIP phones in a Cisco Unified system.

voice register pool

To enter voice register pool configuration mode for SIP phones, use the **voice register pool** command in global configuration mode. To remove the pool configuration, use the **no** form of this command.

voice register pool *pool-tag*

no voice register pool *pool-tag*

Syntax Description

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

Command Default

No default behavior or values

Command Modes

Global configuration

Command History

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

Usage Guidelines

Cisco Unified CME

Use this command to set phone-specific parameters for SIP phones in a Cisco Unified CME system. Before using this command, enable the **mode cme** command and set the maximum number of SIP phones supported in your system by using the **max-pool** command.

Cisco Unified SIP SRST

Use this command to enable user control on which registrations are to be accepted or rejected by a SIP SRST device. The voice register pool command mode can be used for specialized functions and to restrict registrations on the basis of MAC, IP subnet, and number range parameters.

Examples

Cisco Unified CME

The following example shows how to enter voice register pool configuration mode and forward calls to extension 9999 when extension 2001 is busy:

```
Router(config)# voice register pool 10
Router(config-register-pool)# type 7960
Router(config-register-pool)# number 1 2001
Router(config-register-pool)# call-forward busy 9999 mailbox 1234
```

Cisco Unified SIP SRST

The following partial sample output from the **show running-config** command shows that several voice register pool commands are configured within voice register pool 3:

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

Related Commands

Command	Description
max-pool (voice register global)	Sets the maximum number of SIP phones that are supported by a Cisco Unified CME system.
mode (voice register global)	Enables the mode for provisioning SIP phones in a Cisco Unified CME system.
number (voice register pool)	Configures a valid number for a SIP phone.
type (voice register pool)	Defines a Cisco IP phone type.

voice register session-server

To enter voice register session-server configuration mode to enable and configure a session manager in Cisco Unified CME for an external feature server, use the **voice register session-server** command in global configuration mode. To remove a session manager, use the **no** form of this command.

voice register session-server *session-server-tag*

no voice register session-server *session-server-tag*

Syntax Description

session-server-tag	Explicitly identifies session manager during configuration tasks. Range: 1 to 8.

Command Default

No session manager is created.

Command Modes

Global configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(6th)T	Cisco Unified CME 4.2	This command was introduced.

Usage Guidelines

This command enters voice register session-server configuration mode to configure and enable a session manager for a feature-server, such as Cisco Unified CCX on a Cisco CRS platform.

A single Cisco Unified CME can support multiple session managers.

After creating one or more session managers, use the **session-server** command in voice register pool configuration mode to identify a session manager to be used to control a route point.

After creating one or more session managers, use the **session-server** command in ephone-dn configuration mode to specify which session managers can be used to monitor a directory number.

Note Provisioning and configuration information in Unified CCX is automatically provided to Cisco United CME. Configure this command in Cisco Unified CME only if the configuration from Unified CCX is deleted or must be modified.

Examples

The following partial output from the **show running-configuration** command shows the configuration for session manager, session-server 1:

```
!
voice register session-server 1
 keepalive 300
 register-id SB-SJ3-UCCX1_1164774025000
!
```

Related Commands

Command	Description
session-server	Specifies a session server to manage and monitor registration and subscription messages.

voice register template

To enter voice register template configuration mode and define a template of common parameters for SIP phones, use the **voice register template** command in global configuration mode. To remove a template, use the **no** form of this command.

voice register template *template-tag*

no voice register template *template-tag*

Syntax Description

template-tag	Declares a template tag. Range: 1 to 10.

Defaults

No default behavior or values

Command Modes

Global configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.
12.4(11)XJ	Cisco Unified CME 4.1	The maximum number of templates was increased from 5 to 10.
12.4(15)T	Cisco Unified CME 4.1	The increase in the template number was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

Up to ten different templates can be defined and applied to SIP phones. You create the template with this command and then apply the template to a phone by using the **template** command in voice register pool configuration mode.

Examples

In the following example, template 1 is created by using the **voice register template** command.

```
Router(config)# voice register template 1
Router(config-register-temp)# anonymous block
Router(config-register-temp)# caller-id block
Router(config-register-temp)# voicemail 5001 timeout 15
```

Related Commands

Command	Description
anonymous block (voice register template)	Enables anonymous call blocking in a SIP phone template.
caller-id block (voice register template)	Enables caller-ID blocking for outbound calls from a specific SIP phone.

Command	Description
template (voice register pool)	Applies a template to a SIP phone.
voicemail (voice register template	Defines the extension that calls are forwarded to when an extension does not answer.

voice user-profile

To enter voice user-profile configuration mode and create a user profile to be downloaded by extension mobility for a particular individual phone user who is logged into an IP phone that is registered in Cisco Unified CME and enabled for extension mobility, use the **voice user-profile** command in global configuration mode. To delete an logout profile, use the **no** form of this command.

voice user-profile *profile-tag*

no voice user-profile *profile-tag*

Syntax Description

profile-tag	Unique number that identifies this profile during configuration tasks. Range: 1 to three times maximum number supported phones, where maximum is platform and version dependent and defined by the **max-ephone** command.

Command Default

No user profile is created.

Command Modes

Global configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XW1	Cisco Unified CME 4.2	This command was introduced.

Usage Guidelines

Use this command to create a user profile containing a user's own personal settings, such as directory number, speed-dial lists, and services that will replace the existing configuration of a supported Cisco Unified IP phone that is registered in Cisco Unified CME and enabled for extension mobility, after the individual phone user logs into the phone.

Type **?** in voice user-profile configuration mode to see the commands that are available in this mode and that can be included in a logout profile. The following example shows CLI help for voice user-profile configuration mode at the time that this document was written:

```
Router(config-user-profile)#?

Logout profile configuration commands:
 name          Define username and password for extension mobility.
 number        Create ip-phone line definition
 pin
 reset         Reset all phones associated with the profile being configured
 speed-dial    Define ip-phone speed-dial number
```

All directory numbers to be included in a default logout profile or voice-user profile must be already configured in Cisco Unified CME.

After creating or modifying a profile, use the **reset** (voice user-profile) command to reset all phones associated with the profile being configured to propagate changes made to this profile.

Examples

The following example shows the configuration for a voice-user profile to be downloaded when a phone user logs into a Cisco Unified IP phone that is enabled for extension mobility. Which lines and speed-dial buttons in this profile are configured on a phone after the user logs in depends on phone type. For example, if the user logs into a Cisco Unified IP Phone 7970, all buttons are configured according to voice-user profile1. However, if the phone user logs into a Cisco Unified IP Phone 7960, all six lines are mapped to phone buttons and the speed dial is ignored because there is no button available for speed dial.

```
pin 12345
user me password pass123
number 2001 type silent-ring
number 2002 type beep-ring
number 2003 type feature-ring
number 2004 type monitor-ring
number 2005,2006 type overlay
number 2007,2008 type cw-overly
speed-dial 1 3001
speed-dial 2 3002 blf
```

Related Commands

Command	Description
logout-profile	Enables Cisco Unified IP phone for extension mobility and assigns a logout profile to this phone.
reset (voice logout-profile and voice user-profile)	Performs a complete reboot of all IP phones to which a particular logout profile or user profile is downloaded.

voice-class codec (voice register pool)

To assign a previously configured codec selection preference list, use the **voice-class codec** command in voice register pool configuration mode. To remove the codec preference assignment from the voice register pool, use the no form of this command.

voice-class codec *tag*

no voice-class codec

Syntax Description

tag	Unique number assigned to the voice class. Range is from 1 to 10000. The tag number maps to the tag number created by using the **voice class codec** command in dial-peer configuration mode.

Command Default

None

Command Modes

Voice register pool configuration

Command History

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

Usage Guidelines

During Cisco Unified Session Initiation Protocol (SIP) Survivable Remote Site Telephony (SRST) or Cisco Unified CallManager Express (Cisco Unified CME) registration, a dial peer is created, and that dial peer includes codec g729r8 by default. The **voice-class codec** command allows you to change the automatically selected default codec, if desired.

You can assign one voice class to each voice register pool. If you assign another voice class to a pool, the last voice class assigned replaces the previous voice class.

Note The **id** (voice register pool) command is required and must be configured before any other voice register pool commands. The **id** command identifies a locally available individual Cisco SIP IP phone or set of Cisco SIP IP phones.

Examples

The following partial sample output from the **show running-config** command shows that voice register pool 1 has been set up to use the previously configured codec voice class 1:

```
voice register pool 1
 id mac 0030.94C2.A22A
 preference 5
```

```
cor incoming call91 1 91011
translate-outgoing called 1
proxy 10.2.161.187 preference 1 monitor probe icmp-ping
alias 1 94... to 91011 preference 8
voice-class codec 1
```

Related Commands

Command	Description
codec (voice register pool)	Specifies the codec supported by a single Cisco SIP phone or a VoIP dial peer in a Cisco Unified SIP SRST or a Cisco Unified CME environment.
id (voice register pool)	Explicitly identifies a locally available individual Cisco SIP IP phone, or when running Cisco Unified SIP SRST, set of Cisco SIP IP phones.
voice register pool	Enters voice register pool configuration mode for SIP phones.
voice class codec (dial-peer)	Assigns a previously configured codec selection preference list (codec voice class) to a VoIP dial peer.

voicemail (telephony-service)

To define the telephone number that is speed-dialed when the Messages button on a Cisco IP phone is pressed, use the **voicemail** command in telephony-service configuration mode. To disable the Messages button, use the **no** form of this command.

voicemail *phone-number*

no voicemail

Syntax Description

phone-number	Phone number that is configured as a speed-dial number for retrieving messages.

Defaults

No phone number is configured, and the Messages button is disabled.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.1(5)YD	1.0	This command was introduced on the Cisco 2600 series, Cisco 3600 series, and Cisco IAD2420 series.
12.2(2)XT	2.0	This command was implemented on the Cisco 1750 and Cisco 1751.
12.2(8)T	2.0	This command was integrated into Cisco IOS Release 12.2(8)T and implemented on the Cisco 3725 and Cisco 3745.
12.2(8)T1	2.0	This command was implemented on the Cisco 2600XM and Cisco 2691.
12.2(11)T	2.01	This command was implemented on the Cisco 1760.

Usage Guidelines

This command configures the telephone number that is speed-dialed when the Messages button on a Cisco IP phone is pressed. The same telephone number is configured for voice messaging for all Cisco IP phones connected to the router.

Examples

The following example sets the phone number 914085550100 as the speed-dial number that is dialed to retrieve messages when the Messages button is pressed:

```
Router(config)# telephony-service
Router(config-telephony)# voicemail 914085550100
```

Related Commands

Command	Description
telephony-service	Enters telephony-service configuration mode.
vm-device-id (ephone)	Defines the voice-mail ID string.

voicemail (voice register global)

To define the telephone number that is speed-dialed when the Messages button on a Cisco IP phone is pressed, use the **voicemail** command in voice register global configuration mode. To disable the Messages button, use the **no** form of this command.

voicemail *phone-number*

no voicemail

Syntax Description

phone-number	Telephone number that is speed-dialed for retrieving messages.

Defaults

Disabled

Command Modes

Voice register global configuration

Command History

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

Usage Guidelines

This command configures the telephone number that is speed-dialed when the Messages button on a Cisco IP phone is pressed. The same telephone number is configured for voice messaging for all Cisco IP phones connected to the router.

Examples

The following example shows how to set telephone number 914085550100 as the speed-dial number to retrieve messages when the Messages button is pressed:

```
Router(config)# voice register global
Router(config-register-global)# voicemail 914085550100
```

Related Commands

Command	Description
url (voice register global)	Provision uniform resource locators (URLs) for feature buttons on Cisco IP phones.
voicemail (voice register template)	Defines the extension that calls are forwarded to when an extension does not answer.
voice register global	Enters voice register global configuration mode in order to set global parameters for all supported Cisco SIP phones in a Cisco CME or Cisco SIP SRST environment.

voicemail (voice register template)

To define the extension that calls are forwarded to when an extension does not answer, use the **voicemail** command in voice register template configuration mode. To disable the voicemail extension, use the **no** form of this command.

voicemail *phone-number* **timeout** *timeout*

no voicemail

Syntax Description

phone-number	Telephone number to which calls are forwarded when an extension does not answer.
timeout *seconds*	Duration that a call can ring with no answer before the call is forwarded to the voicemail extension. Range is 5 to 60000. There is no default value.

Defaults

No default behavior or values

Command Modes

Voice register template configuration

Command History

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

Usage Guidelines

This command defines the destination extension for voicemail when an extension on a SIP phone does not answer. To apply the template to a SIP phone, use the **template** command in voice register pool configuration mode.

Examples

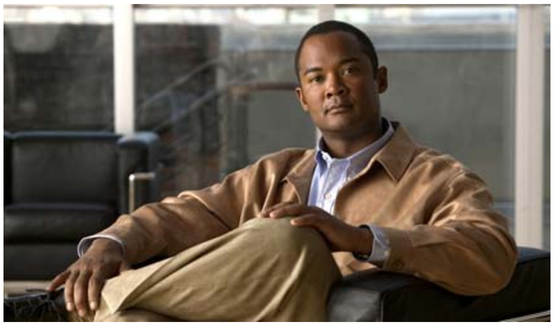
The following example shows how to set telephone number 914085550100 as the number to be dialed to retrieve messages when the Messages button is pressed:

```
Router(config)# voice register template 1
Router(config-register-temp)# voicemail 50100 timeout 15
```

Related Commands

Command	Description
template (voice register pool)	Applies a template to a SIP phone.
url (voice register global)	Provisions uniform resource locators (URLs) for feature buttons on Cisco IP phones.
voice register global	Enters voice register global configuration mode in order to set global parameters for all supported Cisco SIP phones in a Cisco CME or Cisco SIP SRST environment.

Command	Description
voice register template	Enters voice register template configuration mode and defines a template of common parameters for SIP phones.
voicemail (voice register global)	Defines the telephone number that is speed-dialed when the Messages button on a Cisco IP phone is pressed.



Cisco Unified CME Commands: W

Last Updated: January 17, 2007
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This chapter contains commands to configure and maintain Cisco Unified Communications Manager Express (formally known as Cisco Unified CallManager Express). The commands are presented in alphabetical order. Some commands required for configuring Cisco Unified Communications Manager Express (Cisco Unified CME) may be found in other Cisco IOS command references. Use the command reference master index or search online to find these commands.

web admin customer

To define a username and password for a Cisco Unified CME customer administrator, use the **web admin customer** command in telephony-service configuration mode. To disable a customer administrator login, use the **no** form of this command.

web admin customer name *username* {**password** *string* | **secret** {**0** | **5**} *string*}

no web admin customer

Syntax Description

name *username*	Defines the username for the customer administrator. String can contain a maximum of 28 alphanumeric characters. Default is Customer.
password *string*	Defines a character string for login authentication, which will be stored in the running configuration as plain text. String can contain a maximum of 28 alphanumeric characters. Default is no password.
secret {**0** | **5**} *string*	Defines a character string for login authentication, which will be stored in the running configuration as encrypted using Message Digest 5 (MD5). The digit 0 or 5 specifies whether the displayed string that follows is encrypted: • **0**—Password that follows is not encrypted. • **5**—Password that follows is encrypted.

Command Default

A customer administrator with username Customer and no password.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.2(11)YT	2.1	This command was introduced.
12.2(15)T	2.1	This command was integrated into Cisco IOS Release 12.2(15)T.

Usage Guidelines

This command enables customer administrator access for the Cisco Unified CME graphical user interface (GUI).

Examples

The following example defines a customer administrator named user22 whose password is pw567890:

```
Router(config)# telephony-service
Router(config-telephony)# web admin customer name user22 password pw567890
```

Related Commands

Command	Description
telephony-service	Enters telephony-service configuration mode.
web customize load	Loads and parses an XML file in router flash memory to customize a GUI for a customer administrator.

web admin system

To define a username and password for a Cisco Unified CME system administrator, use the **web admin system** command in telephony-service configuration mode. To disable a system administrator login, use the **no** form of this command.

web admin system name *username* {**password** *string* | **secret** {**0** | **5**} *string*}

no web admin system

Syntax Description

name *username*	Defines a login name for the system administrator. String can contain a maximum of 28 alphanumeric characters. Default name is Admin.
password *string*	Defines a character string for login authentication, which will be stored in the running configuration as plain text. String can contain a maximum of 28 alphanumeric characters. Default is no password.
secret {**0** | **5**} *string*	Defines a character string for login authentication, which will be stored in the running configuration as encrypted using Message Digest 5 (MD5). The digit 0 or 5 specifies whether the displayed string that follows is encrypted: • **0**—Password that follows is not encrypted. • **5**—Password that follows is encrypted.

Command Default

A system administrator with username Admin and no password.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.2(11)YT	2.1	This command was introduced.
12.2(15)T	2.1	This command was integrated into Cisco IOS Release 12.2(15)T.

Usage Guidelines

This command enables system administrator access for the Cisco Unified CME graphical user interface (GUI).

Use the **secret 0** keyword pair when entering a plain-text password string. This keyword pair instructs the system to encrypt the system administrator password with MD5. An encrypted version of the string is saved in the running configuration, as shown in the following example. The digit 5 that appears after the **secret** keyword in the running configuration indicates that the password that follows is shown in its encrypted version.

```
web admin system name jsmith secret 5 $1$TCyK$OU/NSQ/VtAU2ibHdi8Uau
```

Examples

The following example establishes a system administrator named user1 whose password will be encrypted in the running configuration:

```
Router(config)# telephony-service
Router(config-telephony)# web admin system name user1 secret 0 pw234567
```

Related Commands

Command	Description
telephony-service	Enters telephony-service configuration mode.

web customize load

To load and parse an eXtensible Markup Language (XML) file in router flash memory to customize a Cisco CallManager Express graphic user interface (GUI) for a customer administrator, use the **web customize load** command in telephony-service configuration mode. To disable the customized GUI and use the system administrator GUI for the customer administrator, use the **no** form of this command.

web customize load *filename*

no web customize load

Syntax Description

filename	Name of the XML file in router flash memory that defines the customer administrator GUI.

Defaults

The standard system administrator GUI is used.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.2(11)YT	2.1	This command was introduced.
12.2(15)T	2.1	This command was integrated into Cisco IOS Release 12.2(15)T.

Usage Guidelines

Use this command with Cisco IOS Telephony Services V2.1, Cisco CallManager Express 3.0, or a later version.

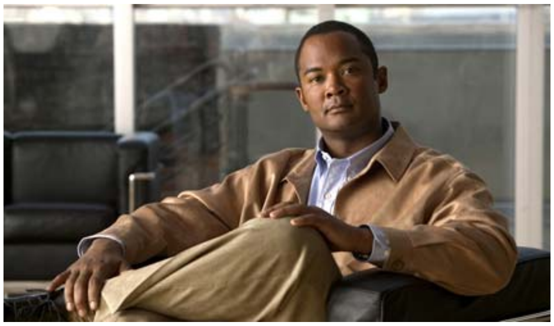
Examples

The following example specifies a file named cust_admin_gui.xml as the file that defines the GUI for Cisco CME customer administrators:

```
Router(config)# telephony-service
Router(config-telephony)# web customize load cust_admin_gui.xml
```

Related Commands

Command	Description
telephony-service	Enters telephony-service configuration mode.



Cisco Unified CME Commands: X

Last Updated: June 19, 2006
First Published: February 27, 2006

This chapter contains commands to configure and maintain Cisco Unified Communications Manager Express (formally known as Cisco Unified CallManager Express). The commands are presented in alphabetical order. Some commands required for configuring Cisco Unified Communications Manager Express (Cisco Unified CME) may be found in other Cisco IOS command references. Use the command reference master index or search online to find these commands.

xmlschema

Effective with Cisco Unified CME 4.0, the **xmlschema** command was made obsolete.

For earlier releases, to specify the URL for a Cisco CME eXtensible Markup Language (XML) application program interface (API) schema, use the **xmlschema** command in telephony-service configuration mode. To set the URL for the XML API schema to the default, use the **no** form of this command.

xmlschema *schema-url*

no xmlschema

Syntax Description

schema-url	Local or remote URL as defined in RFC 2396.

Command Default

srst-its.xsd

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.4(4)XC	Cisco Unified CME 4.0	This command was made obsolete.
12.4(9)T	Cisco Unified CME 4.0	This command was made obsolete in Cisco IOS Release 12.4(9)T.

Examples

The following example specifies a URL for an XML API schema:

```
Router(config)# telephony-service
Router(config-telephony)# xmlschema http://server2.example.com/schema/schema1.xsd
```

Related Commands

Command	Description
telephony-service	Enters telephony-service configuration mode.

xmltest

Effective with Cisco Unified CME 4.0, the **xmltest** command was made obsolete.

For earlier releases, to specify that the HTTP payload in eXtensible Markup Language (XML) application program interface (API) queries be interpreted as having form format, use the **xmltest** command in telephony-service configuration mode. To specify that the HTTP payload should be interpreted as plain text (no form) format, use the **no** form of this command.

xmltest

no xmltest

Syntax Description

This command has no arguments or keywords.

Command Default

Plain text (no form) format

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)Ts	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.4(4)XC	Cisco Unified CME 4.0	This command was made obsolete.
12.4(9)T	Cisco Unified CME 4.0	This command was made obsolete in Cisco IOS Release 12.4(9)T.

Examples

The following example specifies that the HTTP payload in XML API queries be interpreted as having form format:

```
Router(config)# telephony-service
Router(config-telephony)# xmltest
```

Related Commands

Command	Description
telephony-service	Enters telephony-service configuration mode.

xmlthread

Effective with Cisco Unified CME 4.0, the **xmlthread** command was made obsolete.

For earlier releases, to set the maximum number of concurrent Cisco CME eXtensible Markup Language (XML) application program interface (API) queries, use the **xmlthread** command in telephony-service configuration mode. To set the maximum number of queries to the default, use the **no** form of this command.

xmlthread *number*

no xmlthread

Syntax Description

number	Maximum number of XML API queries. Range is from 1 to 5. Default is 2.

Command Default

The maximum number of queries is 2.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.4(4)XC	Cisco Unified CME 4.0	This command was made obsolete.
12.4(9)T	Cisco Unified CME 4.0	This command was made obsolete in Cisco IOS Release 12.4(9)T.

Examples

The following example sets the maximum number of XML API queries to 5:

```
Router(config)# telephony-service
Router(config-telephony)# xmlthread 5
```

xml user

Command	Description
telephony-service	Enters telephony-service configuration mode.

To define a user who is authorized to use XML applications to execute commands, use the **xml user** command in telephony-service configuration mode. To delete the user, use the **no** form of this command.

xml user *user-name* **password** *password privilege-level*

no xml user *user-name* **password** *password privilege-level*

Syntax Description

user-name	User name of the authorized user.
password *password*	Password to use for access.
privilege-level	Level of access to Cisco IOS commands to be granted to this user. Only the commands with the same or a lower level can be executed via XML. Range is 0 to 15.

Command Default

An authorized user is not named.

Command Modes

Telephony-service configuration

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)XC	Cisco Unified CME 4.0	This command was introduced.
12.4(9)T	Cisco Unified CME 4.0	This command was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

You can configure additional levels of access to commands, called privilege levels, to meet the needs of your users while protecting the system from unauthorized access. Up to 16 privilege levels can be configured using the **privilege** command, from level 0, which is the most restricted level, to level 15, which is the least restricted level.

Examples

The following example defines user23 as an authorized user at level 15:

```
Router(config)# telephony-service
Router(config-telephony)# xml user user23 password 3Rs92uzQ 15
```

Related Commands

Command	Description
privilege	Configures a new privilege level for users and associate commands with that privilege level.

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