Cisco Unified CME Commands: T

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.

Command Default

No Cisco Unified CME configuration for SCCP IP phones is present.

Command Modes

Global configuration (config)

Command History

Cisco IOS Release	Cisco Product	Modification
12.1(5)YD	Cisco ITS 1.0	This command was introduced.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T.
12.2(15)ZJ	Cisco CME 3.0	The **setup** keyword was 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 enters the telephony-service configuration mode for configuring system wide parameters for SCCP IP phones in Cisco Unified CME.

Note The voice-gateway system is tied to the telephony service. The **telephony-service** command must be configured before the voice-gateway system is configured; otherwise, the voice gateway is hidden from the user.

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.

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 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.

The table shows a sample dialog with the Cisco CME setup tool and explains possible responses to the Cisco CME setup tool prompts.

Table 1: 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.
`Do you want dual-line extensions assigned to phones? [yes for dual-line / no for single-line]:`	• **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.

Cisco CME Setup Tool Prompt	Description

`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]:`
	Language for IP phone displays, selected from the list.
	• Default is 0, English.

`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]:`
	Locale for the tone set used to indicate call status or progress, selected from the list.
	• Default is 0, United States.

`What is the first extension number you want to configure :[0]:`	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.
`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.

Cisco CME Setup Tool Prompt	Description
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 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
```

telnet-support

To enable the telnet access for the phone, use the **telnet-support** command in voice register pool-type mode. To disable telnet support, use the **no** form of this command.

telnet-support
notelnet-support

Syntax Description

This command has no arguments or keywords.

Command Default

The telnet support is not enable. When the **reference-pooltype** command is configured, the telnet-support value of the reference phone is inherited.

Command Modes

Voice Register Pool Configuration (config-register-pool)`

Command History

Cisco IOS Release	Cisco Product	Modification
15.3(3)M	Cisco SIP CME 10.0	This command was introduced.

Usage Guidelines

Use this command to enable the telnet access for the phone. When you use the no form of this command, the inherited properties of the reference phone takes precedence over the default value.

Example

The following example shows how to specify a description for a phone model using the **description** command:

```
Router(config)# voice register pool-type 9900
Router(config--register-pool-type)# telnet-support
```

Related Commands

Command	Description
voice register pool-type	Adds a new Cisco Unified SIP IP phone to Cisco Unified CME.

template (auto-register)

To create a basic configuration template that supports all the configurations available on the voice register template, use the **template** command in voice auto register configuration mode. This command is a sub-mode CLI of the command **auto-register** . To disable creation of the basic configuration template as part of the auto registration process , use the **no** form of this command.

template *tag*
no template

Syntax Description

template *tag*	Creates a basic configuration template that supports all the configurations available on the voice register template. Range: 1 to 10.

Command Default

By default, this command is disabled.

Command Modes

voice auto register configuration (config-voice-auto-register)

Command History

Cisco IOS Release	Cisco Product	Modification
15.6(3)M 16.3.1	Cisco Unified CME 11.5	This command was introduced.

Usage Guidelines

This command provides the option to create a basic configuration template that can be applied to all phones registering automatically on Unified CME. It is mandatory that voice register template is configured with the same template tag.

Examples

The following example shows how to create a basic configuration template for auto registration of SIP phones:

```
Router(config)#voice register global
Router(config-register-global)#auto-register
Router(config-voice-auto-register)# ?

VOICE auto register configuration commands:
auto-assign Define DN range for auto assignment
default Set a command to its defaults
exit Exit from voice register group configuration mode
no Negate a command or set its defaults
password Default password for auto-register phones
service-enable Enable SIP phone Auto-Registration
template Default template for auto-register phones

Router(config-voice-auto-register)#template ?
 <1-10> template tag
Router(config-voice-auto-register)#template 10
```

Related Commands

Command	Description
service-enable (auto-register)	Temporarily disables the auto registration process, but retains the password and DN range configurations. Once auto-register command is entered, the service is enabled by default.
password (auto-register)	Configures the mandatory password that administrator sets for auto registration of SIP phones on Unified CME.
auto-assign (auto-register)	Configures the mandatory range of directory numbers for phones auto registering on Unified CME.
auto-register	Enables automatic registration of SIP phones with the Cisco Unified CME system.
auto-reg-ephone	Enables automatic registration of ephones with the Cisco Unified CME system.

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.

Command Default

Template is not applied to a SIP IP phone.

Command Modes

Voice register pool configuration (config-register-pool)

Command History

Cisco IOS Release	Cisco Product	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

	Description
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{**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.

Command Default

The default directory is system memory (system:/cme/sipphone/).

Command Modes

Voice register global configuration (config-register-global)

Command History

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

Usage Guidelines

This command defines the location for configuration 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.
reset (voice register global)	Performs a “hard” reboot similar to a power-off-power-on sequence for all SIP phones in Cisco Unified CME, including contacting the Dynamic Host Configuration Protocol (DHCP) server and the TFTP server for updated information.

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 (config-telephony)

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.

Command Default

Time is displayed in 12-hour clock format.

Command Modes

Telephony-service configuration (config-telephony)

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.

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

	Description
date-format	Selects a format to display the date on Cisco IP phones.

time-format (voice register global)

To set the time display format on SIP phones in a Cisco CallManager Express (Cisco CME) system, use the **timeformat** command in voice register global configuration mode. To display the time 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.

Command Default

Time is displayed in 12-hour clock format.

Command Modes

Voice register global configuration (config-register-global)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4	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

	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

Default is the value of the **timeouts ringing** which has a default of 180 seconds if it is not set to another value.

Command Modes

Ephone-hunt configuration (config-ephone-hunt)

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 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 with modifications up 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** for an ephone hunt group, the **max-timeout** 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 is 20 seconds.

```
ephone-hunt 3 peer
 pilot 4200
 list 1001, 1002, 1003
 timeout 7, 9, 15
 max-timeout 20
 final 4500
```

Related Commands

	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 voice 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

Syntax Description

seconds	Number of seconds. Range is 3 to 60000. Default is 180.

Command Default

Timeout period is 180 seconds.

Command Modes

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

Command History

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

Usage Guidelines

If Call Forward No Answer is configured for a directory number in the voice hunt group, set the timeout value of this command to a value that is less than the timeout value of the **call-forward noan** command.

Examples

The following example shows how to define a no-answer timeout of 15 seconds for 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
call-forward noan	Enables call forwarding so that incoming calls to an extension (ephone-dn) that does not answer are forwarded to another number.
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.

Command Default

Timeout busy period is 10 seconds.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(8)T	Cisco ITS 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

	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 Unified 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.

Command Default

Timeout period is 10 seconds.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(2)XB	Cisco ITS 1.0	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

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

	Description
timeouts interdigit (voice-port)	Configures the interdigit timeout value for a specified voice port.

timeouts interdigit (voice register global)

To set the interdigit timeout value for all Cisco SIP phones in a Cisco Unified CME system, use the **timeouts interdigit** command in voice register global 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 SIP phones, in seconds. Range is from 2 to 120. Default is 10.

Command Default

Timeout period is 10 seconds.

Command Modes

Voice register global configuration (config-register-global)

Command History

Cisco IOS Release	Cisco Product	Modification
Cisco IOS XE Everest 16.4.1	Cisco Unified CME 11.6	This command was introduced.

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 SIP phones:

```
Router(config)# voice register global
Router(config-register-global)# timeouts interdigit 5
```

Related Commands

	Description
timeouts interdigit (telephoy-service)	Configures the interdigit timeout value for a SCCP phone in Cisco Unified CME system.

timeouts night-service-bell

To specify the interval between two night-service notification bells, use the **timeouts night-service-bell** command in telephony-service configuration mode. To reset to the default value, use the **no** form of this command.

timeouts night-service-bell *seconds*
no timeouts night-service-bell

Syntax Description

seconds	Duration, in seconds, between night-service notification bells. Range: 4 to 30. Default: 12.

Command Default

Default is 12 seconds.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XW5	Cisco Unified CME 4.2	This command was introduced.

Usage Guidelines

This command modifies the repeat interval between two night-service notification bells for the same call from the default (12 seconds) to the specified number of seconds.

When an ephone-dn is marked for night-service treatment, incoming calls that ring during the night-service time period on that directory number send a notification to all IP phones that are marked to receive night-service bell notification.

Examples

The following partial output shows that the night-service notification bell is configured for 4 seconds between bells for the same call:

```
Router# show running-configuration
.
.
.
telephony-service
.
.
.
 night-service code *1234
 night-service day Tue 00:00 23:00
 night-service day Wed 01:00 23:59
 timeouts night-service-bell 4
!
!
```

Related Commands

Command	Description
night-service bell (ephone)	Marks an IP phone to receive night-service bell notification when incoming calls are received during night-service time periods on ephone-dns that are marked for night service.

Command	Description
night-service bell (ephone-dn)	Marks an ephone-dn to send night-service bell notification to designated IP phones during night-service time periods.

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.

Command Default

Timeout is 180 seconds.

Command Modes

Telephony-service configuration (config-telephony)

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.

Examples

The following example allows incoming calls to ring for 600 seconds:

```
Router(config)# telephony-service
Router(config-telephony)# timeouts ringing 600
```

Related Commands

	Description
telephony-service	Enters telephony-service configuration mode.

timeouts transfer-recall

To enable Cisco Unified CME to recall a transferred call if the transfer-to party does not answer or is busy, use the **timeouts transfer-recall** command in ephone-dn, ephone-dn template, or telephony-service configuration mode. To reset to the default value, use the **no** form of this command.

timeouts transfer-recall *seconds*
no timeouts transfer-recall

Syntax Description

seconds	Duration, in seconds, to wait before recalling a transferred call. Range: 1 to 1800. Default: 0 (disabled).

Command Default

Transfer recall is disabled (0 seconds).

Command Modes

Ephone-dn (config-ephone-dn)
Ephone-dn template (config-ephone-dn-template)
Telephony-service configuration (config-telephony)

Command History

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

Usage Guidelines

This command enables Call Transfer Recall and sets the number of seconds that Cisco Unified CME waits before sending a transferred call back to the phone that initiated the transfer (transferor).

If the transfer-recall timer set with this command expires before the transfer-to party answers a call, the call is directed back to the transferor and the message, “Transfer Recall From xxxx” displays on the transferor phone. After the first recall, the timer restarts. The maximum number of retries is two if the transfer-to party remains busy or does not answer. The transferor and transfer-to party must be on the same Cisco Unified CME router; the transferee party can be remote.

Transfer recall is not supported if the transfer-to party has Call Forward Busy configured or is a member of any hunt group. If the transfer-to directory number has Call Forward No Answer (CFNA) enabled, Cisco Unified CME recalls a transferred call only if the transfer-recall timeout is less than the timeout set with the **call-forward noan** command. If the transfer-recall timeout is set to more than the CFNA timeout, the call is forwarded to the CFNA target number if the transfer-to party does not answer.

If the transferor is busy at the time of the recall, Cisco Unified CME attempts the recall again after the retry timer expires. The maximum number of retries is two. If the transferor phone remains busy, the call is disconnected after the third recall attempt.

Use this command in telephony-service configuration mode to enable the transfer-recall timer at the system level for all directory numbers. Use this command in ephone-dn configuration mode to enable the transfer-recall timer for a particular directory number, or use the command in ephone-dn template mode to apply it to one or more directory numbers.

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 directory number, the value that you set in ephone-dn configuration mode has priority. This command, set in telephony-service configuration mode, has the lowest priority.

Examples

The following example shows that transfer recall is enabled for extension 1030 (ephone-dn 103), which is assigned to ephone 3. If extension 1030 forwards a call and the transfer-to party does not answer, after 60 seconds the unanswered call is sent back to extension 1030 (transferor).

```
ephone-dn  103
 number 1030
 name Smith, John
 timeouts transfer-recall 60
!
ephone  3
 mac-address 002D.264E.54FA
 type 7962
 button  1:103
```

Related Commands

Command	Description
call-forward busy	Enables call forwarding so that incoming calls to a busy extension (ephone-dn) are forwarded to another extension.
call-forward noan	Enables call forwarding so that incoming calls to an extension (ephone-dn) that does not answer are forwarded to another number.
transfer-mode	Specifies the call transfer method for an individual directory number.
transfer-system	Specifies the call transfer method globally for all directory numbers.
trunk	Associates an ephone-dn with a foreign exchange office (FXO) port.

timeouts transfer-recall (voice register global)

To enable Cisco Unified CME to recall a transferred call if the transfer-to party does not answer or is busy, use the **timeouts transfer-recall** command in voice register global configuration mode. To reset to the default value, use the **no** form of this command.

timeouts transfer-recall *seconds*
no timeouts transfer-recall

Syntax Description

seconds	Duration, in seconds, to wait before recalling a transferred call. Range: 1 to 1800. Default: 0 (disabled).

Command Default

Transfer recall is disabled (0 seconds) on a Cisco Unified SIP IP phone.

Command Modes

Voice register global configuration (config-register-global)

Command History

Cisco IOS Release	Cisco Product	Modification
Cisco IOS XE Everest 16.4.1	Cisco Unified CME 11.6	This command was introduced.

Usage Guidelines

This command enables Call Transfer Recall and sets the number of seconds that Cisco Unified CME waits before sending a transferred call back to the phone that initiated the transfer (transferor).

If the transfer-recall timer set with this command expires before the transfer-to party answers a call, the call is directed back to the transferor and the message, “Transfer Recall From xxxx” displays on the transferor phone. If the transferor is busy after the recall, the timer restarts. The maximum number of retries is two if the transfer-to party remains busy or does not answer. The transferor and transfer-to party must be on the same Cisco Unified CME router; the transferee party can be remote.

Transfer recall is not supported if the transfer-to party has Call Forward Busy configured or is a member of any hunt group. The transferor phone and transfer-to phone must be registered to the same Cisco Unified CME, however the transferee phone can be remote. If the transfer-to directory number has Call Forward No Answer (CFNA) enabled, Cisco Unified CME recalls the call only if the transfer-recall timeout is set to less than the CFNA timeout. If the transfer-recall timeout is set to more than the CFNA timeout, the call is forwarded to the CFNA target number after the transfer-to party does not answer. If the transfer-recall timeout is equal to the CFNA timeout, the call is forwarded to the CFNA target number as the CFNA timeout expires before the transfer-recall timeout.

When Call Forward All is configured in Cisco Unified CME, the call is forwarded directly to call forward target number irrespective of whether the phone is busy or idle. In this scenario, transfer recall is not applicable after the call is forwarded.

If the transferor phone is busy, Cisco Unified CME attempts the recall again after the transfer-recall timeout value expires. Cisco Unified CME attempts a recall up to three times. If the transferor phone remains busy, the call is disconnected after the third recall attempt. Also, if the transferor phone is a shared line, and if one of the phones is idle, the transfer recall is directed to the transferor phone that is idle.

Use this command in voice register global configuration mode to enable the transfer-recall timer at the system level for all directory numbers.

The **timeouts transfer-recall** command in voice register global configuration mode has lesser priority than the value that you set in voice register dn configuration mode for the same directory number.

Examples

The following example shows that transfer recall is enabled for 20 seconds. If the transfer-to party does not answer after 20 seconds, the unanswered call is sent back to the (transferor).

```
Router(config)# voice register global
Router(config-register-global)# timeouts transfer-recall 20
```

Related Commands

Command	Description
timeouts transfer-recall(Ephone-dn (config-ephone-dn) and Telephony-service configuration (config-telephony)	Enables Cisco Unified CME to recall a transferred call if the transfer-to party does not answer or is busy.

timeouts transfer-recall (voice register dn)

To enable Cisco Unified CME to recall a transferred call if the transfer-to party does not answer or is busy, use the **timeouts transfer-recall** command in voice register dn configuration mode. To reset to the default value, use the **no** form of this command.

timeouts transfer-recall *seconds*
no timeouts transfer-recall

Syntax Description

seconds	Duration, in seconds, to wait before recalling a transferred call. Range: 1 to 1800. Default: 0 (disabled).

Command Default

Transfer recall is disabled (0 seconds) on a Cisco Unified SIP IP phone.

Command Modes

Voice register dn configuration (config-register-dn)

Command History

Cisco IOS Release	Cisco Product	Modification
Cisco IOS XE Everest 16.4.1	Cisco Unified CME 11.6	This command was introduced.

Usage Guidelines

This command enables Call Transfer Recall and sets the number of seconds that Cisco Unified CME waits before sending a transferred call back to the phone that initiated the transfer (transferor).

If the transfer-recall timer set with this command expires before the transfer-to party answers a call, the call is directed back to the transferor and the message, "Transfer Recall From xxxx" displays on the transferor phone. If the transferor is busy after the recall, the timer restarts. The maximum number of retries is two if the transfer-to party remains busy or does not answer. The transferor and transfer-to party must be on the same Cisco Unified CME router; the transferee party can be remote.

Transfer recall is not supported if the transfer-to party has Call Forward Busy configured or is a member of any hunt group. The transferor phone and transfer-to phone must be registered to the same Cisco Unified CME, however the transferee phone can be remote. If the transfer-to directory number has Call Forward No Answer (CFNA) enabled, Cisco Unified CME recalls the call only if the transfer-recall timeout is set to less than the CFNA timeout. If the transfer-recall timeout is set to more than the CFNA timeout, the call is forwarded to the CFNA target number after the transfer-to party does not answer. If the transfer-recall timeout is equal to the CFNA timeout, the call is forwarded to the CFNA target number as the CFNA timeout expires before the transfer-recall timeout.

When Call Forward All is configured in Cisco Unified CME, the call is forwarded directly to call forward target number irrespective of whether the phone is busy or idle. In this scenario, transfer recall is not applicable after the call is forwarded.

If the transferor phone is busy, Cisco Unified CME attempts the recall again after the transfer-recall timeout value expires. Cisco Unified CME attempts a recall up to three times. If the transferor phone remains busy, the call is disconnected after the third recall attempt. Also, if the transferor phone is a shared line, and if one of the phones is idle, the transfer recall is directed to the transferor phone that is idle.

Use this command in voice register dn configuration mode to enable the transfer-recall timer for a particular directory number.

If you use the **timeouts transfer-recall** command in voice register dn configuration mode for the same directory number, the value that you set in voice register dn configuration mode has priority than the value set in the voice register global configuration mode (this has the lowest priority).

Examples

The following example shows that transfer recall is enabled for extension 111 (voice register dn 1). If extension 111 forwards a call to voice register dn 2 and the transfer-to party does not answer, after 20 seconds the unanswered call is sent back to extension 1111 (transferor).

```
voice register dn  1
 timeouts transfer-recall 20
 number 111
voice register dn  2
 number 222
```

Related Commands

Command	Description
timeouts transfer-recall(Ephone-dn (config-ephone-dn) and Telephony-service configuration (config-telephony)	Enables Cisco Unified CME to recall a transferred call if the transfer-to party does not answer or is busy.

time-webedit (telephony-service)

To enable the system administrator to set time on the Cisco Unified 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.

Command Default

Time-setting through the web interface is disabled.

Command Modes

Telephony-service configuration (config-telephony)

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.
Cisco IOS XE Gibraltar 16.11.1a Release	Unified CME 12.6	The command is deprecated. It is not supported on Unified CME 12.6 and later releases.

Usage Guidelines

The **time-webedit** allows a local administrator of the Cisco Unified 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** 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

	Description
dn-webedit	Enables adding of directory numbers through a web interface.
telephony-service	Enters telephony-service configuration mode.

time-zone

To set the time zone so that the correct local time is displayed on SCCP Cisco Unified IP phones, 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. **Note** The time shows incorrectly for phones configured in West Africa during the Summer Time. For West Africa, Summer Time or Daylight Savings Time (DST) is not used. There is no correct time zone in this time zone list to account for this time zone. • **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)

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) • 54—Venezuela Standard Time (-270) • 55—Pacific SA Daylight Time (-180) • 56—Pacific SA Standard Time (-240)

Command Default

The default is time-zone 5, Pacific Standard/Daylight Time (-480).

Command Modes

Telephony-service configuration (config-telephony)

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 works with the vendorConfig section of the Sep*.cnf.xml configuration file, which is read by the phone firmware when the Cisco IP Phone is booted up. Certain phones, such as the Cisco Unified IP Phone

7906 , 7911, 7931, 7941, 7942, 7945, 7961, 7962, 7965, 7970 , 7971, and 7975, obtain Coordinated Universal Time (UTC) from the clock of the Cisco router. To display the correct local time, the time display on these phones must be offset by using this command.

This command is not required for Cisco Unified IP Phone 7902G, 7905G, 7912G, 7920, 7921, 7935, 7936, 7940, 7960, or 7985G.

For changes to the time-zone settings take effect, the Sep*.cnf.xml file must be updated by using the **create cnf-files** command and the Cisco IP phones must rebooted by using 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*.cnf.xml configuration file.
reset (telephony-service)	Performs a complete reboot of one or all phones associated with a Cisco Unified 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

Command Default

Default is 5, Pacific Standard/Daylight Time.

Command Modes

Voice register global configuration (config-register-global)

Command History

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

Usage Guidelines

The following table lists the supported time zone numbers and the corresponding description.

Table 2: 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

Number	Description	Offset in Minutes
15	SA Western Standard Time	-240
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

Number	Description	Offset in Minutes
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
52	Fiji Standard Time	+720
53	New Zealand Standard/Daylight Time	+720
54	Venezuela Standard Time	-270
55	Pacific SA Daylight Time	-180
56	Pacific SA Standard Time	-240

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

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

Syntax Description

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

Command Default

No maximum length is specified for the transfer number.

Command Modes

Voice register pool configuration (config-register-pool)

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

Command History

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

Usage Guidelines

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

Examples

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

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

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

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

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

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.

Command Default

Soft key is enabled.

Command Modes

Voice register template configuration (config-register-temp)

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 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

	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.

Command Default

Soft key is enabled.

Command Modes

Voice register template configuration (config-register-template)

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

	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-digit-collect

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

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

Syntax Description

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

Command Default

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

Command Modes

Telephony-service configuration (config-telephony)

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

Command History

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

Usage Guidelines

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

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

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

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

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

- The **transfer-system full-consult** command (default) is set in telephony-service configuration mode.
- The **transfer-mode consult** command (default) is set for transferor's directory number (ephone-dn).
- An idle line or channel is available for seizing, digit collection, and dialing.

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

Examples

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

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

Related Commands

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

transfer-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 (config-ephone)

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 system-wide basis by using the **transfer-system** command. The system-wide 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

	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 (config-ephone)
Ephone-template configuration (config-ephone-template)

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 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.

Command Default

Transfer of calls is enabled only to local Cisco IP phones.

Command Modes

Telephony-service configuration (config-telephony)

Command History

Cisco IOS Release	Cisco CME Version	Modification
12.1(5)YD	Cisco ITS 1.0	This command was introduced.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T.
12.2(15)T	Cisco ITS 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.

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

	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 block all call transfers for a specific Cisco Unified SIP IP phone or a set of Cisco Unified SIP IP phone, use the **transfer-pattern blocked** command in voice register pool and voice register template configuration mode. To allow call transfers, use the **no** form of this command.

transfer-pattern blocked
no transfer-pattern blocked

Syntax Description

This command has no arguments or keywords.

Command Default

Call transfers for a specific Cisco Unified SIP IP phone or a set of Cisco Unified SIP IP phone are allowed.

Command Modes

Voice register pool configuration (config-register-pool)

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

Command History

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

Usage Guidelines

When the **transfer-pattern blocked** command is configured for a specific phone, no call transfers are allowed from that phone over the trunk.

This feature forces unconditional blocking of all call transfers from a specific phone to any other non-local numbers (external calls from one trunk to another trunk). No call transfers from this specific phone are possible even when a transfer pattern matches the dialed digits for transfer.

Examples

The following example shows how to block all call transfers for voice register pool 5:

```
Router(config)# voice register pool 5
Router(config-register-pool)# transfer-pattern ?
  blocked  global transfer pattern not allowed
Router(config-register-pool)# transfer-pattern blocked
```

The following example shows how to block all call transfers for a set of Cisco Unified SIP IP phones defined by voice register template 9:

```
Router(config)# voice register template 9
Router(config-register-temp)# transfer-pattern ?
  blocked  global transfer pattern not allowed
Router(config-register-temp)# transfer-pattern blocked
```

Related Commands

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

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

transfer-system

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 (config-telephony)

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. Check the following table for configuration recommendations for different versions of Cisco Unified CME.

Table 3: Transfer Method Recommendations

Cisco Product	transfer-system Default	transfer-system 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

	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 (config-ephone-dn)
Ephone-dn-template configuration (config-ephone-template-dn)

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.
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

	Description
rule	Defines a translation rule.
translation-rule	Creates a translation identifier and enters translation-rule configuration mode.

translate callback-number

To assign a translation profile for incoming or outgoing call legs on a Cisco IP phone, use the **translation-profile** command in call-manager-fallback configuration mode. To delete the translation profile from the voice port, use the **no** form of this command.

translate callback-number
no translate callblack-number

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

Voice translation-profile configuration (cfg-translation-profile)

Command History

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

Usage Guidelines

Use the tranlsate callback-number command to translate a called number to E.164 format. The translated number allows a called or calling number to be presented in its local form. The translate callback-number command is applied when translation-profile is configured on dialpeers, ephone-dn, and voice register-dn. The translate callback-number command is effective when the configuration setup reached the SCCP and SIP IP phones.

Examples

The following example shows a configuration in which a translation profile called name1 is created with two voice translation rules. Rule1 consists of associated calling numbers, and rule2 consists of redirected called numbers. The Cisco IP phones in SRST mode are configured with name1.

```
voice translation-profile name1
 translation calling rule1
 translation called-direct rule2
call-manager-fallback
 translation-profile incoming name1
```

Related Commands

Command	Description
show voice translation-profile	Displays the configuration of a translation profile.
translate (call-manager- fallback)	Applies a translation rule to modify the phone number dialed or received by any Cisco IP phone user during CallManager fallback.
translation-rule	Creates a translation name and enters translation-rule configuration mode to apply rules to the translation name.

Command	Description
voice translation-profile	Defines a translation profile for voice calls.

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

Translation rules are enabled on the VoIP dial peer.

Command Modes

Voice register pool configuration (config-register-pool)

Command History

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

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.

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

	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 (config-ephone-dn)
Ephone-dn-template configuration (config-ephone-dn-template)

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

	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 (config-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

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

	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.

transport (voice register pool-type)

To define the default transport type supported by the new phone, use the **transport** command in voice register pool-type mode. To remove the description, use the **no** form of this command.

Syntax Description

udp	(Optional) Selects UDP as the transport layer protocol. This is the default transport protocol.
tcp	(Optional) Selects TCP as the transport layer protocol.

Command Default

The default transport protocol is UDP. When the reference-pooltype command is configured, the transport value of the reference phone is inherited.

Command Modes

Voice Register Pool-Type Configuration (config-register-pooltype)

Command History

Cisco IOS Release	Cisco Product	Modification
15.3(3)M	Cisco SIP CME 10.0	This command was introduced.

Usage Guidelines

Use this command to define the default transport type. If this parameter is not configured, UDP is used as default value. Currently, except the CiscoMobile-iOS and Jabber-Android, all other phone types uses UDP as default transport type. The default transport type will be ignored when the 'session-transport {udp | tcp}' command is configured for the pool.

Example

The following example shows how to specify a description for a phone model using the description command:

```
Router(config)# voice register pool-type 9900
Router(config-register-pool-type)# transport tcp
```

Related Commands

Command	Description
voice register pool-type	Adds a new Cisco Unified SIP IP phone to Cisco Unified CME.

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 (config-ephone)

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	This command with modifications was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Use this command to configure 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

	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 (config-credentials)

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

	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 (config-capf-server)

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 assign a phone type to an SCCP phone, use the **type** command in ephone or ephone-template configuration mode. To remove a phone type, 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 phone. The following phone types are predefined in the system: • **12SP**—12SP+ and 30VIP phones. • 6901—Cisco Unified IP Phone 6901. • 6911—Cisco Unified IP Phone 6911. • 6921— Cisco Unified IP Phone 6921. • 6941—Cisco Unified IP Phone 6941. • **6945**—Cisco Unified IP Phone 6945. • 6961—Cisco Unified IP Phone 6961. • **7902**—Cisco Unified IP Phone 7902G. • **7905**—Cisco Unified IP Phone 7905G. • **7906**—Cisco Unified IP Phone 7906. • **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 7921. • **7925**—Cisco Unified Wireless IP Phone 7925. • **7931**—Cisco Unified IP Phone 7931G. • **7935**—Cisco Unified IP Conference Station 7935. • **7936**—Cisco Unified IP Conference Station 7936. • **7937**—Cisco Unified IP Conference Station 7937. • **7940**—Cisco Unified IP Phone 7940G. • **7941**—Cisco Unified IP Phone 7941G. • **7941GE**—Cisco Unified IP Phone 7941G-GE.

	• **7942**—Cisco Unified IP Phone 7942. • **7945**—Cisco Unified IP Phone 7945 • **7960**—Cisco Unified IP Phone 7960G. • **7961**—Cisco Unified IP Phone 7961G. • **7961GE**—Cisco Unified IP Phone 7961G-GE. • **7962**—Cisco Unified IP Phone 7962. • **7965**—Cisco Unified IP Phone 7965. • **7970—Cisco Unified IP Phone 7970G.** • **7971**—Cisco Unified IP Phone 7971G-GE. • **7975**—Cisco Unified IP Phone 7975. • **7985**—Cisco Unified IP Phone 7985. • **8941**—Cisco Unified IP Phone 8941. • **8945**—Cisco Unified IP Phone 8945. • **8961**—Cisco Unified IP Phone 8961. • **9951**—Cisco Unified IP Phone 9951. • **9971**—Cisco Unified IP Phone 9971. • **anl**—Analog. • **ata**—Cisco ATA-186 or Cisco ATA-188. • **bri** —SCCP Gateway (BR). • **vgc-phone** —VG248 phone emulation for analog phone. **Note** You can also add a new phone type to your configuration by using the **ephone-type** command.
addon 1 *module-type*	(Optional) Tells the router that an expansion module is being added to this Cisco Unified IP Phone and the type of module. Valid entries for *module-type* are: • **7914**—Cisco Unified IP Phone 7914 Expansion Module. • **7915-12**—Cisco Unified IP Phone 7915 12-Button Expansion Module. • **7915-24**—Cisco Unified IP Phone 7915 24-Button Expansion Module. • **7916-12**—Cisco Unified IP Phone 7916 12-Button Expansion Module. • **7916-24**—Cisco Unified IP Phone 7916 24-Button Expansion Module. **Note** This keyword is not supported for user-defined phone types created with the **ephone-type** command.
2 *module-type*	(Optional) Tells the router that a second expansion module is being added to this Cisco Unified IP Phone and the type of module. Valid entries for *module-type* are: • **7914**—Cisco Unified IP Phone 7914 Expansion Module. • **7915-12**—Cisco Unified IP Phone 7915 12-Button Expansion Module. • **7915-24**—Cisco Unified IP Phone 7915 24-Button Expansion Module. • **7916-12**—Cisco Unified IP Phone 7916 12-Button Expansion Module. • **7916-24**—Cisco Unified IP Phone 7916 24-Button Expansion Module. **Note** This keyword is not supported for user-defined phone types created with the **ephone-type** command.

Command Default

No phone type or add-on expansion module is defined.

Command Modes

Ephone configuration (config-ephone)
Ephone-template configuration (config-ephone-template)

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 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)	This command was integrated into Cisco IOS Release 12.4(11)T.
12.4(11)XJ2	Cisco Unified CME 4.1	The **7921** and **7985** keywords were introduced.
12.4(15)T	Cisco Unified CME 4.1	This command was integrated into Cisco IOS Release 12.4(15)T.
12.4(15)T1	Cisco Unified CME 4.1(1)	The **7942**, **7945**, **7962**, **7965**, and **7975** keywords were introduced.
12.4(15)XZ	Cisco Unified CME 4.3	Support for user-defined phone types created with the **ephone-type** command was added.
12.4(15)XZ1	Cisco Unified CME 4.3	The **7915-12**, **7915-24**, **7916-12**, **7916-24**, and **7937** keywords were added.
12.4(20)T	Cisco Unified CME 7.0	The **7915-12**, **7915-24**, **7916-12**, **7916-24**, and **7937** keywords were added and this command was integrated into Cisco IOS Release 12.4(20)T.

Cisco IOS Release	Cisco Product	Modification
12.4(20)T1	Cisco Unified CME 7.0	The **7925** keyword was added.
15.0(1)XA	Cisco Unified CME 8.0	This command was modified. The 6921, 6941, 6961, and IP-STE keywords were added.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.
15.1(2)T	Cisco Unified CME 8.1	This command was modified. The **6901** and **6911** keywords were added.
15.2(1)T	Cisco Unified CME 8.8	This command was modified. The **6945**, **8941**, and **8945** keywords were added.
15.3(3)M	Cisco Unified CME 10.0	This command was modified. The **7906**, **8961**, **9951**, and **9971** keywords were added.

Usage Guidelines

Not all phone types support add-on expansion modules. For support information, see User Documentation for Cisco Unified IP Phones.

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 two 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
```

The following example defines the IP phone with phone-tag 10 as a Cisco Unified IP Phone IP-STE:

```
Router(config)# ephone 10
Router(config-ephone)# type IPSTE
```

Related Commands

Command	Description
ephone-type	Adds a Cisco Unified IP phone type by defining a phone-type template.
reset (ephone)	Performs a complete reboot of one phone associated with a Cisco Unified CME router.

Command	Description
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, 7942, 7941GE, 7945, 7960, 7960G, 7961, 7961GE, 7962, 7965, 7970, 7971, or 7975.

Command Default

The phone type is not defined.

Command Modes

Voice register dialplan configuration (config-register-dialplan)

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.
12.4(15)XZ	Cisco Unified CME 4.3	Support for Cisco Unified IP Phone 7942, 7945, 7962, 7965, and 7975 was added.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)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* [**addon 1 CKEM** | **CP-8800-Audio** | **CP-8800-Video**[**2 CKEM** | **CP-8800-Audio** | **CP-8800-Video**[**3 CKEM** | **CP-8800-Audio** | **CP-8800-Video**]]]
no type

Syntax Description

phone-type	

	Type of SIP phone that is being defined. Valid entries are as follows: • **3905**—Cisco Unified IP Phone 3905. • **3951**—Cisco Unified IP Phones 3911 and 3951. • **6901**—Cisco Unified IP Phone 6901. • **6911**—Cisco Unified IP Phone 6911. • **6921**—Cisco Unified IP Phone 6921. • **6922**—Cisco Unified IP Phone 6922. • **6941**—Cisco Unified IP Phone 6941. • **6945**—Cisco Unified IP Phone 6945. • **6961**—Cisco Unified IP Phone 6961. • **7821**—Cisco Unified IP Phones 7821. • **7841**—Cisco Unified IP Phones 7841. • **7861**—Cisco Unified IP Phones 7861. • **7905**—Cisco Unified IP Phones 7905 and 7905G. • **7906**—Cisco Unified IP Phone 7906G. • **7911**—Cisco Unified IP Phone 7911G. • **7912**—Cisco IP Phones 7912 and 7912G. • **7940**—Cisco IP Phones 7940 and 7940G. • **7941**—Cisco IP Phone 7941G. • **7941GE**—Cisco IP Phone 7941GE. • **7942**—Cisco Unified IP Phone 7942. • **7945**—Cisco Unified IP Phone 7945. • **7960**—Cisco IP Phones 7960 and 7960G. • **7961**—Cisco IP Phone 7961G. • **7961GE**—Cisco IP Phone 7961GE. • **7962**—Cisco Unified IP Phone 7962. • **7965**—Cisco Unified IP Phone 7965. • **7970**—Cisco IP Phone 7970G. • **7971**—Cisco IP Phone 7971GE. • **7975**—Cisco Unified IP Phone 7975. • **8851**—Cisco Unified IP Phone 8851. • **8851NR**—Cisco Unified IP Phone 8851NR. • **8861**—Cisco Unified IP Phone 8861. • **8865**—Cisco IP Phone 8865. • **8961**—Cisco Unified IP Phone 8961. • **9900**—Cisco Unified IP Phone 9900. • **9951**—Cisco Unified IP Phone 9951. • **9971**—Cisco Unified IP Phone 9971. • **ATA**—Cisco ATA-186 or Cisco ATA-188. • **ATA-187**—Cisco ATA-187. • **ATA-190**—Cisco ATA-190. • **ATA-191**—Cisco ATA-191. • **DX650**—Cisco DX650. • **Jabber-Android**—Cisco Jabber App on Android. • **P100**—PingTel Xpressa 100.

	• **P600**—Polycom SoundPoint 600. • **Jabber-CSF-Client**—Cisco Jabber CSF Client.
addon 1 CKEM	(Optional) Tells the router that a Cisco SIP IP Phone CKEM 36-Button Line Expansion Module is being added to this Cisco Unified SIP IP phone. **Note** This option is available to Cisco Unified 8961, 9951, and 9971 SIP IP phones only.
2 CKEM	(Optional) Tells the router that a second Cisco SIP IP Phone CKEM 36-Button Line Expansion Module is being added to this Cisco Unified SIP IP phone. **Note** This option is available to Cisco Unified 9951 and 9971 SIP IP phones only.
3 CKEM	(Optional) Tells the router that a third Cisco SIP IP Phone CKEM 36-Button Line Expansion Module is being added to this Cisco Unified SIP IP phone. **Note** This option is available to Cisco Unified 9971 SIP IP phones only.
addon 1 CP-8800-Audio or **addon 1 CP-8800-Video**	(Optional) Tells the router that a Cisco SIP IP Phone 28-Button Line A-KEM or V-KEM is being added to this Cisco Unified SIP IP Phone.
2 CP-8800-Audio or **2 CP-8800-Video**	(Optional) Tells the router that a second Cisco SIP IP Phone A-KEM or V-KEM is being added to this Cisco Unified SIP IP Phone.
2 CP-8800-Audio or **2 CP-8800-Video**	(Optional) Tells the router that a second Cisco SIP IP Phone A-KEM or V-KEM is being added to this Cisco Unified SIP IP Phone.

Command Default

No phone type is defined.

Command Modes

Voice register pool configuration (config-register-pool)

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	This command was modified to add the **3951**, **7911**, **7941**, **7941GE**, **7961**, **7961GE**, **7970**, and **7971** keywords.
12.4(15)T	Cisco Unified CME 4.1	The **3951**, **7911**, **7941**, **7941GE**, **7961**, **7961GE**, **7970**, and **7971** keywords were integrated into Cisco IOS Release 12.4(15)T.
12.4(15)XZ	Cisco Unified CME 4.3	This command was modified to add the **7942**, **7945**, **7962**, **7965**, and **7975** keywords.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.
15.1(3)T	Cisco Unified CME8.5	This command was modified to add the **8961**, **9951**,and **9971** keywords.

Cisco IOS Release	Cisco Product	Modification
15.2(1)T	Cisco Unified CME 8.8	This command was modified to add the **3905** keyword.
15.2(2)T	Cisco Unified CME 9.0	This command was modified to add the **6901**, **6911**, **6921**, **6941**, **6945**, **6961**, **ATA-187**, and **Jabber-Android** keywords.
15.2(4)M	Cisco Unified CME 9.1	This command was modified to include the **addon 1 CKEM**, **2 CKEM**, and **3 CKEM** keywords.
15.3(3)M	Cisco Unified CME 10.0	This command was modified to add the **6922** and **9900** keywords.
15.4(3)M	Cisco Unified CME 10.5	This command was modified. The **78XX**, **DX650** and **Jabber-CSF-Client**keywords were added.
Cisco IOS XE Gibraltar 16.10.1a Release	Unified CME 12.5	This command was modified. The **ATA-191**, **CP-8800-Audio**, and **CP-8800-Video**keywords were added.

Usage Guidelines

The **addon 1 CKEM**, **2 CKEM**, and **3 CKEM** keywords increase the number of speed-dial, busy-lamp-field, and directory number keys that can be configured.

There are two options in removing a Key Expansion Module (KEM) when you have configured all three KEMs.

The first option is to use the **no** form of the **type** command, then use the **type** command to configure only the KEMs to be included. The following example shows how the second and third KEMs are removed from the configuration:

```
Router(config)# voice register pool 9
Router(config-register-pool)# type 9971 addon 1 CKEM 2 CKEM 3 CKEM
Router(config-register-pool)# no
 type 9971 addon 1 CKEM 2 CKEM 3 CKEM
Router(config-register-pool)# type 9971 addon 1 CKEM
```

The second option is to define the same phone type while excluding from the configuration the KEM to be removed. For example, you have configured the following:

```
Router(config)# voice register pool 3
Router(config-register-pool)# type 9971 addon 1 CKEM 2 CKEM 3 CKEM
```

To remove the third KEM, enter the following:

Router(config-register-pool)# **type 9971 addon 1 CKEM 2 CKEM**

To remove the second KEM, enter the following:

```
Router(config-register-pool)# type 9971 addon 1 CKEM
```

From Unfiied CME 12.5 Release, **type** *phone-type* [**addon 1 CKEM** | **CP-8800-Audio** | **CP-8800-Video** [**2 CKEM** | **CP-8800-Audio** | **CP-8800-Video** [**3 CKEM** | **CP-8800-Audio** | **CP-8800-Video**]]] configuration is supported. The phone support is extended to Cisco IP Phone 8865 to support the Video KEM **CP-8800-Video**. Audio KEM **CP-8800-Audio** support is introduced for the Cisco IP Phone models 8851, 8851 NR, and 8861.

Note All the configuration characteristics of CKEM discussed here is applicable to **CP-8800-Audio** and **CP-8800-Video**.

After configuring the phone type, use the **create profile** command in voice register global configuration mode to generate the configuration profile files required for the phone and then reset or restart the phone using the **reset** or **restart** command, respectively.

Note Cisco Unified CME enables the **busy trigger-per-button (voice register pool)** command when phone-type **3905** is specified.

Examples

The following example shows how to define a 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
```

The following is a sample configuration for **CP-8800-Audio** and **CP-8800-Video** on the supported phone models for Unified CME 12.5 Release.

```
Router(config-register-pool)# type 8851 addon 1 CP-8800-Audio 2 CP-8800-Audio
Router(config-register-pool)# type 8851NR addon 1 CP-8800-Audio 2 CP-8800-Audio
Router(config-register-pool)# type 8861 addon 1 CP-8800-Audio 2 CP-8800-Audio 3 CP-8800-Audio
Router(config-register-pool)# type 8865 addon 1 CP-8800-Video 2 CP-8800-Video 3 CP-8800-Video
```

Related Commands

Command	Description
busy-trigger-per-button (voice register pool)	Sets the maximum number of calls allowed on a SIP directory number before activating Call Forward Busy or a busy tone.
load (voice register global)	Associates a type of Cisco Unified SIP IP phone with a phone firmware file.
reset (voice register global)	Performs a complete reboot of all SIP phones associated with a Cisco Unified CME router.
reset (voice register pool)	Performs a complete reboot of one SIP phone associated with a Cisco Unified CME router.
restart (voice register)	Performs a fast reset of one or all SIP phones associated with a Cisco Unified CME router.
voice register pool	Enters voice register pool configuration mode for SIP phones.

type (voice-gateway)

To define the type of voice gateway to autoconfigure, use the **type** command in voice-gateway configuration mode. To remove the type from the configuration, use the **no** form of this command.

type {**vg202** | **vg204** | **vg224**}
no type

Syntax Description

vg202	Cisco VG202 Voice Gateway with 2 FXS ports.
vg204	Cisco VG204 Voice Gateway with 4 FXS ports.
vg224	Cisco VG224 Voice Gateway with 24 FXS ports.

Command Default

No type is defined for the voice gateway to be autoconfigured.

Command Modes

Voice-gateway configuration (config-voice-gateway)

Command History

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

Usage Guidelines

This command specifies the type of Cisco voice gateway for which you are creating an XML configuration file.

Examples

The following example shows a configuration for the Cisco VG224 voice gateway:

```
voice-gateway system  1
 network-locale FR
 type VG224
 mac-address 001F.A30F.8331
 voice-port 0-23
 create cnf-files
```

Related Commands

Command	Description
create cnf-files (voice-gateway)	Generates the XML configuration files that are required to autoconfigure the Cisco voice gateway.
mac-address (voice-gateway)	Defines the MAC address of the voice gateway to autoconfigure.
voice-port (voice-gateway)	Identifies the ports on the voice gateway that register to Cisco Unified CME.