Cisco Unified CME Commands: L

label

To create a text identifier instead of a phone-number display for an extension on an IP phone console, use the **label command in** ephone-dn configuration mode. To delete a label, use the **no** form of this command.

label *string*

no label *string*

Syntax Description

string	Alphanumeric string of up to 30 characters.

Command Default

No label is defined.

Command Modes

Ephone-dn configuration (config-ephone)

Command History

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

Usage Guidelines

One label is allowed per extension (ephone-dn). The ephone-dn must already have a number that was set using the **number** command before a label can be created for it.

This command must be followed by a quick reboot of the phone on which the label appears, using the **restart** command.

Examples

The following example creates three phone labels to appear in place of three phone numbers on IP phone console displays:

```
Router(config)# ephone-dn 10
Router(config-ephone-dn)# label user10
Router(config-ephone-dn)# exit
Router(config)# ephone-dn 20
Router(config-ephone-dn)# label user20
Router(config-ephone-dn)# exit
Router(config)# ephone-dn 30
Router(config-ephone-dn)# label user30
Router(config-ephone-dn)# exit
```

Related Commands

Command	Description
number	Associates a telephone or extension number with an ephone-dn in a Cisco CME system.
restart (ephone)	Performs a fast reboot of a single phone associated with a Cisco CME router.
restart (telephony-service)	Performs a fast reboot of one or all phones associated with a Cisco CME router.

label (voice register dn)

To create a text identifier instead of a phone-number display for an extension on a SIP phone console, use the **label command in** voice register dn configuration mode. To delete a label, use the **no** form of this command.

label string

no label string

Syntax Description

string	Alphanumeric string of up to 30 characters.

Command Default

No label is created.

Command Modes

Voice register dn configuration (config-register-dn)

Command History

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

Usage Guidelines

One label is allowed per extension (directory number). The directory number must already have a number that was set by using the **number** command before a label can be created for it.

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

Examples

The following example shows how to create three phone labels to appear in place of three phone numbers on Cisco IP phone console displays:

```
Router(config)#  voice register dn 10
Router(config-register-dn)# label user10
Router(config-register-dn)# exit
Router(config)# voice register dn 20
Router(config-register-dn)# label user20
Router(config-register-dn)# exit
Router(config)# voice register dn 30
Router(config-register-dn)# label user30
Router(config-register-dn)# exit
```

Related Commands

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

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

list (ephone-hunt)

To create a list of extensions that are members of a Cisco Unified CME ephone hunt group, use the **list** command in ephone-hunt configuration mode. To remove a list from the router configuration, use the **no** form of this command.

list*number***[,***number***...]**

no list

Syntax Description

number	Preconfigured extension or E.164 number. An asterisk (*) can take the place of an extension number to represent a wildcard slot. An agent at an authorized ephone-dn can dynamically join and leave a hunt group if a wildcard slot is available. There can be up to 20 wildcard slots in a hunt group.

Command Default

No list is defined.

Command Modes

Ephone-hunt configuration (config-ephone)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(15)ZJ	Cisco CME 3.0	This command was introduced.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.3(11)XL	Cisco CME 3.2.1	The number of ephone-dns allowed was increased to 20.
12.3(14)T	Cisco CME 3.3	This command was integrated into Cisco IOS Release 12.3(14)T.
12.4(4)XC	Cisco Unified CME 4.0	The use of wildcard asterisks (*) in the *dn-number* argument was introduced.
12.4(9)T	Cisco Unified CME 4.0	The use of wildcard asterisks (*) in the *dn-number* argument was integrated into Cisco IOS Release 12.4(9)T.

Usage Guidelines

Use this command to create a list of member numbers for defining a hunt group.

List must contain 1 to 20 numbers.

A number cannot be added to a list unless it was already defined by using the **number** command.

Add or delete all numbers in a hunt group list at one time. You cannot add or single number to an existing list or remove one number from a list.

Any number in the list cannot be a pilot number of a parallel hunt group.

To allow dynamic membership in a hunt group, use asterisks to represent wildcard slots in the **list** command. To allow an ephone-dn to use one of the wildcard slots to dynamically join a hunt group, use the **ephone-hunt login** command under that ephone-dn. Ephone-dns are disallowed from joining hunt groups by default, so you have to explicitly allow this behavior for each ephone-dn that you want to be able to log into hunt groups.

The **show ephone-hunt** command displays the numbers associated to ephone-dns that are joined to groups at the time that the command is run, in addition to static members of the hunt group. Static hunt group members are the numbers that are explicitly named in the **list** command.

Examples

The following example creates sequential hunt group number 7, which contains four static members (ephone-dns):

```
Router(config)# ephone-hunt 7 sequential
Router(config-ephone-hunt)# list 7711, 7712, 7713, 7714
```

The following example creates five ephone-dns and a hunt group that includes the first two ephone-dns as static members and two wildcard slots for dynamic hunt group members. The last three ephone-dns are enabled for dynamic membership in the hunt group. Any of them can join the hunt group whenever one of the wildcard slots is available. Once an ephone-dn has joined a hunt group, it can leave at any time.

```
ephone-dn 22
 number 4566
ephone-dn 23
 number 4567
ephone-dn 24
 number 4568
 ephone-hunt login
ephone-dn 25
 number 4569
 ephone-hunt login
ephone-dn 26
 number 4570
 ephone-hunt login
ephone-hunt 1 peer
 list 4566,4567,*,*
 final 7777
```

Related Commands

Command	Description
ephone-hunt login	Allows an ephone-dn to dynamically join and leave an ephone hunt group.
final	Defines the last ephone-dn in an ephone hunt group.

Command	Description
hops	Defines the number of times that a call is redirected to the next ephone-dn in a peer ephone-hunt-group list before proceeding to the final ephone-dn.
max-redirect	Changes the current number of allowable redirects in a Cisco CME system.
no-reg (ephone-hunt)	Specifies that the pilot number of this ephone hunt group should not register with the H.323 gatekeeper.
number (ephone-dn)	Associates an extension or telephone number with a directory number.
pilot	Defines the ephone-dn that is dialed to reach an ephone hunt group.
preference (ephone-hunt)	Sets preference order for the ephone-dn associated with an ephone-hunt-group pilot number.
show ephone-hunt	Displays ephone-hunt group configuration, current status, and statistics.
timeout (ephone-hunt)	Sets the number of seconds after which a call that is not answered is redirected to the next number in the hunt-group list.

list (voice hunt-group)

To define a list of extensions that are members of a voice hunt-group, use the **list** command in voice hunt-group configuration mode. To remove a list, use the **no** form of this command.

list*number, number***[,***number***...]**

no list

Syntax Description

number	Extension or E.164 number assigned to a phone in Cisco Unified CME. List must contain 2 to 32 numbers.

Command Default

By default, hunt group list is not defined.

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.
12.4(15)XZ	Cisco Unified CME 4.3	The maximum numbers allowed in a list was expanded from 10 to 32 and support for SCCP phones was added.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.
15.4(3)M	Cisco Unified CME 10.5	This command was modified to include support for wildcards which is indicated by "*" . symbol.

Usage Guidelines

This command creates the list of numbers to include in a voice hunt-group. Phones with these numbers ring when the hunt group pilot number is dialed. The numbers must be assigned to directory numbers with the **number** command.

All numbers in a hunt group list are added or deleted at one time. You cannot add a number to an existing list or remove a number from a list.

The pilot number of a parallel hunt group and shared-line numbers are not supported.

A phone number associated with an FXO port is not supported in parallel hunt groups.

Examples

The following example shows how to create a sequential hunt group containing four extensions and a wildcard slot:

```
Router(config)# voice hunt-group 1 sequential
Router(config-voice-hunt-group)# list 7711, 7712, 7713, 7714, *
```

Related Commands

Command	Description
final (voice hunt-group)	Defines the last extension in a voice hunt group.
hops (voice hunt-group)	Defines the number of times that a call is redirected to the next phone number in a peer hunt-group list before proceeding to the final number.
number (ephone-dn)	Associates an extension or telephone number with a directory number.
number (voice register dn)	Associates an extension or telephone number with a directory number.
pilot (voice hunt-group)	Defines the phone number that callers dial to reach a voice hunt group.
timeout (voice hunt-group)	Sets the number of seconds after which a call that is not answered is redirected to the next number in the hunt-group list and defines the last number in the hunt group.

live-record

To define the extension number that is dialed when the LiveRcd soft key is pressed on a Cisco Unified IP Phone, use the **live-record** command in telephony-service configuration mode. To reset to the default value, use the **no** form of this command.

live-record *phone-number*

no live-record

Syntax Description

phone-number	Telephone number that is dialed when the LiveRcd soft key is pressed.

Command Default

Live record is disabled.

Command Modes

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 specifies the telephone number that is speed-dialed to access the Live Record feature when the LiveRcd soft key on a Cisco Unified IP phone is pressed. This telephone number is used for all Cisco Unified IP phones connected to the router.

This telephone number must match the Live Record number configured in Cisco Unity Express.

Examples

The following example shows that the phone number 914085550100 is speed-dialed to record a call when the LiveRcd button is pressed:

```
Router(config)# telephony-service
Router(config-telephony)# live-record 914085550100
```

Related Commands

Command	Description
ephone-template (ephone)	Applies an ephone template to an ephone.

Command	Description
softkeys connected	Modifies the order and type of soft keys that display on an IP phone during the connected call state.
voicemail	Defines the telephone number that is speed-dialed when the Messages button is pressed on an IP phone.

load (telephony-service)

To associate a type of Cisco Unified IP phone with a phone firmware file, use the **load** command in telephony-service configuration mode. To disassociate a type of phone from a phone firmware file, use the **no** form of this command.

load *phone-type firmware-file*

no load *phone-type*

Syntax Description

phone-type	

	Type of phone. The following phone types are predefined in the system: • **6945**—Cisco Unified IP Phone 6945. • **7902**—Cisco Unified IP Phone 7902G. • **7905**—Cisco Unified IP Phone 7905G. • **7910**—Cisco Unified IP Phone 7910 and 7910G. • **7911**—Cisco Unified IP Phone 7911G. • **7912**—Cisco Unified IP Phone 7912G. • **7914**—Cisco Unified IP Phone 7914 Expansion Module. • **7920**—Cisco Unified Wireless IP Phone 7920. • **7921**—Cisco Unified Wireless IP Phone 7921. • **7931**—Cisco Unified IP Phone 7931G. • **7935**—Cisco Unified IP Conference Station 7935. • **7936**—Cisco Unified IP Conference Station 7936. • **7941**—Cisco Unified IP Phone 7941G. • **7941GE**—Cisco Unified IP Phone 7941G-GE. • **7942**—Cisco Unified IP Phone 7942. • **7945**—Cisco Unified IP Phone 7945 • **7960-7940**—Cisco Unified IP Phones 7960 and 7960G and Cisco Unified IP Phone 7940 and 7940G. • **7961**—Cisco Unified IP Phone 7961G. • **7961GE**—Cisco Unified IP Phone 7961G-GE. • **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.

	• **ata—**Cisco ATA-186 and Cisco ATA-188. **Note** You can also add a new phone type to your configuration by using the **ephone-type** command.
firmware-file	Filename of the IP phone firmware for a particular phone type. • In Cisco Unified CME 7.0/4.3 and earlier versions, do not use the file suffix (.bin, .sbin, .loads) for any phone type except the Cisco ATA and Cisco Unified IP Phone 7905 and 7912. • In Cisco Unified CME 7.0(1) and later versions, you must use the complete filename, including the file suffix, for phone firmware versions later than version 8-2-2 for all phone types. • Filenames are case sensitive.

Command Default

Firmware files are not associated with phone types.

Command Modes

Telephony-service configuration (config-telephony)

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(11)YT	Cisco ITS 2.1	Support was added for the Cisco IP Phone 7914 Expansion Module.
12.2(15)T	Cisco ITS 2.1	This command was integrated into Cisco IOS Release 12.2(15)T.
12.2(15)ZJ	Cisco CME 3.0	The following keywords were added to this command: **7902**, **7905**, and **7912**.
12.3(4)T	Cisco CME 3.0	This command was integrated into Cisco IOS Release 12.3(4)T.
12.3(7)T	Cisco CME 3.1	The **7920** and **7936** keywords were added.
12.3(11)XL	Cisco CME 3.2.1	The **7970** keyword was added.

Cisco IOS Release	Cisco Product	Modification
12.3(14)T	Cisco CME 3.3	The **7971** keyword was added, and this command was integrated into Cisco IOS Release 12.3(14)T.
12.4(4)XC	Cisco Unified CME 4.0	The **7911**, **7941**, **7941GE**, **7961**, and **7961GE** keywords were added.
12.4(9)T	Cisco Unified CME 4.0	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(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.
12.4(20)YA	Cisco Unified CME 7.0(1)	Support for automatically creating bindings for firmware files only if the cnf-file location is flash or slot0 was added.
12.4(20)T1	Cisco Unified CME 7.0	The **7925** keyword was introduced.
12.4(22)T	Cisco Unified CME 7.0(1)	This command was integrated into Cisco IOS Release 12.4(22)T.
15.2(1)T	Cisco Unified CME 8.8	This command was modified. The **6945**, **8941**, and **8945** keywords were added.

Usage Guidelines

This command updates the Cisco Unified CME configuration file for the specified type of Cisco Unified IP phone to add the name of the firmware file to be loaded by a particular phone type. The firmware filename also provides the version number for the phone firmware that is in the file. When a phone is started up or rebooted, the phone reads the configuration file to determine which firmware file it must load and then looks for that firmware file on the TFTP server.

If applicable, Cisco Unified IP phones update themselves with new phone firmware whenever they are started up or rebooted.

A separate **load** command is needed for each type of phone. The Cisco Unified IP Phones 7940 and 7940G and the Cisco Unified IP Phones 7960 and 7960G have the same phone firmware and share the **7960-7940** keyword.

Before Cisco Unified CME 7.0(1):

- Do not include the file suffix (.bin, .sbin, .loads) for any phone type except Cisco ATA and Cisco Unified IP Phone 7905 and 7912 when you configure the **load** command in telephony-service configuration mode. For example:

```
Router(config-telephony)# load 7941 SCCP41.8-2-2SR2S
Router(config-telephony)#
```

- You must also configure the **tftp-server** command to enable TFTP access to the firmware files by Cisco Unified IP phones.

In Cisco Unified CME 7.0(1) and later versions:

- When specifying the load command for phone firmware versions later than version 8-2-2 for all phone types and you use the file suffix in the filename, the tftp-server bindings are automatically added for all the files forwarded for that load. For example:

```
Router(config-telephony)# load 7941 SCCP41.8-3-3S.loads
Router(config-telephony)#
```

- The **load** command is enhanced to automatically create TFTP bindings for phone firmware files if the **cnf-file location** command is configured with the **flash** or **slot0** keyword. You are no longer required to configure the **tftp-server** command to create TFTP bindings only if the location of the cnf files is router flash or slot 0 memory. If the **cnf-file location** command is configured for something other than flash or slot 0, such as a TFTP server (url) or system memory (system:its/), you must still configure the **tftp-server** command to create TFTP bindings for phone firmware files. Use the complete filename, including the file suffix, when you configure the **tftp-server** command for phone firmware versions later than version 8-2-2 for all phone types.

To verify TFTP bindings, including the dictionary, language, and tone configuration files that are associated with the ISO-3166 codes that have been selected, use the **show telephony-service tftp-bindings** command.

After associating a firmware file with a Cisco Unified IP phone, use the **reset** command to reboot the phone.

Examples

Examples

The following example shows how to identify the Cisco Unified IP phone firmware file to be used by the Cisco Unified IP Phones 7960 and 7960G and Cisco Unified IP Phone 7910G:

```
Router(config)# telephony-service
Router(config-telephony)# load 7960-7940 P00303020209
Router(config-telephony)# load 7910 P00403020209
Router(config-telephony)# exit
Router(config)# tftp-server flash:P00303020209.bin
Router(config)# tftp-server flash:P00403020209.bin
```

Related Commands

Command	Description
cnf-file location	Specifies a storage location for phone configuration files.
ephone-type	Adds a Cisco Unified IP phone type by defining a phone-type template.
reset	Resets a Cisco Unified IP phone.
show telephony-service tftp-bindings	Provides a list of configuration files that are accessible to IP phones using TFTP.
tftp-server	Enables TFTP access to firmware files on the TFTP server.

load (voice register global)

To associate a type of IP phone with a phone firmware file, use the **load** command in voice register global configuration mode. To disassociate a type of phone from a phone firmware file, use the **no** form of this command.

load *phone-type firmware-file*

no load *phone-type firmware-file*

Syntax Description

phone-type	

	Type of IP phone. The following choices are valid: • **3905—**Cisco Unified IP Phone 3905. • **3951—**Cisco Unified IP Phone 3911 and 3951. • **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. • **7821—**Cisco Unified IP Phone 7821. • **7841—**Cisco Unified IP Phone 7841. • **7861—**Cisco Unified IP Phone 7861. • **7905—**Cisco Unified IP Phone 7905 and 7905G. • **7906—**Cisco Unified IP Phone 7906G. • **7911—**Cisco Unified IP Phone 7911G. • **7912—**Cisco Unified IP Phone 7912 and 7912G. • **7941—**Cisco Unified IP Phone 7941G. • **7941GE—**Cisco Unified IP Phone 7941GE. • **7942—**Cisco Unified IP Phone 7942. • **7945—**Cisco Unified IP Phone 7945. • **7960–7940—**Cisco Unified IP Phones 7940 and 7940G and Cisco IP Phones 7960 and 7960G. • **7961—**Cisco Unified IP Phone 7961G. • **7961GE—**Cisco Unified IP Phone 7961GE. • **7962—**Cisco Unified IP Phone 7962. • **7965—**Cisco Unified IP Phone 7965. • **7970—**Cisco Unified IP Phone 7970G. • **7971—**Cisco Unified IP Phone 7971GE. • **7975—**Cisco Unified IP Phone 7975. • **ATA—**Cisco ATA-186 and Cisco ATA-188. • **ATA-187—**Cisco ATA-187.

	• **DX650—**Cisco DX650.
firmware-file	Filename for the Cisco Unified IP phone firmware to be associated with the IP phone type. Do not use the .bin or .load file extension, except for the Cisco Unified IP phone 7905, 7912, or ATA. Filenames are case sensitive.

Command Default

The firmware file is not associated with the type of phone.

Command Modes

Voice register global configuration (config-register-global)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(4)T	Cisco CME 3.4	This command was introduced.
12.4(11)XJ	Cisco Unified CME 4.1	The **3951**, **7911**, **7941**, **7941GE**, **7961**, **7961GE**, **7970,** and **7971** keywords were added.
12.4(15)T	Cisco Unified CME 4.1	The **3951**, **7911**, **7941**, **7941GE**, **7961**, **7961GE**, **7970**, and **7971** keywords were integrated into Cisco IOS Software Release 12.4(15)T.
12.4(15)XZ	Cisco Unified CME 4.3	The **7942**, **7945**, **7962**, **7965**, and **7975** keywords were added.
12.4(20)T	Cisco Unified CME 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.
15.2(1)T	Cisco Unified CME 8.8	This command was modified. The **3905** keyword was added.
15.2(2)T	Cisco Unified CME 9.0	This command was modified. The **6901**, **6911**, **6921**, **6941**, **6945**, **6961**, and **ATA-187** keywords were added.
15.4(3)M	Cisco Unified CME 10.5	This command was modified to provide support for Cisco Unified 7821, 7841, 7861 and DX650 IP phones.

Usage Guidelines

This command updates the Cisco Unified CME configuration file for the specified type of IP phone to add the name of the correct firmware file that the phone should load. This filename also provides the version number for the phone firmware that is in the file. Later, whenever a phone is started up or rebooted, the phone reads the configuration file to determine the name of the firmware file that it should load and then looks for that firmware file on the TFTP server.

A separate **load** command is needed for each type of phone. The Cisco Unified IP Phone 7940 and 7940G and Cisco Unified IP Phone 7960 and 7960G have the same phone firmware and share the **7960-7940** keyword. The Cisco Unified IP Phone 3911 and Cisco Unified IP Phone 3951 have the same phone firmware and share the **3951** keyword.

For certain IP phones, such as the Cisco Unified IP Phone 7906G, 7911G, 7941G, 7941GE, 7961G, 7961GE, 7970G, and 7971G, there are multiple firmware files. For these phones, use the TERMnn.x-y-x-w.loads or SIPnn.x-y-x-w.loads firmware filename for the **load** command, without the .loads file extension. For these phones, you do not configure the **load** command for any firmware file other than the TERM.loads or SIP.loads firmware file.

Following the **load** command, use the **tftp-server** command to enable TFTP access to the file by Cisco Unified IP phones. The file extension is required when using the **tftp-server** command.

The **load** command must be followed by a reboot of the phones. Plug in a new IP phone or use the **reset** command to reboot an IP phone that is already connected to the Cisco router.

Examples

The following example shows how to configure the **load** command to indicate which phone firmware is to be used by a Cisco Unified IP Phone 7960 and 7960G, a Cisco Unified IP Phone 7912 and 7912G, and a Cisco Unified IP Phone 7941GEs. The **tftp-server** command is used to specify the location of the phone firmware files, including all firmware files for the Java-based Cisco Unified IP Phone 7941GE. Note that while no file extension is used with the **load** command, the file extension is required when using the **tftp-server** command.

```
Router(config)# voice register global
Router(config-register-global)# load 7960-7940 P00303020209
Router(config-register-global)# load 7912 P00403020209
Router(config-register-global)# load 7941 TERM41.7-0-3-0S
Router(config-register-global)# exit
Router(config)# tftp-server flash:P00303020209.bin
Router(config)# tftp-server flash:P00403020209.bin
Router(config)# tftp-server flash:SIP41.8-0-3-0S.loads
Router(config)# tftp-server flash:term61.default.loadsterm
Router(config)# tftp-server flash:41.default.loads
Router(config)# tftp-server flash:CVM41.2-0-2-26.sbn
Router(config)# tftp-server flash:cnu41.2-7-6-26.sbn
Router(config)# tftp-server flash:Jar41.2-9-2-26.sbn
```

Related Commands

Command	Description
reset (voice register global)	Performs a complete reboot of all SIP phones associated with a Cisco Unified CME router.
show voice register global	Displays all global configuration parameters associated with SIP phones.
tftp-server	Enables TFTP access to firmware files on the TFTP server.
type (voice register pool)	Defines a phone type for a SIP phone.

load-cfg-file

To load configuration files on the TFTP server and to sign configuration files that are not created by Cisco Unified CME, use the **load-cfg-file** command in telephony-service configuration mode. To return to the default, use the **no** form of this command.

load-cfg-file *file-url* **alias** *file-alias* **[sign] [create]**

no load-cfg-file *file-url* **alias** *file-alias*

Syntax Description

file-url	Complete path of a configuration file in a local directory.
alias *file-alias*	Name of the file on the TFTP server.
sign	Signs the file and serves it on the TFTP server.
create	Creates the signed file in the local directory.

Command Default

A file is not loaded on the TFTP server.

Command Modes

Telephony-service configuration (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 to sign configuration files that are not created by Cisco Unified CME. This command also loads the signed and unsigned versions of the file on the TFTP server. To simply serve an already signed file on the TFTP server, use this command without the **sign** and **create** keywords.

The **create** keyword should be used with the **sign** keyword the first time that this command is used for each file. The **create** keyword is not maintained in the running configuration; this prevents signed files from being recreated during every reload.

Examples

The following example creates a file called ringlist.xml.sgn in slot0 and serves both ringlist.xml and ringlist.xml.sgn on the TFTP server.

```
telephony-service
 load-cfg-file slot0:Ringlist.xml alias Ringlist.xml sign create
```

The following example serves P00307010200.sbn on the TFTP server without creating a signed file.

```
telephony-service
 load-cfg-file slot0:P00307010200.sbn alias P00307010200.sbn
```

loc2

To specify the audio file used for the loss of C2 features announcement, use the **loc2** command in voice MLPP configuration mode. To disable use of this audio file, use the **no** form of this command.

loc2 *audio-url*

no loc2

Syntax Description

audio-url	Location of the announcement audio file in URL format. Valid storage locations are TFTP, FTP, HTTP, and flash memory.

Command Default

No announcement is played.

Command Modes

Voice MLPP configuration (config-voice-mlpp)

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

This command specifies the G.711 a-law or u-law 8-KHz encoded audio file (.wav or .au format) for the announcement that plays to callers when the call leaves the Cisco Unified CME router on the trunk or when the user places a call to a different domain.

The **mlpp indication** command must be enabled (default) for a phone to play precedence announcements.

This command is not supported by Cisco IOS help. If you type **?**, Cisco IOS help does not display a list of valid entries.

Examples

The following example shows that the audio file played for the isolated code announcement is named ica.au located in flash:

```
Router(config)# voice mlpp
Router(config-voice-mlpp)# loc2 flash:loc2.au
```

Related Commands

Command	Description
bnea	Specifies the audio file used for the busy station not equipped for preemption announcement.
upa	Specifies the audio file used for the unauthorized precedence announcement.
vca	Specifies the audio file used for the vacant code announcement.
mlpp indication	Enables MLPP indication on an SCCP phone or analog FXS port.
mlpp preemption	Enables preemption capability on an SCCP phone or analog FXS port.

location (voice emergency response zone)

To include a location within an emergency response zone, use the **location** command in voice emergency response zone mode. To assign specific priorites to the locations, use the priority tag. To remove the location, use the **no** form of this command.

location *location-tag***[priority <1-100>]**

no location *location-tag*

Syntax Description

location-tag	Identifier for the emergency response zone location.
priority *1-100*	Identifier (1-100) for the priority ranking of locations, 1 being the highest priority.

Command Modes

Voice emergency response zone configuration (cfg-emrgncy-resp-zone)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)XY	Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1) Cisco Unified SIP SRST 4.2(1)	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0 Cisco Unified SRST 7.0 Cisco Unified SIP SRST 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

Use this command to create locations within emergency response zones. The tag must be the same as the tag that is defined using the **voice emergency response location** command. This allows routing of 911 calls to different public safety answering poins (PSAPs). Priority is optional and allows searching locations in a specified priority order. If there are locations with assigned priorities and locations configured without priorities, the prioritized locations are searched before those without an assigned priority.

Examples

The following example shows an assignment of emergency response location (ERLs) to two zones, 10 and 11, to route callers to two different PSAPs. The locations for ERLs in zone 10 are searched in sequential order for a phone address match. The calls from zone 10 have an emergency location identification number (ELIN) from ERLs 8, 9, and 10. The calls from zone 11 have an ELIN from ERLs 2, 3, 4, and 5. The locations for ERLs in zone 11 have priorities assigned and is searched in order of the assigned priority and not the ERL tag number.

```
voice emergency response zone 10
location 8
location 9
location 10
```

```
voice emergency response zone 11
location 5 priority 1
location 3 priority 2
location 4 priority 3
location 2 priority 10
```

Related Commands

Command	Description
emergency response callback	Defines a dial peer that is used for 911 callbacks from the PSAP.
emergency response location	Associates an ERL to either a SIP phone, ephone, or dial peer.
voice emergency response location	Creates a tag for identifying an ERL for the enhanced 911 service.
voice emergency response zone	Creates an emergency response zone within which ERLs can be grouped.

log password

Effective with Cisco Unified CME 4.0, the **log password** command was replaced by the **xml user** command in telephony-service configuration mode. See the **xml user** command for more information.

For Cisco CME 3.4 and earlier versions, to set a local password for an eXtensible Markup Language (XML) Application Programming Interface (API) query, use the **log password** command in telephony-service configuration mode. To remove the password definition, use the **no** form of this command.

log password *password-string*

no log password *password-string*

Syntax Description

password-string	Character string that is a password for XML API queries. Maximum length is 28 characters. Longer strings are truncated.

Command Default

No password is defined.

Command Modes

Telephony-service configuration (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.
12.4(4)XC	Cisco Unified CME 4.0	This command was replaced by the **xml user** command.
12.4(9)T	Cisco Unified CME 4.0	This command was replaced by the **xml user** command.

Usage Guidelines

The local password is used to authenticate XML API requests on the network management server. If the password is not set, an XML API query fails local authentication.

The password string is stored as plain text. No encryption is supported.

Examples

The following example defines a local password for XML API requests:

```
Router(config)# telephony-service
Router(config-telephony)# log password ewvpil
```

log table

To set parameters for the table used to capture phone events used for the eXtensible Markup Language (XML) Application Programming Interface (API), use the **log table** command in telephony-service configuration mode. To reset parameters to their default values, use the **no** form of this command.

log table {**max-size** *entries*| **retain-timer** *minutes*}

no log table {**max-size**| **retain-timer**}

Syntax Description

max-size *entries*	Number of entries in the log table. Range is from 0 to 1000. Default is 150.
retain-timer *minutes*	Number of minutes to retain entries in the log table. Range is from 2 to 500. Default is 15.

Command Default

Default number of entries in table is 150. default number of minutes to retain entries in table is 15.

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.

Usage Guidelines

Cisco Unified CME captures and time-stamps events, such as phones registering and unregistering and extension status, and stores them in an internal buffer. This command sets the maximum number of events, or entries, that can be stored in the table. One event equals one entry. The **retain-timer** keyword sets the number of minutes that events are kept in the buffer before they are deleted.

The event table can be viewed using the **show fb-its-log** command.

Examples

The following example sets the maximum size of the table at 750 events and sets the retention time at 30 minutes:

```
Router(config)# telephony-service
Router(config-telephony)# log table max-size 750
Router(config-telephony)# log table retain-timer 30
```

Related Commands

Command	Description
show fb-its-log	Displays information about the Cisco CME XML API configuration, statistics on XML API queries, and event logs.

logging (voice emergency response settings)

To enable sylog messages to capture emergency call data, use the **logging** command in voice emergency response settings configuration mode. To disable logging, use the **no** form of this command.

logging

no logging

Syntax Description

This command has no arguments or keywords.

Command Default

This command is enabled by default.

Command Modes

Voice emergency response settings configuration (cfg-emrgncy-resp-settings)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(15)XY	Cisco Unified CME 4.2(1) Cisco Unified SRST 4.2(1) Cisco Unified SIP SRST 4.2(1)	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0 Cisco Unified SRST 7.0 Cisco Unified SIP SRST 7.0	This command was integrated into Cisco IOS Release 12.4(20)T.

Usage Guidelines

Use this command to enable syslog messages to be announced for every 911 emergency call that is made. The syslog messages can be used by third party applications to send pager or e-mail notifications to an in-house support number. This optional command is enabled by default.

Examples

In this example, the ELIN (4085550101) defined in the voice emergency response settings configuration is used if the 911 caller's IP phones address does not match any of the voice emergency response locations. After the 911 call is placed to the PSAP, the PSAP has 120 minutes to call back 408 555-0101 to reach the 911 caller. If the call history has expired (after 120 minutes), any callback is routed to extension 7500. The outbound 911 calls do not emit a syslog message to the logging facility (for example, a local buffer, console, or remote host).

```
voice emergency response settings
callback 7500
elin 4085550101
expiry 120
no logging
```

Related Commands

Command	Description
callback	Default phone number to contact if a 911 callback cannot find the last 911 caller from the ERL.
elin	E.164 number used as the default ELIN if no matching ERL to the 911 caller's IP phone address is found.
expiry	Number of minutes a 911 call is associated with an ELIN in case of a callback from the 911 operator.
voice emergency response settings	Creates a tag for identifying settings for E911 behavior.

login (telephony-service)

To define the timer for automatically deactivating user login on SCCP phones in a Cisco Unified CME system, use the **login** command in telephony-service configuration mode. To revert to the default values for automatic logout, use the **no** form of this command.

login [**timeout** [*minutes*]] [**clear** *time*]

no login

Syntax Description

timeout	(Optional) Period of phone idleness after which user login is deactivated.
minutes	(Optional) Number of minutes for which an IP phone can be idle before the user is logged out automatically. Range: 1 to 1440. Default: 60.
clear *time*	(Optional) Time of day after which user login for all IP phones is deactivated. Range: 00:00 to 24:00 on a 24-hour clock. Default: 24:00 (midnight).

Command Default

User login is deactivated after a phone is idle for 60 minutes. User login for all phones is deactivated at 24:00.

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.
12.4(11)XJ2	Cisco Unified CME 4.1	Minimum value for the *minutes* argument was lowered from 5 minutes to 1 minute.
12.4(15)T	Cisco Unified CME 4.1	This command with the modifications was integrated into Cisco IOS Release 12.4(15)T.

Usage Guidelines

This command defines the after-hours login timer. Individual users on specified phones can override call blocking by logging in using a personal identification number (PIN). The after-hours login timer deactivates user login on all phones at a specific time and deactivates a login session automatically after a phone is idle for a specified period of time.

The **login** command applies only to IP phones that have soft keys, such as the Cisco Unified IP Phone 7940 and 7940G and the Cisco Unified IP Phone 7960 and 7960G.

For this command to take effect, fast reboot and reregister all phones in Cisco Unified CME by using the **restart all** command in telephony-service configuration mode.

When a Cisco Unified CME router is rebooted, the login status for all phones is reset to the default.

Examples

The following example sets the after-hours login timer to deactivate logged in phone users automatically after a 2-hour idle time and after 11:30 p.m.

```
Router(config)# telephony-service
Router(config-telephony)# login timeout 120 clear 2330
```

Related Commands

Command	Description
after-hour exempt	Specifies that an IP phone does not have any of its outgoing calls blocked even though call blocking has been defined.
after-hours block pattern	Defines a pattern of digits for blocking outgoing calls from IP phones.
after-hours date	Defines a recurring period based on date during which outgoing calls that match defined block patterns are blocked on IP phones.
after-hours day	Defines a recurring period based on day of the week during which outgoing calls that match defined block patterns are blocked on IP phones.
pin	Sets a global/individual PIN for phone users to deactivate call blocking during call blocking periods.
restart (telephony-service)	Performs a fast reboot of one or all phones associated with a Cisco Unified CME router.
show ephone login	Displays the login states of all phones.

logo (voice register global)

To specify a file to display on SIP phones, use the **logo** command in voice register global configuration mode. To disable the display of the file, use the **no** form of this command.

logo *url*

no logo

Syntax Description

url	URL as defined in RFC 2396.

Command Default

No file is specified for display on idle phones.

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

Use this command to define the URL for the file to be used by SIP phones connected in Cisco Unified CME. The file that is displayed must be encoded in eXtensible Markup Language (XML) by using the Cisco XML document type definition (DTD). For more information about Cisco DTD formats, see the *Cisco IP Phone Services Application Development Notes.*

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

Examples

The following example shows how to specify that the file logo.xml should be displayed on SIP phones:

```
Router(config)# voice register global
Router(config-register-global)# logo http://mycompany.com/files/logo.xml
```

Related Commands

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

logout-profile

To enable an IP phone for extension mobility and to apply a default logout profile to the phone, use the **logout-profile** command in ephone configuration mode. To disable extension mobility, use the **no** form of this command.

logout-profile *profile-tag*

no logout-profile *profile-tag*

Syntax Description

profile-tag	Unique identifier for a default logout profile to be applied. Previously created by using the **voice logout-profile** command in voice logout-profile configuration mode. Range: 1 to maximum number of phones supported by platform.

Command Default

IP phone is not enabled for extension mobility.

Command Modes

Ephone configuration (config-ephone), Voice Register Pool configuration (voice register pool)

Command History

Cisco IOS Release	Cisco Product	Modification
12.4(11)XW	Cisco Unified CME 4.2	This command was introduced.
12.4(15)XY	Cisco Unified CME 4.2(1)	This command was introduced.
12.4(15)XZ	Cisco Unified CME 4.3	This command was introduced.
12.4(20)T	Cisco Unified CME 7.0	This command is integrated into Cisco IOS Release 12.4(20)T.
15.1(4)M	Cisco Unified CME 8.6	This command was introduced.

Usage Guidelines

Use this command in ephone configuration mode to enable a supported IP phone registered in Cisco Unified CME for extension mobility and to apply a default logout profile to the ephone being configured.

In Cisco Unified CME 4.2, extension mobility is supported only on SCCP IP phones.

In Cisco Unified CME 8.6 extension mobility is supported on SIP phones.

Extension mobility is not supported on non-display IP phones.

Extension mobility is not supported for analog devices.

Before using this command, you must create a logout profile to be applied to this phone by using the **voice logout-profile** command.

You cannot apply more than one logout profile to an ephone. If you attempt to apply a second logout profile to an ephone to which a profile has already been applied, the second profile will overwrite the first logout profile configuration.

Examples

The following example shows the ephone configuration for three different Cisco Unified IP phones. All three phones are enabled for extension mobility and share the same logout profile number (1), to be downloaded when these phones boot and when no phone user is logged into these phones:

```
ephone 1
 mac-address 000D.EDAB.3566
 type 7960
 logout-profile 1
ephone 2
 mac-address 0012.DA8A.C43D
 type 7970
 logout-profile 1
ephone 3
 mac-address 1200.80FC.9B01
 type 7911
 logout-profile 1
```

The following example shows the ephone configuration for two different Cisco Unified IP phones. Both phones are enabled for extension mobility and share the same logout profile number (22), to be downloaded when these phones boot and when no phone user is logged into these phones:

```
voice register pool  1
 logout-profile 22
 id mac 0012.0034.0056
 type 7960
voice register pool  2
 logout-profile 22
 id mac 0001.0023.0045
 type 7912
```

Related Commands

Command	Description
reset (voice logout-profile and voice user-profile)	Performs a complete reboot of all IP phones to which a particular logout-profile or user-profile is downloaded.
voice logout-profile	Enters voice profile configuration mode to configure a default logout profile for extension mobility.

loopback-dn

To create a virtual loopback voice port (loopback-dn) to establish a demarcation point for VoIP calls and supplementary services, use the **loopback-dn** command in ephone-dn configuration mode. To delete a loopback-dn configuration, use the **no** form of this command.

loopback-dn *dn-tag* [**forward** *number-of-digits*| **strip** *number-of-digits*] [**prefix** *prefix-digit-string*] [**suffix** *suffix-digit-string*] [**retry** *seconds*] **[auto-con]** [**codec** {**g711alaw**| **g711ulaw**}]

no loopback-dn

Syntax Description

dn-tag	Unique sequence number that identifies the ephone-dn that is being paired for loopback with the ephone-dn that is currently being configured. The paired ephone-dn must be one that is already defined in the system.
forward *number-of-digits*	(Optional) Number of digits in the original called number to forward to the other ephone-dn in the loopback-dn pair. Range is from 1 to 32 digits. Default is to forward all digits.
strip *number-of-digits*	(Optional) Number of leading digits to be stripped from the original called number before forwarding to the other ephone-dn in the loopback-dn pair. Range is from 1 to 32 digits. Default is not to A-law strip any digits.
prefix *prefix-digit-string*	(Optional) Defines a string of digits to add in front of the forwarded called number. Maximum number of digits in the string is 32. Default is that no prefix is defined.
suffix *suffix-digit-string*	(Optional) Defines a string of digits to add to the end of the forwarded called number. Maximum number of digits in the string is 32. Default is that no suffix is defined. If you add a suffix that starts with the pound character (#), the string must be enclosed in quotation marks.
retry *seconds*	(Optional) Number of seconds to wait before retrying the loopback target when it is busy or unavailable. Range is from 0 to 32767. Default is that retry is disabled and appropriate call-progress tones are passed to the call originator.

auto-con	(Optional) Immediately connects the call and provides in-band alerting while waiting for the far-end destination to answer. Default is that automatic connection is disabled.
codec	(Optional) Explicitly forces the G.711 A-law or G.711 mu-law voice coding type to be used for calls that pass through the loopback-dn. This overrides the G.711 coding type that is negotiated for the call and provides mu-law to A-law conversion if needed. Default is that Real-Time Transport Protocol (RTP) voice packets are passed through the loopback-dn without considering the G.711 coding type negotiated for the calls.
g711alaw	G.711 A-law, 64000 bits per second, for T1.
g711ulaw	G.711 mu-law, 64000 bits per second, for E1.

Command Default

All calls are set to forward all digits and not to strip any digits. Prefix is not defined. Suffix is not defined. Retry is disabled. Automatic connection is disabled. RTP voice packets are passed through the loopback-dn without considering the G.711 coding type negotiated for the call.

Command Modes

Ephone-dn configuration (config-ephone-dn)

Command History

Cisco IOS Release	Cisco Product	Modification
12.2(2)XT	Cisco ITS 2.0	This command was introduced.
12.2(2)XT3	Cisco ITS 2.0	The **suffix** keyword was added.
12.2(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T and the **auto-con** keyword was added.
12.2(11)T	Cisco ITS 2.01	The **suffix** keyword was added**.**
12.2(11)YT	Cisco ITS 2.1	The **strip** keyword was added.
12.2(11)YT2	Cisco ITS 2.1	The **codec** keyword was added.
12.2(15)T	Cisco ITS 2.1	This command was integrated into Cisco IOS Release 12.2(15)T.

Usage Guidelines

The **loopback-dn** command is used to configure two ephone-dn virtual voice ports as back-to-back-connected voice-port pairs. A call presented on one side of the loopback-dn pair is reoriginated as a new call on the opposite side of the loopback-dn pair. The **forward**, **strip, prefix**, and **suffix** keywords can be used to manipulate the original called number that is presented to the incoming side of the loopback-dn pair to generate a modified called number to use when reoriginating the call at the opposite side of the loopback-dn pair. For loopback-dn configurations, you must always configure ephone-dn virtual voice ports as cross-coupled pairs.

Note Use of loopback-dn configurations within a VoIP network should be restricted to resolving critical network interoperability service problems that cannot otherwise be solved. Loopback-dn configurations are intended to be used in VoIP network interworking situations in which the only other alternative would be to make use of back-to-back-connected physical voice ports. Loopback-dn configurations emulate the effect of a back-to-back physical voice-port arrangement without the expense of the physical voice-port hardware. A disadvantage of loopback-dn configurations is that, because digital signal processors (DSPs) are not involved in a loopback-dn arrangement, the configuration does not support interworking or transcoding between calls that use different voice codecs. In many cases, the use of back-to-back physical voice ports that do use DSPs to resolve VoIP network interworking issues is preferred, because it introduces fewer restrictions in terms of supported codecs and call flows. Also, loopback-dns do not support T.38 fax relay.

Note We recommend that you create the basic ephone-dn configuration for both ephone-dn entries before configuring the loopback-dn option under each ephone-dn. The loopback-dn mechanism should be used only in situations where the voice call parameters for the calls on either side of the loopback-dn use compatible configurations; for example, compatible voice codec and dual tone multifrequency (DTMF) relay parameters. Loopback-dn configurations should be used only for G.711 voice calls.

The loopback-dn arrangement allows an incoming telephone call to be terminated on one side of the loopback-dn port pair and a new pass-through outgoing call to be originated on the other side of the loopback-dn port pair. The loopback-dn port pair normally works with direct cross-coupling of their call states; the alerting call state on the outbound call segment is associated with the ringing state on the inbound call segment.

The loopback-dn mechanism allows for call operations (such as call transfer and call forward) that are invoked for the call segment on one side of the loopback-dn port pair to be isolated from the call segment that is present on the opposite side of the loopback-dn port pair. This approach is useful when the endpoint devices associated with the two different sides have mismatched call-transfer and call-forwarding capabilities. The loopback-dn arrangement allows for call-transfer and call-forward requests to be serviced on one side of the loopback-dn port pair by creating hairpin-routed calls when necessary. The loopback-dn arrangement avoids the propagation of call-transfer and call-forward requests to endpoint devices that do not support these functions.

The **loopback-dn** command provides options for controlling the called-number digits that are passed through from the incoming side to the outgoing side. The available digits can be manipulated with the **forward**, **strip**, **prefix**, and **suffix** keywords.

The **forward** keyword defines the number of digits in the original called number to forward to the other ephone-dn in the loopback-dn pair. The default is set to forward all digits. The **strip** keyword defines the number of leading digits to be stripped from the original called number before forwarding to the other ephone-dn in the loopback-dn pair. The default is set to not strip any digits. The **forward** and **strip** commands are mutually exclusive and can be used with any combination of the **prefix** and **suffix** keywords.

The **prefix** keyword defines a string of digits to add in front of the forwarded number.

The **suffix** keyword is most commonly used to add a terminating "#" (pound-sign) character to the end of the forwarded number to indicate that no more digits should be expected. The pound-sign character indicates to the call-routing mechanism that is processing the forwarded number that the forwarded number is complete. Providing an explicit end-of-number character also avoids a situation in which the call-processing mechanism waits for the interdigit timeout period to expire before routing the call onward using the forwarded number.

Note The Cisco IOS command-line interface (CLI) requires that arguments with character strings that start with the pound-sign (#) character be enclosed within quotation marks; for example, "#".

The **retry** keyword is used to suppress a far-end busy indication on the outbound call segment. Instead of returning a busy signal to the call originator (on the incoming call segment), a loopback-dn presents an alerting or ringing tone to the caller and then periodically retries the call to the final far-end destination (on the outgoing call segment). This is not bidirectional. To prevent calls from being routed into the idle outgoing side of the loopback-dn port pair during the idle interval that occurs between successive outgoing call attempts, configure the outgoing side of the loopback-dn without a number so that there is no number to match for the inbound call.

The **auto-con** keyword is used to configure a premature trigger for a connected state for an incoming call segment while the outgoing call segment is still in the alerting state. This setup forces the voice path to open for the incoming call segment and support the generation of in-band call progress tones for busy, alerting, or ringback. The disadvantage of the **auto-con** keyword is premature opening of the voice path during the alerting stage and also triggering of the beginning of billing for the call before the call has been answered by the far end. These disadvantages should be considered carefully before you use the **auto-con** keyword.

The **codec** keyword is used to explicitly select the A-law or mu-law type of G.711 and to provide A-law to mu-law conversion if needed. Setting the codec type on one side of the loopback-dn forces the selection of A-law or mu-law for voice packets that are transmitted from that side of the loopback-dn. To force the A-law or mu-law G.711 codec type for both voice packet directions, set the codec type on both sides of the loopback-dn. Loopback-dn configurations are used only with G.711 calls. Other voice codec types are not supported.

Examples

The following example creates a loopback-dn configured with the **forward** and **prefix** keywords:

```
Router(config)# ephone-dn 7
Router(config-ephone-dn)# loopback-dn 15 forward 5 prefix 41
```

The following example creates a loopback-dn that appends the pound-sign (#) character to forwarded numbers to indicate the end of the numbers:

```
Router(config)# ephone-dn 7
Router(config-ephone-dn)# loopback-dn 16 suffix "#"
```

The following example shows a loopback-dn configuration that pairs ephone-dns 15 and 16. An incoming call (for example, from VoIP) to 4085550101 matches ephone-dn 16. The call is then reoriginated from ephone-dn 15 and sent to extension 50101. Another incoming call (for example, from a local IP phone) to extension 50151 matches ephone-dn 15. It is reoriginated from ephone-dn 16 and sent to 4085550151.

```
ephone-dn 15
 number 5015.
 loopback-dn 16 forward 5 prefix 40855
 caller-id local
 no huntstop
!
ephone-dn 16
 number 408555010.
 loopback-dn 15 forward 5
```

```
caller-id local
no huntstop
```

Related Commands

Command	Description
ephone-dn	Enters ephone-dn configuration mode.
show ephone-dn loopback	Displays information about loopback ephone-dns that have been created in a Cisco CME system.

lpcor incoming

To associate an incoming call with a logical partitioning class of restriction (LPCOR) resource-group policy, use the **lpcor incoming** command in ephone, ephone-template, trunk group, voice port, voice register pool, voice register template, or voice service configuration mode. To reset to the default, use the **no** form of this command.

lpcor incoming *lpcor-group*

no lpcor incoming

Syntax Description

lpcor-group	Name of the LPCOR resource group.

Command Default

LPCOR policy is not associated with the incoming call.

Command Modes

Ephone configuration (config-ephone) Ephone template configuration (config-ephone-template) Trunk group configuration (config-trunk-group) Voice port configuration (config-voiceport) Voice register pool configuration (config-register-pool) Voice register template configuration (config-register-temp) Voice service configuration (conf-voi-serv)

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

Analog Phones

An incoming call to an analog phone is associated with the LPCOR policy specified with this command in the voice port. Otherwise the LPCOR policy specified in the trunk group is used.

SCCP IP Phones (Local or Remote)

An incoming call to an SCCP IP phone is associated with the LPCOR policy specified with this command in ephone or ephone template configuration mode. The ephone configuration has precedence over the ephone-template configuration. All directory numbers on the phone share the same LPCOR setting.

SIP IP Phones (Local or Remote)

An incoming call to a SIP IP phone is associated with the LPCOR policy specified with this command in voice register pool or voice register template configuration mode. The voice register pool configuration has precedence over the voice register template configuration. All directory numbers on the phone share the same LPCOR setting.

Note This command is not supported for phones configured with the **lpcor type mobility** command.

- Phones that share a directory number must be configured with the same LPCOR policy. Different LPCOR settings on shared-line phones are not supported.

PSTN Trunks

An incoming call to the PSTN is associated with the LPCOR policy specified with this command in the voice port. Otherwise the LPCOR policy specified in the trunk group is used. The voice port configuration takes precedence.

VoIP Trunks (H.323 or SIP)

An incoming call to a VoIP trunk is associated with the LPCOR policy specified with this command in voice service configuration mode if the remote IP address is not found in the IP trunk subnet table created with the **voice lpcor ip-trunk subnet incoming** command.

Examples

The following example shows the command used in different configuration modes:

```
voice service voip
 lpcor incoming voip_group1
!
trunk group analog1
 lpcor incoming analog_group1
 lpcor outgoing analog_group1
!
voice-port 1/1/0
 lpcor incoming vp_group1
 lpcor outgoing vp_group1
!
voice register pool  3
 lpcor type remote
 lpcor incoming sip_group3
 lpcor outgoing sip_group3
 id mac 001E.BE8F.96C0
 type 7940
 number 1 dn 3
!
ephone 2
 mac-address 001C.821C.ED23
 type 7960
 button 1:2
 lpcor type remote
 lpcor incoming ephone_group2
 lpcor outgoing ephone_group2
```

Related Commands

Command	Description
lpcor outgoing	Associates an outgoing call with a LPCOR resource-group policy.
lpcor type	Specifies the LPCOR type for an IP phone.
voice lpcor ip-trunk subnet incoming	Creates a LPCOR IP-trunk subnet table for incoming calls from a VoIP trunk.

Command	Description
voice lpcor policy	Creates a LPCOR policy for a resource group.

lpcor outgoing

To associate an outgoing call with a logical partitioning class of restriction (LPCOR) resource-group policy, use the **lpcor outgoing** command in dial peer, ephone, ephone template, trunk group, voice port, voice register pool, or voice register template configuration mode. To reset to the default, use the **no** form of this command.

lpcor outgoing *lpcor-group*

no lpcor outgoing

Syntax Description

lpcor-group	Name of the LPCOR resource group.

Command Default

LPCOR policy is not associated with the outgoing call.

Command Modes

Dial peer configuration (config-dial-peer) Ephone configuration (config-ephone) Ephone template configuration (config-ephone-template) Trunk group configuration (config-trunk-group) Voice port configuration (config-voiceport) Voice register pool configuration (config-register-pool) Voice register template configuration (config-register-temp)

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

Analog Phones

An outgoing call from an analog phone is associated with the LPCOR policy specified with this command in the voice port. Otherwise the LPCOR policy specified in the trunk group is used.

SCCP IP Phones (Local or Remote)

An outgoing call from an SCCP IP phone is associated with the LPCOR policy specified with this command in ephone configuration or ephone template configuration mode. The ephone configuration has precedence over the ephone-template configuration. All directory numbers on the phone share the same LPCOR setting.

SIP IP Phones (Local or Remote)

An outgoing call from a SIP IP phone is associated with the LPCOR policy specified with this command in voice register pool or voice register template configuration mode. The voice register pool configuration has precedence over the voice register template configuration. All directory numbers on the phone share the same LPCOR setting.

Note This command is not supported for phones configured with the **lpcor type mobility** command.

- Phones that share a directory number must be configured with the same LPCOR policy. Different LPCOR settings on shared-line phones are not supported.

PSTN Trunks

An outgoing call from the PSTN uses the LPCOR policy specified with this command in the voice port if the outbound dial peer is configured with the **port** command. Otherwise the outgoing call uses the LPCOR policy specified with this command in the trunk group if the outbound dial peer is configured with the **trunkgroup** command.

VoIP Trunks (H.323 or SIP)

An outgoing VoIP call uses the LPCOR policy specified with this command in the outbound dial peer. Otherwise the outgoing call uses the default LPCOR policy.

Examples

The following example shows the command used in different configuration modes:

```
trunk group analog1
 lpcor incoming analog_group1
 lpcor outgoing analog_group1
!
voice-port 1/1/0
 lpcor incoming vp_group1
 lpcor outgoing vp_group1
!
dial-peer voice 2 voip
 destination-pattern 2...
 lpcor outgoing voip_group2
 session protocol sipv2
 session target ipv4:192.168.97.1
!
voice register pool  3
 lpcor type remote
 lpcor incoming sip_group3
 lpcor outgoing sip_group3
 id mac 001E.BE8F.96C0
 type 7940
 number 1 dn 3
!
ephone 2
 mac-address 001C.821C.ED23
 type 7960
 button 1:2
 lpcor type remote
 lpcor incoming ephone_group2
 lpcor outgoing ephone_group2
```

Related Commands

Command	Description
lpcor incoming	Associates an incoming call with a LPCOR resource-group policy.
lpcor type	Specifies the LPCOR type for an IP phone.
port (dial-peer)	Associates a dial peer with a voice port.

Command	Description
trunkgroup	Associates a dial peer with a trunk group.
voice lpcor policy	Creates a LPCOR policy for a resource group.

lpcor type

To specify the logical partitioning class of restriction (LPCOR) type for an IP phone, use the **lpcor type** command in ephone, ephone-template, voice register pool, or voice register template configuration mode. To reset to the default, use the **no** form of this command.

lpcor type {**local**| **mobile**| **remote**}

no lpcor type

Syntax Description

local	IP phone always registers to Cisco Unified CME through the LAN.
mobile	IP phone can register to Cisco Unified CME through the LAN or WAN.
remote	IP phone always registers to Cisco Unified CME through the WAN.

Command Default

LPCOR feature is disabled for the IP phone.

Command Modes

Ephone configuration (config-ephone) Ephone-template configuration (config-ephone-template) Voice register pool configuration (config-register-pool) Voice register template configuration (config-register-temp)

Command History

Cisco IOS Release	Cisco Product	Modification
15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.
15.1(1)T	Cisco Unified CME 8.0	This command was integrated into Cisco IOS Release 15.1(1)T.

Usage Guidelines

According to the Telecom Regulatory Authority of India (TRAI) requirements, an IP phone can accept both PSTN and VoIP calls if it is locally registered to Cisco Unified CME through the LAN. Select the **local** keyword for this type of phone.

If an IP phone is registered remotely to Cisco Unified CME through the WAN, PSTN calls must be blocked from that remote IP phone. Select the **remote** keyword for this type of phone.

A static LPCOR policy is applied to an IP phone if the phone registers to Cisco Unified CME from the same region (local or remote) permanently.

If an IP phone moves between the local and remote regions, such as an Extension Mobility phone, Cisco IP Communicator softphone, or remote teleworker, select the **mobile** keyword. The LPCOR policy is assigned dynamically based on the phone's currently registered IP address.

If you use a phone template to apply a command to a phone and you also use the same command in the phone configuration of the same phone, the value in phone configuration has priority.

Examples

The following example shows that SCCP IP phone 2 is set to the remote LPCOR type:

```
ephone 2
 mac-address 001C.821C.ED23
 type 7960
 button 1:2
 lpcor type remote
 lpcor incoming ephone_group2
 lpcor outgoing ephone_group2
```

Related Commands

Command	Description
lpcor incoming	Associates a LPCOR resource-group policy with an incoming call.
lpcor outgoing	Associates a LPCOR resource-group policy with an outgoing call.
voice lpcor policy	Creates a LPCOR policy for a resource group.