



# Cisco Unified CME Commands: I

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

# ica

To specify the audio file used for the isolated code announcement, use the **ica** command in voice MLPP configuration mode. To disable use of this audio file, use the **no** form of this command.

**ica** *audio-url*

**no ica**

## Syntax Description

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

## 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 service or equipment problems prevent completion of their call.

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)# ica flash:ica.au
```

## Related Commands

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

## id (voice register pool)

To explicitly identify a locally available individual Cisco SIP IP phone, or when running Cisco Unified Session Initiation Protocol (SIP) Survivable Remote Site Telephony (SRST), set of Cisco SIP IP phones, use the **id** command in voice register pool configuration mode. To remove local identification, use the **no id** form of this command.

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

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

### Syntax Description

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

### Command Default

No SIP IP phone is configured.

### Command Modes

Voice register pool configuration (config-register-pool)

### Command History

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

### Usage Guidelines

Configure this command before configuring any other command in voice register pool configuration mode.

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

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

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

**Note**

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

**Examples**

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

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

**Related Commands**

Command	Description
<b>mode (voice register global)</b>	Enables the mode for provisioning SIP phones in a Cisco Unified CallManager Express (Cisco Unified CME) system.

## index (lpcor ip-phone)

To add a logical partitioning class of restriction (LPCOR) group to the IP-phone subnet table, use the **index** command in LPCOR ip-phone subnet configuration mode. To remove a resource, use the **no** form of this command.

```
index index-number lpcor-group {ipv4-address network-mask [vrf vrf-name] | dhcp-pool
pool-name}
```

```
no index index-number
```

Syntax Description		
<i>index-number</i>		Number of the LPCOR subnet index entry. Range: 1 to 50.
<i>lpcor-group</i>		Name of a LPCOR resource-group policy.
<i>ipv4-address</i>		IPv4 address of the LPCOR policy.
<i>network-mask</i>		Subnet mask for the associated IPv4 address.
<b>vrf</b> <i>vrf-name</i>		(Optional) Dynamic Host Configuration Protocol (DHCP) server uses the VPN routing and forwarding (VRF) table that is associated with the access point name (APN).
<b>dhcp-pool</b> <i>pool-name</i>		User-defined name of the DHCP pool. The pool name can be a symbolic string (such as Sales) or an integer (such as 0).

**Command Default** No index entry is configured.

**Command Modes** LPCOR ip-phone subnet configuration (cfg-lpcor-ipphone-subnet)

Command History	Cisco IOS Release	Cisco Product	Modification
	15.0(1)XA	Cisco Unified CME 8.0	This command was introduced.

**Usage Guidelines**

This command is used for mobility-type phones only, which can include Extension Mobility phones, teleworker remote phones, and Cisco IP Communicator softphones.

Two IP-phone subnet tables, containing up to 50 index entries, can be defined on each Cisco Unified CME router. One table is for incoming calls and the other table is for outgoing calls.

A LPCOR policy is dynamically associated with calls to and from a mobility-type phone by matching its current IP address or DHCP pool in the IP-phone subnet table. If the LPCOR policy cannot be provisioned from the IP-phone subnet table, the default LPCOR policy for mobility-type phones is used.

Entries in the IP-phone subnet tables are indexed in ascending order. The lookup of entries is in sequential ascending order. After Cisco Unified CME finds a matching entry, the corresponding LPCOR policy is associated with the call. Even if there are other entries that are a better match, only the first match is used.

For instance, in the example below, if a call originates from an IP phone with IP address 10.1.10.3, LPCOR policy local\_g4 is associated with the incoming call instead of LPCOR policy local\_g5 even though local\_g5 is a better match.

### Examples

The following example shows an IP-phone subnet table for incoming calls that has four entries:

```
voice lpcor ip-phone subnet incoming
index 1 local_g4 10.1.10.0 255.255.255.0
index 2 remote_g4 171.19.0.0 255.255.0.0
index 3 local_g5 10.1.10.2 255.255.255.255
index 4 local_g5 10.1.10.3 255.255.255.255
```

### Related Commands

Command	Description
<b>lpcor type</b>	Specifies the LPCOR type for an IP phone.
<b>voice lpcor ip-phone mobility</b>	Sets the default LPCOR policy for mobility-type phones.
<b>voice lpcor policy</b>	Creates a LPCOR policy for a resource group.

## index (lpcor ip-trunk)

To add a logical partitioning class of restriction (LPCOR) resource group to the IP trunk subnet table, use the **index** command in LPCOR IP-trunk subnet configuration mode. To remove a resource, use the **no** form of this command.

```
index number lpcor-group {ipv4-address network-mask | hostname host-name}
```

```
no index number
```

### Syntax Description

<i>number</i>	Number of the LPCOR subnet index entry. Range: 1 to 50.
<i>lpcor-group</i>	Name of a LPCOR resource-group policy.
<i>ipv4-address</i>	IPv4 address of the LPCOR policy.
<i>network-mask</i>	Subnet mask of the associated IPv4 address.
<b>hostname</b> <i>host-name</i>	User-defined IP host name.

### Command Default

No index entry is configured.

### Command Modes

LPCOR IP-trunk subnet configuration (cfg-lpcor-iptrunk-subnet)

### Command History

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

### Usage Guidelines

One IP-trunk subnet table, containing up to 50 index entries, can be defined on each Cisco Unified CME router for incoming VoIP trunk calls (H.323 or SIP).

An incoming VoIP trunk call is associated with a LPCOR policy by matching the remote IP address to an entry in the incoming IP-trunk subnet table. If that is not successful, the LPCOR policy in voice service configuration mode is applied.

Entries in the IP-trunk subnet table are indexed in ascending order. The lookup of entries is in sequential ascending order. After Cisco Unified CME finds a matching entry, it associates the corresponding LPCOR policy with the call. Even if there are other entries that are a better match, only the first match is used.

In the following example, an incoming VoIP call with a remote IP address of 172.19.22.25 is associated with sip\_group1 even though voip\_group2 is a better match.

### Examples

The following example shows an IP-trunk subnet table with six index entries:

```
voice lpcor ip-trunk subnet incoming
  index 1 h323_group1 172.19.33.0 255.255.255.0
  index 2 sip_group1 172.19.22.0 255.255.255.0
  index 3 voip_group2 172.19.33.25 255.255.255.255
  index 4 voip_group3 172.19.22.26 255.255.255.255
```

## ■ index (lpcor ip-trunk)

```
index 5 sip_s1 hostname sipserver1
index 6 sip_s2 hostname sipserver2
```

---

**Related Commands**

<b>Command</b>	<b>Description</b>
<b>lpcor incoming</b>	Associates an incoming call with a LPCOR resource-group policy.
<b>voice lpcor policy</b>	Ccreates a LPCOR policy for a resource group.

# intercom (ephone-dn)

To create an intercom by programming a pair of extensions (ephone-dns) to automatically call and answer each other, use the **intercom** command in ephone-dn configuration mode. To remove an intercom, use the **no** form of this command.

```
intercom extension-number [[barge-in [no-mute] | no-auto-answer | no-mute] [label label]] |
label label]
```

```
no intercom
```

Syntax Description		
<i>extension-number</i>		Extension or telephone number to which calls are placed.
<b>barge-in</b>		(Optional) Allows inbound intercom calls to force an existing call into the call-hold state and the intercom call to be answered immediately.
<b>label</b> <i>label</i>		(Optional) Defines an alphanumeric label for the intercom, of up to 30 characters.
<b>no-auto-answer</b>		(Optional) Disables the intercom auto-answer feature.
<b>no-mute</b>		(Optional) Allows an intercom call to be answered without deactivating a speaker's mute key.

**Command Default** Intercom functionality is disabled.

**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(8)T	Cisco ITS 2.0	This command was integrated into Cisco IOS Release 12.2(8)T.
	12.3(11)XL	Cisco CME 3.2.1	The <b>no-mute</b> keyword was added.
	12.3(14)T	Cisco CME 3.3	This command was integrated into Cisco IOS Release 12.3(14)T.

**Usage Guidelines** This command is used to dedicate a pair of Cisco ephone-dns for use as a “press to talk” two-way intercom between Cisco IP phones. Intercom lines cannot be used in shared-line configurations. If an ephone-dn is configured for intercom operation, it must be associated with one Cisco IP phone only. The intercom attribute causes an IP extension (ephone-dn) to operate in autodial fashion for outbound calls and autoanswer-with-mute for inbound calls.

The **barge-in** keyword allows inbound intercom calls to force an existing call on the called phone into the call-hold state to allow the intercom call to be answered immediately. The **no-auto-answer** keyword creates for the IP phone line a connection that resembles a private line, automatic ringdown (PLAR). The **label** keyword defines a text label for the intercom.

Following this command, the intercom ephone-dns are assigned to ephones using the **button** command. Following the **button** command, the **restart** command must be used to initiate a quick reboot of the phones to which this intercom is assigned.

The default **intercom** command behavior is speakers are set to mute automatically when phones receive intercom calls. For example, if phone user 1 places an intercom call and connects to phone user 2, user 2 will hear user 1, but user 1 will not hear user 2. To be heard, user 2 must first disable the speaker's mute function. The benefit is people who receive intercom calls can use the mute button to control when they will be heard initially.

The **no-mute** keyword deactivates the speaker mute function when IP phones receive intercom calls. For example, if phone user 1 makes an intercom call to phone user 2, both users will hear each other upon connection. The benefit is that people who receive intercom calls do not have to disable their speaker's mute function to be heard, *but* their conversations and nearby background sounds will be heard the moment an intercom call to them is connected—regardless of whether they are ready to take a call or not.

## Examples

The following example sets the intercom on Cisco IP phone directory number 1:

```
Router(config)# ephone-dn 1
Router(config-ephone-dn) number A5001
Router(config-ephone-dn) name "intercom"
Router(config-ephone-dn) intercom A5002 barge-in
```

The following example shows intercom configuration between two Cisco IP phones:

```
ephone-dn 18
 number A5001
 name "intercom"
 intercom A5002 barge-in
ephone-dn 19
 number A5002
 name "intercom"
 intercom A5001 barge-in
ephone 4
 button 1:2 2:4 3:18
 ephone 5
 button 1:3 2:6 3:19
```

In the example, ephone-dn 18 and ephone-dn 19 are set as an intercom pair. Ephone-dn 18 is associated with button 3 of Cisco IP phone (ephone) 4, and ephone-dn 19 is associated with button number 3 of Cisco IP phone (ephone) 5. Button 3 on Cisco IP phone 4 and button 3 on Cisco IP phone 5 are set as a pair to provide intercom service to each other.

The intercom feature acts as a combination speed-dial PLAR and autoanswer with mute. If the **barge-in** keyword is set on the ephone-dn that receives the intercom call, the existing call is forced into the hold state, and the intercom call is accepted. If the phone user has the handset off hook (that is, not in speakerphone mode), the user hears a warning beep, and the intercom call is immediately connected with two-way audio. If the phone user is using speakerphone mode, the intercom connects with the microphone mute activated.



### Note

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

<b>Related Commands</b>	<b>Command</b>	<b>Description</b>
	<b>button</b>	Associates ephone-dns with individual buttons on Cisco IP phones and specifies ring behavior per button.
	<b>number</b>	Associates a telephone or extension number with an extension (ephone-dn).
	<b>restart (ephone)</b>	Performs a fast reboot of a single phone associated with a Cisco CME router.
	<b>restart (telephony-service)</b>	Performs a fast reboot of one or all phones associated with a Cisco CME router.

# internal-call

To assign an MOH group for calls from an internal directory number, use the **internal-call** command in telephony-service configuration mode. To disable the internal-call command, use the **no** form of this command.

**internal-call** *moh-group-tag*

**no internal-call**

## Syntax Description

<i>moh-group-tag</i>	Specifies a MOH-group number to be used for calls from an internal directory number. Range is from 0 to 5, where 0 represents MOH configuration in telephony-service configuration mode.
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## Command Default

No internal-call is configured.

## Command Modes

Telephony-service configuration (config-telephony-service)

## Command History

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

## Usage Guidelines

Before using this command make sure you have MOH-groups configured under **voice-moh-group** configuration mode. This command allows you to assign a MOH-group for all calls from an internal directory number. MOH group *tag* identifies the unique number assigned to a MOH group. Range for MOH group tag is from 0 to 5, where 0 represents MOH configuration in telephony service.

## Examples

The following example shows MOH-group 4 assigned for an internal directory number:

```
telephony-service
  internal-call moh-group 4
  em logout 0:0 0:0 0:0
  max-ephones 58
  max-dn 192
  ip source-address 15.1.0.161 port 2000
  max-conferences 8 gain -6
  moh music-on-hold.au
  multicast moh 239.1.1.1 port 2000
  transfer-system full-consult
```

## Related Commands

Command	Description
<b>voice-moh-group</b>	Enter voice-moh-group configuration mode.
<b>moh filename</b>	Enables music on hold from a flash audio feed

<b>Command</b>	<b>Description</b>
<b>multicast moh</b>	Enables multicast of the music-on-hold audio stream.
<b>extension-range</b>	Specifies the extension range for a clients calling a voice-moh-group.

## ip qos dscp (telephony-service and voice register global)

To set the Differentiated Services Code Point (DSCP) for marking the quality of service (QoS) requirements for each packet, use the **ip qos dscp** command in telephony-service or voice register global configuration mode. To reset to the default value, use the **no** form of this command.

```
ip qos dscp {{number | af | cs | default | ef} {media | service | signaling | video}}
```

```
no ip qos dscp {{number | af | cs | default | ef} {media | service | signaling | video}}
```

### Syntax Description

<i>number</i>	DSCP value. Range: 0 to 63.
<i>af</i>	Sets DSCP to assured forwarding bit pattern. <ul style="list-style-type: none"> <li>• <b>af11</b>—bit pattern 001010</li> <li>• <b>af12</b>—bit pattern 001100</li> <li>• <b>af13</b>—bit pattern 001110</li> <li>• <b>af21</b>—bit pattern 010010</li> <li>• <b>af22</b>—bit pattern 010100</li> <li>• <b>af23</b>—bit pattern 010110</li> <li>• <b>af31</b>—bit pattern 011010</li> <li>• <b>af32</b>—bit pattern 011100</li> <li>• <b>af33</b>—bit pattern 011110</li> <li>• <b>af41</b>—bit pattern 100010</li> <li>• <b>af42</b>—bit pattern 100100</li> <li>• <b>af43</b>—bit pattern 100110</li> </ul>
<i>cs</i>	Sets DSCP to class-selector codepoint. <ul style="list-style-type: none"> <li>• <b>cs1</b>—codepoint 1 (precedence 1)</li> <li>• <b>cs2</b>—codepoint 2 (precedence 2)</li> <li>• <b>cs3</b>—codepoint 3 (precedence 3)</li> <li>• <b>cs4</b>—codepoint 4 (precedence 4)</li> <li>• <b>cs5</b>—codepoint 5 (precedence 5)</li> <li>• <b>cs6</b>—codepoint 6 (precedence 6)</li> <li>• <b>cs7</b>—codepoint 7 (precedence 7)</li> </ul>
<b>default</b>	Sets DSCP to default bit pattern of 000000.
<b>ef</b>	Sets DSCP to expedited forwarding bit pattern 101110.
<b>media</b>	Applies DSCP to media payload packets.
<b>service</b>	Applies DSCP to phone service including HTTP traffic.
<b>signaling</b>	Applies DSCP to signaling packets.
<b>video</b>	Applies DSCP to video stream.

**Defaults**

DSCP for media is **ef**.  
 DSCP for service is **0**.  
 DSCP for signaling is **cs3**.  
 DSCP for video is **af41**.

**Command Modes**

Telephony-service configuration (config-telephony)  
 Voice register global configuration (config-register-global)

**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 allows you to set different priority levels for different types of network traffic sent by the Cisco Unified CME router. Differentiated Services is a method of prioritizing specific network traffic based on the QoS specified by each packet. You can set different DSCP values, for example, for video and audio streams.

Cisco Unified CME downloads the configured DSCP value to the phones in their configuration files and all control messages and RTP streams are marked with the preferred DSCP value. Use this command in telephony-service mode to set the DSCP for SCCP phones. Use the command in voice register global mode to set the value for SIP phones.

If the DSCP is configured for the gateway interface using the **service-policy** command or in the dial peer using the **ip qos dscp** command, the value set with those commands takes precedence over the DSCP value configured with this command.

**Examples**

The following examples show the configuration of DSCP for different types of packets.

```
voice register global
 mode cme
 ip qos dscp af11 media
 ip qos dscp cs2 signal
 ip qos dscp af43 video
 ip qos dscp 25 service

telephony-service
 load 7960-7940 P00308000500
 max-ephones 100
 max-dn 240
 ip source-address 10.7.0.1 port 2000
 ip qos dscp af11 media
 ip qos dscp cs2 signal
 ip qos dscp af43 video
 ip qos dscp 25 service
```

**Related Commands**

## ■ ip qos dscp (telephony-service and voice register global)

Command	Description
<b>ip qos dscp</b>	Sets the DSCP for QoS in a dial peer.
<b>service-policy</b>	Assigns a policy map to an interface that will be used as the service policy for the interface.

# ip source-address (credentials)

To enable the Cisco Unified CME or Cisco Unified SRST router to receive credential service messages through the specified IP address and port, use the **ip source-address** command in credentials configuration mode. To disable the router from receiving messages, use the **no** form of this command.

**ip source-address** *ip-address* [**port** [*port*]]

**no ip source-address**

## Syntax Description

<i>ip-address</i>	Router IP address, typically one of the addresses of the Ethernet port of the local router.
<b>port</b> <i>port</i>	(Optional) TCP port for credentials service communication. Range is from 2000 to 9999. Cisco Unified CME default is 2444. SRST default is 2445.

## Command Default

Default port number in Cisco Unified CME is 2444.  
Default port number in Cisco Unified SRST is 2445.

## 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 in Cisco IOS Release 12.4(9)T.

## Usage Guidelines

### Cisco Unified CME

This command is used with Cisco Unified CME phone authentication to identify a Cisco Unified CME router on which a CTL provider is being configured.

### Cisco Unified SRST

The **ip source-address** command is a mandatory command to enable secure SRST. If the port number is not provided, the default value (2445) is used. The IP address is usually the IP address of the secure SRST router.

**Examples****Cisco Unified CME**

The following example creates a CTL provider on a Cisco Unified CME router that is not running the CTL client.

```
Router(config)# credentials
Router(config-credentials)# ip source-address 172.19.245.1 port 2444
Router(config-credentials)# trustpoint ct1pv
Router(config-credentials)# ctl-service admin user4 secret 0 c89L8o
```

**Cisco Unified SRST**

The following example enters credentials configuration mode and sets the IP source address and port:

```
Router(config)# credentials
Router(config-credentials)# ip source-address 10.6.21.4 port 2445
```

**Related Commands**

	Description
<b>ctl-service admin</b>	Specifies a user name and password to authenticate the CTL client during the CTL protocol.
<b>debug credentials</b>	Sets debugging on the credentials service that runs between a Cisco Unified CME CTL provider and the CTL client or between an SRST router and Cisco Unified CallManager.
<b>show credentials</b>	Displays the credentials settings on a Cisco Unified CME or SRST router.
<b>trustpoint (credentials)</b>	Specifies the name of the trustpoint to be associated with a Cisco Unified CME CTL provider certificate or with an SRST router certificate.

## ip source-address (telephony-service)

To identify the IP address and port through which IP phones communicate with a Cisco Unified CME router, use the **ip source-address** command in telephony-service or group configuration mode. To disable the router from receiving messages from Cisco Unified IP phones, use the **no** form of this command.

```
ip source-address { <ipv4 address> | <ipv6 address> } [port port] [secondary { ipv4 address> | <ipv6 address> } ] [rehome seconds] [strict-match]
```

```
no ip source-address
```

### Syntax Description

<i>ipv4-address</i>	IPv4 address of the router, typically one of the addresses of the Ethernet port of the router. IPv4 address can only be configured with IPv4 address or dual-stack (IPv4/IPv6).
<i>ipv6 address</i>	IPv6 address of the router, IPv6 can only be configured with IPv6 address or dual-stack (IPv4/IPv6).
<b>port</b> port	(Optional) TCP/IP port number to use for Skinny Client Control Protocol (SCCP). Range is from 2000 to 9999. Default is 2000.  <b>Note</b> For IPv6, do not change the default port number (2000) in the “ip source-address” configuration command. If you change the port number, IPv6 CEF packet switching engine will not be able to handle the IPv6 SCCP phones and various packet handling problems may occur when more than a dozen (approximately) calls in IPv6 are going on.
<b>secondary</b>	(Optional) Second Cisco Unified CME router with which phones can register if the primary Cisco Unified CME router fails. For dual-stack (IPv4/IPv6) port is only configurable with IPv4 address.
<b>rehome</b> seconds	(Optional) Used only by Cisco Unified IP phones that have registered with a Cisco Unified Survivable Remote Site Telephony (SRST) router. This keyword defines a delay that is used by phones to verify the stability of their primary SCCP controller (Cisco Unified Communications Manager or Cisco Unified CME) before the phones reregister with it. This parameter is ignored by phones unless they are registered to a secondary Cisco Unified SRST router. The range is from 0 to 65535 seconds. The default is 120 seconds.  The use of this parameter is a phone behavior and is subject to change, based on the phone type and phone firmware version.
<b>strict-match</b>	(Optional) Requires strict IP address checking for registration.

### Command Default

The IP address for communicating with phones is not defined.

### Command Modes

Telephony-service configuration (config-telephony)  
Group configuration (conf-tele-group)

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.
	12.4(4)XC	Cisco Unified CME 4.0	The <b>secondary ip-address</b> and <b>rehome seconds</b> keyword-argument pairs were added.
	12.4(9)T	Cisco Unified CME 4.0	The <b>secondary ip-address</b> and <b>rehome seconds</b> keyword-argument pairs were added.
	12.4(22)T		This command was added to VRF group mode.
	15.0(1)XA	Cisco Unified CME 8.0	This command was modified. Support for IPv6 was added. The <i>ipv6 address</i> keyword-argument was added to <b>ip source-address</b> and <b>secondary</b> keywords.

### Usage Guidelines

This command enables a router to receive messages from Cisco Unified IP phones through the specified IP address and port. The Cisco Unified CME router cannot communicate with the Cisco Unified CME phones if the IP address is not provided. If the port number is not provided, the default is port 2000. The IP address is usually the IP address of the Ethernet port to which the phones are connected.

Use the **strict-match** keyword to instruct the router to reject IP phone registration attempts if the IP server address used by the phone does not match the source address.

Prior to Cisco IOS Telephony Services (Cisco ITS) V2.1, this command helped the router to autogenerate the SEPDEFAULT.cnf file, which was stored in the flash memory of the router. The SEPDEFAULT.cnf file contains the IP address of one of the Ethernet ports of the router to which the phone should register. In ITS V2.1, Cisco CME 3.0, and later versions, the configuration files were moved to system:/its/. The file named Flash:SEPDEFAULT.cnf that was used with previous Cisco ITS versions is obsolete, but is retained as system:/its/SEPDEFAULT.cnf to support upgrades from older phone firmware.

For systems using Cisco ITS V2.1 or later versions, the IP phones receive their initial configuration information and phone firmware from the TFTP server associated with the router. In most cases, the phones obtain the IP address of their TFTP server using the **option 150** command and Dynamic Host Configuration Protocol (DHCP). For Cisco ITS or Cisco CME operation, the TFTP server address obtained by the Cisco Unified IP phones should point to the router IP address. The Cisco IP phones attempt to transfer a configuration file called XmlDefault.cnf.xml. This file is automatically generated by the router through the **ip source-address** command and is placed in router memory. The XmlDefault.cnf.xml file contains the IP address that the phones use to register for service, using the SCCP. This IP address should correspond to a valid Cisco CME router IP address (and may be the same as the router TFTP server address).

Similarly, when an analog telephone adapter (ATA) such as the ATA-186 is attached to the Cisco Unified CME router, the ATA receives very basic configuration information and firmware from the TFTP server XmlDefault.cnf.xml file. The XmlDefault.cnf.xml file is automatically generated by the Cisco Unified CME router with the **ip source-address** command and is placed in the router's flash memory.

By specifying a second Cisco Unified CME router in the **ip source-address** command, you improve the failover time for phones.

In Cisco Unified CME 8.0 and later versions you can specify an IPv6 address for the router. Cisco Unified CME router is enabled to receive messages from IPv6 enabled IP phones or phones in dual-stack (both IPv6 and IPv4) mode.

### Examples

The following example sets the IP source address and port:

```
Router(config)# telephony-service
Router(config-telephony)# ip source-address 10.6.21.4 port 2000 strict-match
```

The following example establishes the router at 10.5.2.78 as a secondary router:

```
Router(config)# telephony-service
Router(config-telephony)# ip source-address 10.0.0.1 port 2000 secondary 10.5.2.78
```

The following example shows an IPv6 address:

```
Router(config)# telephony-service
Router(config-telephony)# ip source-address 10.10.10.33 port 2000 ip source-address
2001:10:10:10::3
```

### Related Commands

Command	Description
<b>option</b>	Configures DHCP server options.

