

System-Level Parameters

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Prerequisites for System-Level Parameters

- To directly connect Cisco Unified IP phones that are running Session Initiation Protocol (SIP) in Cisco Unified CME, Cisco CME 3.4 or a later version must be installed on the router. For installation information, see Install and Upgrade Cisco Unified CME Software.
- Cisco Unified CME must be configured to work with your IP network. For configuration information, see Network Parameters.

Information About Configuring System-Level Parameters

Bulk Registration Support for SIP Phones

Cisco Unified CME 8.6 enhances the bulk registration feature for Cisco Unified SIP IP phones by optimizing the two main transactions involved in bulk registration process and minimizing the number of required messages to be sent to the phones. The bulk registration process involves the following two main transactions:

- Register—Register transaction handles per line REGISTER messages coming to Cisco Unified CME and provisions phone DNs by creating dialpeers and various phone data structures.
- Phone Status Update—Phone status update transaction sends back device information using REFER and NOTIFY messages.

In Cisco Unified CME 8.6, the bulk registration process consists of only one REGISTER message per phone instead of one REGISTER message per phone per line, thus reducing any negative impact on your router's performance. For information on configuring bulk registration, see Configure Bulk Registration for SIP IP Phones, on page 25.

The **show voice register pool** command displays the registration method a phone uses: per line, bulk-in progress, or bulk-completed. The per line option indicates that the phone is using the per line registration process. The bulk-in progress option indicates that the phone is using the bulk registration process but the registration process is not complete yet. The bulk-completed option indicates that the phone is registered using the bulk registration process and the registration process is complete. For information on verifying the phone registration process, see Verify Phone Registration Type and Status, on page 26.



Note The bulk registration feature in Cisco Unified CME 8.6 optimizes line registration on SIP phones and is a phone interop feature. The bulk registration feature is not related to the **bulk** command under voice register global configuration mode.

In earlier versions of Cisco Unified CME, the registration process was very lengthy and several SIP messages were exchanged between the end points and Cisco Unified CME to properly provision the phone.

Table 1: Number of Messages Required for an Eight-Button IP Phone, on page 2 lists the number of messages required to register an eight-button Cisco Unified SIP IP phone, where all of the eight buttons can be configurd as a shared line with message waiting indicator (MWI) notification enabled, to Cisco Unified CME.

Table 1: Number of Messages Required for an Eight-Button IP Phone

Transactions	Method	Messages Per Transaction	Number of Transactions	Total number of messages (per line)	Total number of messages (bulk)
Register	REGISTER	2	8	24	3
Phone Status Update	REFER remotecc NOTIFY (mwi, service-control)	2 2	3 8	6 16	2
Subscription	SUBSCRIBE (sharedline)	4	8	32	32
Total				78	37

You can see from the preceding table that more than 70 messages are required to register one 8-button IP phone. If there is a simultaneous registration of more phones, the amount of messages can be overwhelming and can have a negative impact on the performance of the router.

With the enhanced bulk registration process, the two main transactions (Register and Phone Status Update) are optimized to minimize the number of messages required to complete the phone registration process. Table 1: Number of Messages Required for an Eight-Button IP Phone, on page 2 shows that the total number of messages required for bulk registration is only 37.

Register Transaction

The following is an example of the REGISTER message:

```
REGISTER sip:28.18.88.1 SIP/2.0
Via: SIP/2.0/TCP 28.18.88.33:44332;branch=z9hG4bK53f227fc
From: <sip:6010@28.18.88.1>;tag=001b2a893698027db8ea0454-26b9fb0c
```

```
 To: <sip:6010@28.18.88.1>

 Call-ID: 001b2a89-3698011e-280209a4-567e339c@28.18.88.33

 Max-Forwards: 70

 Date: Wed, 03 Mar 2010 01:18:34 GMT

 CSeq: 240 REGISTER

 User-Agent: Cisco-CP7970G/8.4.0

 Contact: <sip:6010@28.18.88.33:44332;transport=tcp >
;+sip.instance="<urn:uuid:00000000-0000-0000-0000-001b2a893698 >
 ";+u.sip!model.ccm.cisco.com="30006"

 Supported:
replaces,join,norefersub,extended-refer,X-cisco-callinfo,X-cisco-serviceuri,X-cisco-escapecodes,
X-cisco-service-control,X-cisco-srtp-fallback,X-cisco-monrec,X-cisco-config,X-cisco-sis-3.0.0,X-cisco-xsi-7.0.1


 Reason: SIP;cause=200;text="cisco-alarm:23 Name=SEP001B2A893698 Load=SIP70.8-4-2-30S
Last=reset-restart"

 Expires: 3600

 Content-Type: multipart/mixed; boundary=uniqueBoundary

 Mime-Version: 1.0

 Content-Length: 982

 --uniqueBoundary

 Content-Type: application/x-cisco-remotecc-request+xml

 Content-Disposition: session;handling=optional


 >

  < x-cisco-remotecc-request >

 <bulkregisterreq >

  < contact all="true" >

  < register >  < /register >

  < /contact >

  < /bulkregisterreq >

  < /x-cisco-remotecc-request >


 --uniqueBoundary

 Content-Type: application/x-cisco-remotecc-request+xml

 Content-Disposition: session;handling=optional


  >

  < x-cisco-remotecc-request >

   < optionsind >

    < combine max="6" >

     < remotecc >

      < status >  < /status >

     < /remotecc >
```

```
    < service-control >  < /service-control >
   < /combine >
   < dialog usage="hook status" >
    < unot >  < /unot >
    < sub >  < /sub >
   < /dialog >
   < dialog usage="shared line" >
    < unot >  < /unot >
    < sub >  < /sub >
   < /dialog >
   < presence usage="blf speed dial" >
    < unot >  < /unot >
    < sub >  < /sub >
   < /presence >
   < joinreq >  < /joinreq >
  < /optionsind >
 < /x-cisco-remotecc-request >

--uniqueBoundary--
```

The following is an example of a response to the preceding REGISTER message:

```
SIP/2.0 200 OK
Date: Wed, 03 Mar 2010 01:18:41 GMT
From:  < sip:6010@28.18.88.1 > ;tag=001b2a893698027db8ea0454-26b9fb0c
Content-Length: 603
To:  < sip:6010@28.18.88.1 > ;tag=E2556C-6C1
Contact:  < sip:6010@28.18.88.33:44332;transport=tcp > ;expires=3600;x-cisco-newreg
Expires:  3600
Content-Type: multipart/mixed;boundary=uniqueBoundary
Call-ID: 001b2a89-3698011e-280209a4-567e339c@28.18.88.33
Via: SIP/2.0/TCP 28.18.88.33:44332;branch=z9hG4bK53f227fc
Server: Cisco-SIPGateway/IOS-12.x
CSeq: 240 REGISTER
Mime-Version: 1.0
  >  < x-cisco-remotecc-response >  < response >  < code > 200 < /code >  < optionsind >
< combine max="6" >  < remotecc >
  < status/ >  < /remotecc >  < service-control/ >  < /combine >  < dialog usage="shared
line" >  < sub/ >  < /dialog >
< presence usage="blf speed dial" >  < sub/ >  < /presence >  < /optionsind >  < /response
 >  < /x-cisco-remotecc-response >
```

Phone Status Update Transaction

Cisco Unified IP phones use the option indication to negotiate supported options with Cisco Unified CME via remotecc request. Cisco Unified CME selects an option or options that it wishes to support and return it in the response. Cisco Unified CME ignores items (elements, attributes, and values) that it fails to understand. A new phone option, combine, is defined to optimize phone status update. This option combines remotecc status information (cfwdall, privacy, dnd, bulk mwi) and service-control. The following is an example of a combined status update:

```
<optionsind>
<combine max="5">
<remotecc><status/></remotecc>
<service-control/>
</combine>
</optionsind>
```

The following is another example of a combined status update:

```
<optionsind>
<combine max="4">
<remotecc><status/></remotecc>
<service-control/>
</combine>
</optionsind>
```

To minimize the data size, Cisco Unified CME and the phone agree ahead of time on a default value to apply updates. Therefore, during initial registration, Cisco Unified CME will not send the value if it matches the agreed upon default. Table 2: Status Information and Default, on page 5 captures the existing status information and applicable default value.

Table 2: Status Information and Default

Status	Default	Initialization
CallForwardAll Update	No default	Always send regardless of the value
Privacyrequest	Disabled	Only send if the value is not equal to the default
DnDupdate	Disabled	Only send if value is not equal to the default
Bulkupdate (MWI)	No default	Always send regardless of value

During bulk registration, Cisco Unified CME uses a single REFER message to send combined phone status update message for phone status updates such as cfwdallupdate, privacyrequet, DnDupdate, and Bulkupdate (MWI) instead of sending phone status in individual NOTIFY or REFER message to the phone. The following is an example of the single REFER message sent by Cisco Unified CME to the phone:

```
REFER sip:6010@28.18.88.33:44332 SIP/2.0
Content-Id: <1483336>
From: <sip:28.18.88.1>;tag=E256D4-2316
Timestamp: 1267579121
Content-Length: 934
User-Agent: Cisco-SIPGateway/IOS-12.x
```

```
Require: norefersub
Refer-To: cid:1483336
To: <sip:6010@28.18.88.33>
Contact: <sip:28.18.88.1:5060>
Referred-By: <sip:28.18.88.1>
Content-Type: multipart/mixed;boundary=uniqueBoundary
Call-ID: 89CBE590-259911DF-80589501-4E753388@28.18.88.1
Via: SIP/2.0/UDP 28.18.88.1:5060;branch=z9hG4bKA22639
CSeq: 101 REFER
Max-Forwards: 70
Mime-Version: 1.0

 --uniqueBoundary
 Content-Type: application/x-cisco-remotecc-request+xml

<x-cisco-remotecc-request>
<cfwdallupdate><fwdaddress></fwdaddress><tovoicemail>off</tovoicemail></cfwdallupdate></x-cisco-remotecc-request>



--uniqueBoundary
Content-Type: application/x-cisco-remotecc-request+xml



<x-cisco-remotecc-request>
<privacyreq><status>true</status></privacyreq>
</x-cisco-remotecc-request>
--uniqueBoundary
Content-Type: application/x-cisco-remotecc-request+xml


<x-cisco-remotecc-request>
<bulkupdate>
<contact all="true"><mwi>no</mwi></contact>
<contact line=" 1"><mwi>yes</mwi></contact>
<contact line=" 3"><mwi>yes</mwi></contact>
</bulkupdate>
</x-cisco-remotecc-request>


--uniqueBoundary
Content-Type: text/plain
action=check-version
RegisterCallId={001b2a89-3698011e-280209a4-567e339c@28.18.88.33}
ConfigVersionStamp={0106514225374329}
DialplanVersionStamp={}
SoftkeyVersionStamp={0106514225374329}

--uniqueBoundary--
```



Note Cisco Unified IP phones use the TCP for registration refresh. TCP socket has a default keepalive time out session of 60 minutes. If registration refresh to Cisco Unified CME does not takes place within an hour (60 minutes), the TCP connection will be removed. This will make the phones restart instead of refresh. To stop the phones from restarting, adjust the registrar expire timer under voice service voip or set the timer connection aging under sip-ua to a value greater than what the phone uses for registration refreshes. For example, if the phone does a registration refresh every 60 minutes, then setting up a timer connection aging to 100 minutes will guarantee that the TCP keeps the connection open. Or you can set the registrar expire maximum value to less than 3600.

DSCP

Differentiated Services Code Point (DSCP) packet marking is used to specify the class of service for each packet. Cisco Unified IP Phones get their DSCP information from the configuration file that is downloaded to the device.

In earlier versions of Cisco Unified CME, the DSCP value is predefined. In Cisco Unified CME 7.1 and later versions, you can configure the DSCP value for different types of network traffic. Cisco Unified CME downloads the configured DSCP value to SCCP and SIP phones in their configuration files and all control messages and flow-through RTP streams are marked with the configured DSCP value. This allows you to set different DSCP values, for example, for video streams and audio streams.

For configuration information, see Set Up Cisco Unified CME for SCCP Phones , on page 27 or Set Up Cisco Unified CME for SIP Phones, on page 43.

Maximum Ephones in Cisco Unified CME 4.3 and Later Versions

In Cisco Unified CME 4.3 and later versions, the **max-ephones** command is enhanced to set the maximum number of SCCP phones that can register to Cisco Unified CME, without limiting the number that can be configured. In previous versions of Cisco Unified CME, the **max-ephones** command defined the maximum number of phones that could be both configured and registered.

This enhancement expands the maximum number of phones that can be configured to 1000. The maximum number of phones that can register to Cisco Unified CME has not changed; it is dependent on the number of phones supported by the hardware platform and is limited by the **max-ephones** command.

This enhancement supports features, such as Extension Assigner, that require you to configure more phones than can register. For example, if you set the **max-ephones** command to 50 and configure 100 ephones, only 50 phones can register to Cisco Unified CME, one at a time in random order. The remaining 50 phones cannot register and an error message displays for each rejected phone. This enhancement also allows you to assign ephone tags that match the extension number of the phone, for extensions up to 1000.

If you reduce the value of the **max-ephones** command, currently registered phones are not forced to unregister until a reboot. If the number of registered phones, however, is already equal to or more than the max-ephones value, no additional phones can register to Cisco Unified CME. If you increase the value of the **max-ephones** command, the previously rejected ephones are able to register immediately until the new limit is reached.

Note For Cisco Integrated Services Router 4351, you can set the max-ephones value to 3925. For Cisco Integrated Services Router 4331, you can set the max-ephones value to 2921. For Cisco Integrated Services Router 4321, you can set the max-ephones value to 2901. For Cisco Integrated Services Router 4400 series, you can set the max-ephones value to 4451.

Network Time Protocol for SIP Phones

Although SIP phones can synchronize to a Cisco Unified CME router, the router can lose its clock after a reboot causing phones to display the wrong time. SIP phones registered to a Cisco Unified CME router can synchronize to a Network Time Protocol (NTP) server. Synchronizing to an NTP server ensures that SIP phones maintain the correct time. For configuration information, see Set Network Time Protocol for SIP Phones, on page 50.

Per-Phone Configuration Files

In Cisco Unified CME 4.0 and later versions, you can use an external TFTP server to off load the TFTP server function on the Cisco Unified CME router. Using flash memory or slot 0 memory on the Cisco Unified CME router allows you to use different configuration files for each phone type or for each phone, permitting you to specify different user locales and network locales for different phones. Before Cisco Unified CME 4.0, you could specify only a single default user and network locale for a Cisco Unified CME system.

You can specify one of the following four locations to store configuration files:

- System—This is the default. When system:/its is the storage location, there is only one default configuration file for all phones in the system. All phones, therefore, use the same user locale and network locale. User-defined locales are not supported.
- Flash or slot 0—When flash memory or slot 0 memory on the router is the storage location, you can create additional configuration files to apply per phone type or per individual phone. Up to five user and network locales can be used in these configuration files.

Note When the storage location you selected is flash memory and the file system type on this device is Class B (LEFS), you must check the free space on the device periodically and use the **squeeze** command to free the space used up by deleted files. Unless you use the **squeeze** command, the space used by the moved or deleted configuration files cannot be used by other files. Rewriting flash memory space during the squeeze operation may take several minutes. We recommend that you use this command during scheduled maintenance periods or off-peak hours.

- TFTP—When an external TFTP server is the storage location, you can create additional configuration files that can be applied per phone type or per individual phone. Up to five user and network locales can be used in these configuration files.

You can then specify one of the following ways to create configuration files:

- Per system—This is the default. All phones use a single configuration file. The default user and network locale in a single configuration file are applied to all phones in the Cisco Unified CME system. Multiple locales and user-defined locales are not supported.
- Per phone type—This setting creates separate configuration files for each phone type. For example, all Cisco Unified IP Phone 7960s use XMLDefault7960.cnf.xml, and all Cisco Unified IP Phone 7905s use XMLDefault7905.cnf.xml. All phones of the same type use the same configuration file, which is generated using the default user and network locale. This option is not supported if you store the configuration files in the system:/its location.
- Per phone—This setting creates a separate configuration file for each phone by MAC address. For example, a Cisco Unified IP Phone 7960 with the MAC address 123.456.789 creates the per-phone configuration file SEP123456789.cnf.xml. The configuration file for a phone is generated with the default user and network locale unless a different user and network locale is applied to the phone using an ephone template. This option is not supported if you store the configuration files in the system:/its location.

For configuration information, see Define Per-Phone Configuration Files and Alternate Location for SCCP Phones, on page 33.

HFS Download Support for IP Phone Firmware and Configuration Files

Legacy IP phones access the TFTP server to download firmware and configuration files but Cisco Unified CME 8.8 enhances download support for SIP phone firmware, scripts, midlets, and configuration files using the HTTP File-Fetch Server (HFS) infrastructure.

In Cisco Unified CME 8.8 and later versions, SIP phones use an HTTP server as the primary download service when it is configured and access a TFTP server as a secondary or fallback option when the HTTP server fails.

Note When the HFS download service is not configured, SIP phones automatically access the TFTP server.

The following scenario shows a successful download sequence using an HTTP server:

An IP phone initiates TCP connection to port 6970. A connection is established and an internal request for a file is sent to the HTTP server. The phone receives the HTTP response status code of 200, signifying that the download is successful.

The following scenario shows a download sequence that begins with an IP phone using an HTTP server to download files and ends with a TFTP server as a fallback option when the initial download attempt fails:

An IP phone initiates TCP connection to port 6970 but is unable to establish a connection. The phone contacts the TFTP server and sends an internal request for a file. The file is successfully downloaded from the TFTP server.

The following scenario shows how a download sequence that starts with an HTTP server does not always fall back to the TFTP server when the initial download attempt fails:

An IP phone initiates TCP connection to port 6970. A connection is established and an internal request for a file is sent to the HTTP server. The phone receives the HTTP response status code of 404, signifying that the file requested could not be found. Because the file cannot be found, the request is not sent to the TFTP server.

Note The configuration files are shared by the HTTP and TFTP servers. However, the firmware files are different for each server.

For more information on Phone Firmware Files, see Install and Upgrade Cisco Unified CME Software.

For more information on Per-Phone Configuration Files, see Per-Phone Configuration Files, on page 8.

For more information on Configuration Files for Phones in Cisco Unified CME, see Generate Configuration Files for Phones.

Enable HFS Service

To enable the HFS download service, the underlying HTTP server must be enabled first because the HFS infrastructure is built on top of an existing IOS HTTP server.

```
Router(config)# ip http server
```

This HFS infrastructure enables multiple HTTP services to co-exist. The HFS download service runs on custom port 6970 but can also share default port 80 with other services. Other HTTP services run on other non-standard ports like 1234.

```
Router(config)# ip http server
Router(config)# ip http port1234
```

The HFS download service starts when the following is configured in telephony-service configuration mode.

For the default port:

```
Router(config-telephony)# hfs enable
```

For the custom port:

```
Router(config-telephony)# hfs enable port 6970
```

Note If the entered custom HFS port clashes with the underlying IP HTTP port, an error message is displayed and the command is disallowed.

In the following example, port 6970 is configured as the IP HTTP port. When the HFS port is configured with the same value, an error message is displayed to show that the port is already in use.

```
Router (config)# ip http port 6970
.
.
Router (config)# telephony-service
Router (config-telephony)# hfs enable port 6970
```

Error Message Invalid port number or port in use by other application

Explanation The HFS port number is already in use by the underlying IP HTTP server.

Recommended Action Use an HFS port that is different from the underlying IP HTTP port.

Note Because IP phones are hardcoded to use port 6970 to connect to Cisco Unified CME, you must search for other applications running on port 6970 and assign them with ports different from 6970 to prevent a failure in connecting to Cisco Unified CME.

For configuration information, see Enable HFS Download Service for SIP Phones, on page 51.

File Binding and Fetching

File binding and fetching using the HTTP server can be classified into two:

- Explicit binding – The **create profile** command triggers the system to generate the configuration and firmware files and store them in RAM or a flash memory. The system asks the new internal application programming interfaces (APIs) implemented by the HFS download service to bind the filename and alias that an IP phone wants to access to their corresponding URL.
- Loose binding – The HFS download service enables the Cisco Unified CME system to configure a home path from where any requested firmware file that has no explicit binding can be searched and fetched. The files can be stored on any device (such as flash memory or NVRAM) under a root directory or a suitable subdirectory.

  No matter how the system is configured, if there is no explicit binding, the files will go to the home path.

  An advantage of the HFS service over the TFTP service is that only the absolute path where the firmware files are located needs to be configured in telephony-service configuration mode.

  For example:

  ```
  Router(config-telephony)# hfs home-path flash:/cme/loads/
  ```

  In contrast, the TFTP service requires that each file be explicitly bound to its URL using the following **tftp-server** command:

  ```
  tftp-server flash: SCCP70.8-3-3-14S.loads
  ```

The method is inefficient because this step must be repeated for each file that needs to be fetched using the TFTP server.

For information on verifying HFS file bindings, see Example for Verifying the HFS File Bindings of Cisco Unified SIP IP Phone Configuration and Firmware Files, on page 66.

For information on how to configure the home path, see Configure HFS Home Path for SIP Phone Firmware Files, on page 53.

Locale Installer

Installing and configuring locale files in Cisco Unified CME when using an HTTP server is the same as when using a TFTP server.

For configuration information, see Use the Locale Installer in Cisco Unified CME 7.0(1) and Later Versions.

Security Recommendations

Like any access interface, the HFS download service can open router files that should only be accessed by authorized persons. Security issues are made more severe by the fact that the HFS download service is HTTP based, enabling anyone with a simple web browser to access sensitive files, such as configuration or image files, by entering a random string of words.

However, the HFS security problem is restricted to the loose binding operation, where the administrator provides an HFS home path in which the phone firmware and other related files are stored.

In the case where a unique directory path (where only the phone firmware files are stored) is used as the HFS home path

```
(config-telephony)# hfs home-path flash:/cme/loads/
```

only those files that are in flash:/cme/loads/ can be accessed.

But when it is the root directory path that is used as the HFS home path

```
(config-telephony)# hfs home-path flash:/
```

there is a risk of making configuration files and system images, which are stored in the root directory shared with the phone firmware files, accessible to unauthorized persons.

The following are two recommendations on how to make firmware files inaccessible to unauthorized persons:

- Create a unique directory, which is not shared by any other application or used for any other purpose, fpr IP phone firmware files. Using a root directory as the HFS home path is not recommended.
- Use the **ip http access-class** command to specify the access list that should be used to restrict access to the HTTP server. Before the HTTP server accepts a connection, it checks the access list. If the check fails, the HTTP server does not accept the request for a connection.

Redundant Cisco Unified CME Router for SCCP Phones

A second Cisco Unified CME router can be configured to provide call-control services if the primary Cisco Unified CME router fails. The secondary Cisco Unified CME router provides uninterrupted services until the primary router becomes operational again.

When a phone registers to the primary router, it receives a configuration file from the primary router. Along with other information, the configuration file contains the IP addresses of the primary and secondary Cisco Unified CME routers. The phone uses these addresses to initiate a keepalive (KA) message to each

router. The phone sends a KA message after every KA interval (30 seconds by default) to the router with which it is registered and after every two KA intervals (60 seconds by default) to the other router. The KA interval can be adjusted.

If the primary router fails, a phone will not receive an acknowledgment (ACK) to its KA message to the primary router. If the phone does not get an ACK from the primary router for three consecutive KAs, it registers with the secondary Cisco Unified CME router.

During the time that the phone is registered to the secondary router, it keeps sending a KA probe to the primary router to see if it has come back up, now every 60 seconds by default or two times the normal KA interval. After the primary Cisco Unified CME router returns to normal operation, the phone starts receiving ACKs for its probes. After the phone receives ACKs from the primary router for three consecutive probes, it switches back to the primary router and re-registers with it. The re-registration of phones with the primary router is also called rehoming.

The physical setup for redundant Cisco Unified CME routers is as follows. The FXO line from the PSTN is split using a splitter. From the splitter, one line goes to the primary Cisco Unified CME router and the other line goes to the secondary Cisco Unified CME router. When a call comes in on the FXO line, it is presented to both the primary and secondary Cisco Unified CME routers. The primary router is configured by default to answer the call immediately. The secondary Cisco Unified CME router is configured to answer the call after three rings. If the primary router is operational, it answers the call immediately and changes the call state so that the secondary router does not try to answer it. If the primary router is unavailable and does not answer the call, the secondary router sees the new call coming in and answers after three rings.

The secondary Cisco Unified CME router should be connected in some way on the LAN, either through the same switch or through another switch that may or may not be connected to the primary Cisco Unified CME router directly. As long as both routers and the phones are connected on the LAN with the appropriate configurations in place, the phones can register to whichever router is active.

Configure primary and secondary Cisco Unified CME routers identically, with the exception that the FXO voice port from the PSTN on the secondary router should be configured to answer after more rings than the primary router, as previously explained. The same command is used on both routers to specify the IP addresses of the primary and secondary routers.

For configuration information, see Configure Redundant Router for SCCP Phones, on page 36.



Restriction

- Due to lack of High Availability support, Stateful Swtichover or preservation of active calls is not supported in the redundancy feature offered by Unified CME.
- The physical setup for redundant Cisco Unified CME routers only support Loop start signaling. The Ground start signaling is not supported.

Redundant Cisco Unified CME Router for SIP Phones

A secondary Cisco Unified CME router can be configured to provide call-control services if the primary Cisco Unified CME router fails. The secondary Cisco Unified CME router provides uninterrupted services until the primary router becomes operational again.

When a SIP phone registers to the primary router, it receives a configuration file from the primary router. Along with other information, the configuration file contains the IP addresses of the primary and secondary Cisco Unified CME routers. The phone uses these addresses to initiate a keepalive (KA) message to the secondary CME router. The phone sends a REGISTER message to the primary router for registration and a

keepalive REGISTER message with Expires=0, to the secondary router during the keepalive interval (every 120 seconds by default). The keepalive interval can be configured (Range is 120 to 65535).

If primary router fails, a SIP phone (on registration refresh) will not receive a successful response for its REGISTER message. On unsuccessful response from primary router, phone registers with the secondary router. When the phone is registered to the secondary router, phone sends keepalive REGISTER (Expires=0) messages to the primary router.

After the primary Cisco Unified CME router returns to normal operation, the phone sends a "token-registration" to the primary router seeking permission to move registration of the phone from the standby secondary router to the primary router. To obtain a token, the SIP phones sends a Out-of-Dialog REFER message to the primary router for registration. The primary router accepts the token by responding with a 202 Accepted response. When the SIP phones receive the token (202 Accepted response) from the primary router, the phones will immediately de-register from the secondary router by sending a REGISTER message with Expires=0 for each line and registers back to the primary router. The re-registration of phones with the primary router is called rehoming.

No signaling or media preservation is done for any active calls on Unified CME. Hence during failover on primary CME, calls would remain in active state. But media would not be present for those calls. The SIP phones will not register to the secondary router until the active call is disconnected.

The secondary Cisco Unified CME router is connected directly to the same SIP trunk as the primary Cisco Unified CME router. As long as both routers and the phones are connected on the LAN with the appropriate configurations in place, the phones can register to whichever router is active. You should configure the primary and secondary Cisco Unified CME routers identically. The same command is used on both routers to specify the IP addresses of the primary and secondary routers.

For configuration information, see Configure Redundant Router for SIP Phones, on page 38.



Restriction

- Due to lack of High Availability support, Stateful Swtichover or preservation of active calls is not supported in the redundancy feature offered by Unified CME.

Timeouts

The following system-level timeout parameters have default values that are generally adequate:

- Busy Timeout—Length of time that can elapse after a transferred call reaches a busy signal before the call is disconnected.
- Interdigit Timeout—Length of time that can elapse between the receipt of individual dialed digits before the dialing process times out and is terminated. If the timeout ends before the destination is identified, a tone sounds and the call ends. This value is important when using variable-length dial-peer destination patterns (dial plans).
- Ringing Timeout—Length of time a phone can ring with no answer before returning a disconnect code to the caller. This timeout is used only for extensions that do not have no-answer call forwarding enabled. The ringing timeout prevents hung calls received over interfaces, such as FXO, that do not have forward-disconnect supervision.
- Keepalive—Interval determines how often a message is sent between the router and Cisco Unified IP phones, over the session, to ensure that the keepalive timeout is not exceeded. If no other traffic is sent over the session during the interval, a keepalive message is sent.

For configuration information, see Modify Defaults for Timeouts for SCCP Phones, on page 34.

IPv6 Support for Cisco Unified CME SCCP Endpoints

Internet Protocol version 6 (IPv6), which is the latest version of the Internet Protocol (IP) that uses packets to exchange data, voice, and video traffic over digital networks, increases the number of network address bits from 32 bits in IPv4 to 128 bits. IPv6 support in Cisco Unified CME allows the network to behave transparently in a dual-stack (IPv4 and IPv6) environment and provides additional IP address space to SCCP phones and devices that are connected to the network. For information on configuring DHCP for IPv6, see Network Parameters.

Before Cisco Unified CME 8.0, SCCP supported IPv4 addresses (4 bytes) only. With Cisco Unified CME 8.0, the SCCP version is upgraded to store IPv6 address (16 bytes) also.

The following SCCP phones and devices are supported on IPv6: 7911, 7931, 7941G, 7941GE, 7961G, 7961GE, 7970G, 7971G, 7971G-GE, 7942, 7962, 7945, 7965, 7975, SCCP analogue gateway, Xcoder, and Hardware Conference devices. For more information on configuring SCCP IP phones for IPv6 source address, see Configure IPv6 Source Address for SCCP IP Phones, on page 20.



Note You must disable Alternative Network Address Transport (ANAT) globally for SIP lines if you have a Cisco Unified CME with a dual-stack SIP trunk and enable ANAT at dial-peer level for the SIP trunk.

Support for IPv4-IPv6 (Dual-Stack)

Cisco Unified CME 8.0 can interact with and support any SCCP devices that support IPv4 only or both IPv4 and IPv6 (dual-stack). In dual-stack mode, two IP addresses are assigned to an interface, one is an IPv4 address and the other is an IPv6 address. Both IPv4 and IPv6 stacks are enabled on the voice gateways so that applications can interact with both versions of IP addresses. To support devices that use IPv4 only, IPv6 only, or both IPv4 and IPv6 (dual-stack) addresses, you must ensure that the Cisco Unified CME has both IPv4 address and IPv6 address enabled. For more information, see Configure IP Phones in IPv4, IPv6, or Dual Stack Mode, on page 18.

Media Flow Through and Flow Around

Media transport modes, such as flow around and flow through, are used to transport media packets across endpoints. Media flow around enables media packets to pass directly between the endpoints, without the intervention of the IP-IP Gateway (IPIPGW). Media flow through enables media packets to pass through the endpoints, without the intervention of the IPIPGW.

Table 3: Call Flow Scenarios Between IPv4 only, IPv6 only, and Dual-Stack, on page 15 lists media flow-through and flow-around scenarios between endpoints that support IPv4, IPv6, and dual- stack. When both endpoints are IPv4 only or IPv6 only, the call flows around. When one endpoint is IPv4 and the other is IPv6, calls flow through. When one endpoint is dual-stack and the other IPv4 or IPv6 the calls flow around. When both endpoints are dual-stack calls flow around or follows the preference (preferred IP address version) selected by protocol mode in dual-stack.

Table 3: Call Flow Scenarios Between IPv4 only, IPv6 only, and Dual-Stack

IP Versions	IPv4 Only	IPv6 Only	Dual-Stack
IPv4 Only	Flow Around [1]	Flow Through	Flow Around
IPv6 Only	Flow Through	Flow Around	Flow Around/IPv6
Dual-Stack	Flow Around/IPv4	Flow Around/IPv6	Flow Around/Preference

[1] When MTP is configured under ephones all the call flow-around scenarios change to flow-through. This is also applicable to cross-VRF endpoints.

Media Flow Around Support for SIP-SIP Trunk Calls

Cisco Unified CME 8.5 and later versions support the media flow around functionality for SIP to SIP trunk calls on Cisco Unified CME, allowing less consumption of resources on Cisco Unified CME.

The media flow around feature eliminates the need to terminate RTP and re-originate on Cisco Unified CME. This reduces media switching latency and increases the call handling capacity for a Cisco Unified CME SIP trunk.

Media flow around is supported in the following scenarios:

- Single Number Reach (SNR) Push—If an SNR call on a SIP trunk is pushed over to a mobile user over another SIP trunk, the resulting connection is a SIP-SIP trunk call connection. If both SIP trunks are configured for media flow around, the media is allowed to flow around Cisco Unified CME for the resulting call.
- Call Forward—If a SIP trunk call is forwarded over another SIP trunk and both the SIP trunks are configured for media flow around, media flows around Cisco Unified CME for the resulting SIP-SIP trunk call. Media flow around is supported for all types of call forwarding, such as call forward night-service, call forward all, call forward busy, and call forward no-answer.
- Call Transfer—If a SIP trunk call is transferred over another SIP trunk and both SIP trunks are configured for media flow around, media flows around Cisco Unified CME for the resulting SIP-SIP trunk call. Media flow around is supported on both SIP-line-initiated call transfer and SCCP-line-initiated call transfers. It is supported for all types of call transfers, such as blind transfer, consult transfer, and full consult transfer.

Media is forced to flow through on different types of call flows including the SIP to SIP trunk call with asymmetric flow mode configurations or symmetric flow through configuration. In asymmetric flow mode configurations, one SIP leg is configured in the media flow around mode and another SIP leg is configured in the media flow through mode. In such cases, media is forced to flow through Cisco Unified CME.

Media is forced to flow through Cisco Unified CME for the following types of call flows:

- Any calls involving a SIP endpoint, a SCCP endpoint, PSTN trunks (BRI/PRI/FXO), or FXO circuits.
- SIP to SIP trunk call with either asymmetric flow mode configurations or symmetric flow through configurations.
- SIP to SIP trunk call that requires transcoding services on Cisco Unified CME.

- SIP to SIP trunk calls that require DTMF interworking with RFC2833 on one side, and SIP-Notify on the other side.
- SNR pullback to SCCP— When an SNR call is pulled back from a mobile phone to the local SCCP SNR extension, the call is connected to the SCCP SNR extension. Media is required to flow through Cisco Unified CME because one of the calls is from a SCCP SNR extension, which is local to Cisco Unified CME.

In Cisco Unified CME 8.5, the media flow around feature is turned on or turned off using the **media** command in voice service voip, dial-peer voip, and voice class media configuration modes. The configuration specified under voice class media configuration mode takes precedence over the configuration in dial-peer configuration mode. If the media configuration is not specified under voice class media or dial-peer configuration mode, then the global configuration specified under voice service voip takes precedence. For more information, see Enable Media Flow Mode on SIP Trunks, on page 55.

Overlap Dialing Support for SIP and SCCP IP Phones

Cisco Unified CME 8.5 and later versions support overlap dialing on SCCP and SIP IP phones such as 7942, 7945, 7962, 7965, 7970, 7971, and 7975.

In earlier versions of Cisco Unified CME, overlap dialing was not supported over PRI/BRI trunks for calls originating from SCCP or SIP IP phones. Dialing was always converted into enbloc dialing based on the dial-peer configuration and the dial-peer mapping application. Once dialpeer matching took place, no further dialing was possible and no overlap digit were sent over ISDN trunk, even though overlap dialing was supported over ISDN trunks.

SCCP IP phones currently support overlap dialing, but digits are converted to enbloc digits when it reaches Cisco Unified CME. Overlap dialing is supported on SIP IP phones using the KeyPad Markup Language (KPML) method.

With overlap dialing support, the dialed digits from the SIP or SCCP IP phones are passed across to the PRI/BRI trunks as overlap digits and not as enbloc digits, enabling overlap dialing on the PRI/BRI trunks as well.

For information on how to configure SCCP and SIP IP phones for overlap dialing, see Configure Overlap Dialing on SCCP IP Phones, on page 42 and Configure Overlap Dialing on SIP Phones, on page 57.

Unsolicited Notify for Shared Line and Presence Events for Cisco Unified SIP IP Phones

Before Cisco Unified CME 9.0, a Cisco Unified SIP IP phone receives NOTIFY messages that convey shared line and presence events from the Cisco Unified CME only by subscribing to such events. To subscribe, the IP phone sends a SUBSCRIBE message to the Cisco Unified CME with the type of event for which it wants to be notified. The Cisco Unified CME sends a NOTIFY message to alert the subscribed IP phone or subscriber of event updates.

In Unsolicited Notify, the Cisco Unified CME acquires the required information from the router configuration to create the implicit subscription and adds subscribers without a subscription request from Cisco Unified SIP IP phones. The Cisco Unified CME sends out NOTIFY messages to the IP phones for shared line or presence updates.

In Cisco Unified CME 9.0 and later versions, the Unsolicited Notify mechanism reduces network traffic particularly during Cisco Unified SIP IP phone registration using the bulk registration method. Through this registration method, the preferred notification method of the IP phone is embedded in the registration message.

Note Configuring TCP as the transport layer protocol under voice register pool configuration mode enables bulk registration with negotiation for the Unsolicited Notify mechanism.

The Unsolicited Notify mechanism supports backward compatibility with all existing Cisco Unified SIP IP phone features. This mechanism is also the defacto notify mechanism in newer IP phone and Cisco Unified CME features, such as SNR Mobility.

From the end-user perspective, the following are the only two discernible differences between the SUBSCRIBE/NOTIFY and the Unsolicited Notify mechanisms:

- **show presence subscription** and **show shared-line** commands display different subscription IDs for each mechanism.
- With the SUBSCRIBE/NOTIFY mechanism, a Cisco Unified SIP IP phone needs to refresh the Cisco Unified CME subscription. In Unsolicited Notify mode, the subscription is permanent and does not need a refresh as long as the IP phone remains registered.

Restriction

- Because Unsolicited Notify is negotiated during bulk registration, the mechanism is not available on Cisco Unified SIP IP phones that do not have bulk registration turned on or have firmware that do not support bulk registration.
- Cisco Unified CME cannot disable the Unsolicited Notify mechanism. The system complies with and cannot override the requests of Cisco Unified SIP IP phones.
- In the absence of Cisco Unified SIP IP phone subscription information to distinguish if a notification event is for line or device monitoring, local device monitoring is not supported in the Unsolicited Notify mode.

Interface Support for Unified CME and Unified SRST

Unified CME and Unified SRST routers have multiple interfaces that are used for signaling and data packet transfers. The two types of interfaces available on a Cisco router include the physical interface and the virtual interface. The types of physical interfaces available on a router depends on its interface processors or port adapters. Virtual interfaces are software-based interfaces that you create in the memory of the networking device using Cisco IOS commands. When you need to configure a virtual interface for connectivity, you can use the Loopback Interface for Unified CME and Unified SRST.

The following interfaces are supported on Unified CME and Unified SRST:

- Gigabit Ethernet Interface (IEEE 802.3z) (**interface gigabitethernet**)
- Loopback Interface (**interface loopback**)
- Fast Ethernet Interface (**interface fastethernet**)

The remaining Cisco IOS interfaces are not validated on Unified CME and Unified SRST. Hence, Unified CME and Unified SRST do not claim support for these interfaces. For more information on the Cisco IOS Interface commands, see Cisco IOS Interface and Hardware Component Command Reference.

For physical interfaces such as **interface gigabitethernet** and **interface fastethernet**, subinterfaces are supported. In a subinterface, virtual interfaces are created by dividing a physical interface into multiple logical interfaces. For Cisco routers, a subinterface uses the parent physical interface for sending and receiving data. Virtual interfaces (For example, **interface loopback**) do not support subinterfaces.

A subinterface for **interface gigabitethernet** is configured as follows:

```
Router(config)#interface gigabitEthernet 0/0.1
Router(config-subif)#exit
Router(config)#exit
```

Configure System-Level Parameters

Configure IP Phones in IPv4, IPv6, or Dual Stack Mode



Restriction

- Legacy IP phones are not supported.
- Multicast MOH and multicast paging features are not supported on IPv6 only phones. If you want to receive paging calls on IPv6 enabled phones, use the default multicast paging.
- Primary and secondary CME need to be provisioned with the same network type.
- MWI relay server must be in IPv4 network.
- Presence server must be IPv4 only.
- Video endpoints, such as CUVA and 7985, are not supported in IPv6
- TAPI client is not supported in IPv6.
- All HTTP based IPv6 services are not supported.
- IOS TFTP server is not supported in IPv6.
- If protocol mode is IPv4, you can only configure IPv4 as the source IP-address, if protocol mode is IPv6 you can only configure IPv6 as the source IP address and if the protocol mode is dual-stack, you can configure both IPv4 and IPv6 source addresses.

Before you begin

- Cisco Unified CME 8.0 or later version.
- IPv6 CEF must be enabled for dual-stack configuration.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **telephony-service**
4. **protocol mode** {**ipv4** | **ipv6** | **dual-stack** [**preference** {**ipv4** | **ipv6**}]}

5. **end**

DETAILED STEPS

Procedure

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**telephony-service** **Example:** `Router(config)# telephony-service`	Enters telephony-service configuration mode.
Step 4	**protocol mode** {**ipv4** | **ipv6** | **dual-stack** [**preference** {**ipv4** | **ipv6**}]} **Example:** `Router(config-telephony)# protocol mode dual-stack preference ipv6`	Allows SCCP phones to interact with phones on IPv6 voice gateways. You can configure phones for IPv4 addresses, IPv6 address es, or for a dual-stack mode • **ipv4**—Allows you to set the protocol mode as an IPv4 address. • **ipv6**—Allows you to set the protocol mode as an IPv6 address. • **dual-stack—**Allows you to set the protocol mode for both IPv4 and IPv6 addresses. • **preference**—Allows you to choose a preferred IP address family if protocol mode is dual-stack.
Step 5	**end** **Example:** `Router(config-telephony)# end`	Returns to privileged EXEC mode.

Example

```
telephony-service
protocol mode dual-stack preference ipv6
....
ip source-address 10.10.2.1 port 2000
ip source-address 2000:A0A:201:0:F:35FF:FF2C:697D
```

Configure IPv6 Source Address for SCCP IP Phones

Restriction

- IPv6 option only appears if protocol mode is in dual-stack or 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 may not be able to handle the IPv6 SCCP phones and various packet handling problems may occur.

Before you begin

Cisco Unified CME 8.0 or a later version.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **telephony-service**
4. **ip source-address** {*ipv4 address* | *ipv6 address*} **port** *port* [**secondary** {*ipv4 address* | *ipv6 address* } [**rehome** *seconds*]] [**strict-match**]
5. **end**

DETAILED STEPS

Procedure

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**telephony-service** **Example:** `Router(config)# telephony-service`	Enters the telephony-service configuration mode.
Step 4	**ip source-address** {*ipv4 address* | *ipv6 address*} **port** *port* [**secondary** {*ipv4 address* | *ipv6 address* } [**rehome** *seconds*]] [**strict-match**] **Example:** `Router(config-telephony)# ip source-address 10.10.10.33 port 2000 ip source-address 2001:10:10:10::`	Allows to configure an IPv4 or IPv6 address as an IP source-address for phones to communicate with a Cisco Unified CME router. • *ipv4 address*—Allows phones to communicate with phones or voice gateways in an IPv4 network. *ipv4 address* can only be configured with an IPv4 address or a dual-stack mode.

	Command or Action	Purpose
		• *ipv6 address*—Allows phones to communicate with phones or voice gateways in an IPv6 network. *ipv6 address* can only be configured with an IPv6 address or a dual-stack mode. • (Optional) **port** *port*—TCP/IP port number to use for SCCP. Range is from 2000 to 9999. Default is 2000. For dual-stack, port is only configured with an IPv4 address. • (Optional) **secondary**—Cisco Unified CME router with which phones can register if the primary Cisco Unified CME router fails. • (Optional) **rehome** *seconds*—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 Communication Manager or Cisco Unified CME) before the phones re-register 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. • (Optional) **strict-match**— Requires strict IP address checking for registration.
Step 5	**end** **Example:** `outer(config-telephony)# end`	Returns to privileged EXEC mode.

Verify IPv6 and Dual-Stack Configuration

Procedure

Step 1 The following example shows a list of success messages that are printed during Cisco IOS boot up. These messages confirm whether IPv6 has been enabled on interfaces (for example, EDSP0.1 to EDSP0.5) specific to exchanging RTP packets with SCCP endpoints.

Example:

```
Router#
00:00:33: %EDSP-6-IPV6_ENABLED: IPv6 on interface EDSP0 added.
00:00:34: %EDSP-6-IPV6_ENABLED: IPv6 on interface EDSP0.1 added.
00:00:34: %EDSP-6-IPV6_ENABLED: IPv6 on interface EDSP0.2 added.
00:00:34: %EDSP-6-IPV6_ENABLED: IPv6 on interface EDSP0.3 added.
```

```
00:00:34: %EDSP-6-IPV6_ENABLED: IPv6 on interface EDSP0.4 added.
00:00:34: %EDSP-6-IPV6_ENABLED: IPv6 on interface EDSP0.5 added.
00:00:34: %LINEPROTO-5-UPDOWN: Line protocol on Interface FastEthernet0/1, changed state to down
00:00:34: %LINK-3-UPDOWN: Interface ephone_dsp DN 1.1, changed state to up
00:00:34: %LINK-3-UPDOWN: Interface ephone_dsp DN 1.2, changed state to up
.
```

Step 2 Use the **show ephone socket** command to verify if IPv4 only, IPv6 only, or dual-stack (IPv4/IPv6) is configured in Cisco Unified CME. In the following example, SCCP TCP listening socket (skinny_tcp_listen_socket fd) values 0 and 1 verify dual-stack configuration. When IPv6 only is configured, the **show ephone socket** command displays SCCP TCP listening socket values as (-1) and (0). The listening socket is closed if the value is (-1). When IPv4 only is configured, the **show ephone socket** command displays SCCP TCP listening socket values as (0) and (-1).

Example:

```
Router# show ephone socket
skinny_tcp_listen_socket fd = 0

skinny_tcp_listen_socket (ipv6) fd = 1


skinny_secure_tcp_listen_socket fd = -1

skinny_secure_tcp_listen_socket (ipv6) fd = -1


Phone 7,

skinny_sockets[15] fd = 16 [ipv6]

read_buffer 0x483C0BC4, read_offset 0, read_header N, read_length 0

resend_queue 0x47EC69EC, resend_offset 0, resend_flag N, resend_Q_depth 0

MTP 1,

skinny_sockets[16] fd = 17

read_buffer 0x483C1400, read_offset 0, read_header N, read_length 0

resend_queue 0x47EC6978, resend_offset 0, resend_flag N, resend_Q_depth 0

Phone 8,

skinny_sockets[17] fd = 18 [ipv6]

read_buffer 0x483C1C3C, read_offset 0, read_header N, read_length 0

resend_queue 0x47EC6904, resend_offset 0, resend_flag N, resend_Q_depth 0
```

Step 3 Use the **show ephone summary** command to verify the IPv6 or IPv4 addresses configured for ephones. The following example displays IPv6 and IPv4 addresses for different ephones:

Example:

```
Router# show ephone summary
ephone-2[1] Mac:0016.46E0.796A TCP socket:[7] activeLine:0 whisperLine:0 REGISTERED

mediaActive:0 whisper_mediaActive:0 startMedia:0 offhook:0 ringing:0 reset:0 reset_sent:0 debug:0
privacy:1 primary_dn: 1*

IPv6:2000:A0A:201:0:216:46FF:FEE0:796A* IP:10.10.10.12 7970 keepalive 599 music 0 1:1
```

```
sp1:2004

ephone-7[6] Mac:0013.19D1.F8A2 TCP socket:[6] activeLine:0 whisperLine:0 REGISTERED

mediaActive:0 whisper_mediaActive:0 startMedia:0 offhook:0 ringing:0 reset:0 reset_sent:0 debug:0
privacy:0 primary_dn: 13*

IP:10.10.10.14 * Telecaster 7940 keepalive 2817 music 0 1:13 2:28
```

Configure Bulk Registration

To configure bulk registration for registering a block of phone numbers with an external registrar so that calls can be routed to Cisco Unified CME from a SIP network, perform the following steps.

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

Note Use the **no reg** command to specify that an individual directory number should not register with the external registrar. For configuration information, see Disable SIP Proxy Registration for a Directory Number.

Before you begin

Cisco Unified CME 3.4 or a later version.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register global**
4. **mode** *cme*
5. **bulk** *number*
6. **exit**
7. **sip-ua**
8. **registrar** {**dns**: *address* | **ipv4**: *destination-address*} **expires** *seconds* [**tcp**] [**secondary**] **no registrar** [**secondary**]
9. **end**

DETAILED STEPS

Procedure

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice register global** **Example:** `Router(config)# voice register global`	Enters voice register global configuration mode to set parameters for all supported SIP phones in Cisco Unified CME.
Step 4	**mode** *cme* **Example:** `Router(config-register-global)# mode cme`	Enables mode for provisioning SIP phones in Cisco Unified CME.
Step 5	**bulk** *number* **Example:** `Router(config-register-global)# bulk 408526....`	Sets bulk registration for E.164 numbers that will register with a SIP proxy server. • *number*—Unique sequence of up to 32 characters, including wild cards and patterns that represents E.164 numbers that will register with a SIP proxy server.
Step 6	**exit** **Example:** `Router(config-register-pool)# exit`	Exits configuration mode to the next highest mode in the configuration mode hierarchy.
Step 7	**sip-ua** **Example:** `Router(config)# sip-ua`	Enters SIP user agent (UA) configuration mode for configuring the user agent.
Step 8	**registrar** {**dns**: *address* | **ipv4**: *destination-address*} **expires** *seconds* [**tcp**] [**secondary**] **no registrar** [**secondary**] **Example:** `Router(config-sip-ua)# registrar server ipv4:1.5.49.240`	Enables SIP gateways to register E.164 numbers with a SIP proxy server.
Step 9	**end** **Example:** `Router(config-sip-ua)# end`	Exits SIP UA configuration mode and enters privileged EXEC mode.

Examples

The following example shows that all phone numbers that match the pattern “408555...” can register with a SIP proxy server (IP address 1.5.49.240):

```
voice register global
 mode cme
 bulk 408555….
sip-ua
 registrar ipv4:1.5.49.240
```

Configure Bulk Registration for SIP IP Phones

Before you begin

- Cisco Unified CME 8.6 or a later version.
- Phone firmware 8.3 or a later version.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register pool** *tag*
4. **session-transport** {**tcp** | **udp**}
5. **number** *tag* **dn** *tag*
6. **end**

DETAILED STEPS

Procedure

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice register pool** *tag* **Example:** `Router(config)#voice register pool 20`	Enters voice register dn configuration mode to define a directory number for a SIP phone, intercom line, voice port, or an MWI.
Step 4	**session-transport** {**tcp**	**udp**} **Example:** `Router(config-register-pool)#session-transport tcp`	Specifies the transport layer protocol that a SIP phone uses to connect to Cisco Unified CME. • **tcp**—TCP is used for bulk registration.

	Command or Action	Purpose
		• **udp**—UDP is used for line registration.
Step 5	**number** *tag* **dn** *tag* **Example:** `Router(config-register-pool)#number 1 dn 2`	Associates a directory number with the SIP phone being configured. • **dn** *dn-tag*—Identifies the directory number for this SIP phone as defined by the voice register **dn** command.
Step 6	**end** **Example:** `Router(config-register-pool)# end`	Returns to privileged EXEC mode.

Verify Phone Registration Type and Status

You can verify phone registration type and status using the **show voice register pool** command. The following example shows that the Cisco Unified IP phone 7970 used the bulk registration method and completed the registration process:

```
Router#sh voice register pool 20
 Pool Tag 20
Config:
  Mac address is 001B.2A89.3698
  Type is 7970
  Number list 1 : DN 20
  Number list 2 : DN 2
  Number list 3 : DN 24
  Number list 4 : DN 4
  Number list 5 : DN 6
  Number list 6 : DN 7
  Number list 7 : DN 17
  Number list 8 : DN 23
  Proxy Ip address is 0.0.0.0
  Current Phone load version is Cisco-CP7970G/9.0.1
  DTMF Relay is enabled, rtp-nte, sip-notify
  Call Waiting is enabled
  DnD is disabled
  Video is disabled
  Camera is disabled
  Busy trigger per button value is 0
  speed-dial blf 1 6779 label 6779_device
  speed-dial blf 2 3555 label 3555_remote
  speed-dial blf 3 6130 label 6130
```

```
speed-dial blf 4 3222 label 3222_remote_dev
fastdial 1 1234
keep-conference is enabled
username johndoe password cisco
template is 1
kpml signal is enabled
Lpcor Type is none
Transport type is tcp
service-control mechanism is supported
Registration method: bulk - completed
registration Call ID is 001b2a89-3698017e-68646967-126b902e@28.18.88.33
Privacy is configured:  init status: ON, current status: ON
Privacy button is enabled
active primary line is: 6010
```

Set Up Cisco Unified CME for SCCP Phones

To identify filenames and the location of phone firmware for phone types to be connected, specify the port for phone registration, and specify the number of phones and directory numbers to be supported, perform the following steps.



Restriction DSCP requires Cisco Unified CME 7.1 or a later version. If DSCP is configured for the gateway interface using the **service-policy** command or for the dial peer using the ip qos dscp command, the value set with those commands takes precedence over the DSCP value configured in this procedure.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **tftp-server** *device*:*filename*
4. **telephony-service**
5. **load** *phone-type firmware-file*
6. **max-ephones** *max-phones*
7. **max-dn** *max-directory-numbers* [**preference** *preference-order*] [**no-reg primary** | **both**]
8. **ip source-address** *ip-address* [**port** *port*] [**any-match** | **strict-match**]
9. **ip qos dscp** {{*number* | *af* | *cs* | **default** | **ef**} {**media** | **service** | **signaling** | **video**}}
10. **end**

DETAILED STEPS

Procedure

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**tftp-server** *device*:*filename* **Example:** `Router(config)# tftp-server flash:P00307020300.bin`	(Optional) Creates TFTP bindings to permit IP phones served by the Cisco Unified CME router to access the specified file. • A separate **tftp-server** command is required for each phone type. • Required for Cisco Unified CME 7.0/4.3 and earlier versions. • Cisco Unified CME 7.0(1) and later versions: Required only if the location for cnf files is not flash or slot 0, such as system memory or a TFTP server url. Use the complete filename, including the file suffix, for phone firmware versions later than version 8.2(2) for all phone types.
Step 4	**telephony-service** **Example:** `Router(config)# telephony-service`	Enters telephony-service configuration mode.

Step 5	**load** *phone-type firmware-file*
	Example:
	`Router(config-telephony)# load 7960-7940 P00307020300`
		Identifies a Cisco Unified IP phone firmware file to be used by phones of the specified type when they register.
		• A separate **load** command is required for each IP phone type.
		• firmware-file—Filename is case-sensitive.
		  • Cisco Unified CME 7.0/4.3 and earlier versions: Do not use the .sbin or .loads file extension except for the Cisco ATA and Cisco Unified IP Phone 7905 and 7912.
		  • Cisco Unified CME 7.0(1) and later versions: Use the complete filename, including the file suffix, for phone firmware versions later than version 8.2(2) for all phone types.

	Command or Action	Purpose
		Note If you are loading a firmware file larger than 384 KB, you must first load a file for that phone type that is smaller than 384 KB and then load the larger file.
Step 6	**max-ephones** *max-phones* **Example:** `Router(config-telephony)# max-ephones 24`	Sets the maximum number of phones that can register to Cisco Unified CME. • Maximum number is platform and version-specific. Type **?** for range. • In Cisco Unified CME 7.0/4.3 and later versions, the maximum number of phones that can register is different from the maximum number of phones that can be configured. The maximum number of phones that can be configured is 1000. • In versions earlier than Cisco Unified CME 7.0/4.3, this command restricted the number of phones that could be configured on the router.
Step 7	**max-dn** *max-directory-numbers* [**preference** *preference-order*] [**no-reg primary** | **both**] **Example:** `Router(config-telephony)# max-dn 200 no-reg primary`	Limits number of directory numbers to be supported by this router. • Maximum number is platform and version-specific. Type **?** for value.
Step 8	**ip source-address** *ip-address* [**port** *port*] [**any-match** | **strict-match**] **Example:** `Router(config-telephony)# ip source-address 10.16.32.144`	Identifies the IP address and port number that the Cisco Unified CME router uses for IP phone registration. • **port** *port*—(Optional) TCP/IP port number to use for SCCP. Range is 2000 to 9999. Default is 2000. • **any-match**—(Optional) Disables strict IP address checking for registration. This is the default. • **strict-match**—(Optional) Instructs the router to reject IP phone registration attempts if the IP server address used by the phone does not exactly match the source address.
Step 9	**ip qos dscp** {{*number* | *af* | *cs* | **default** | **ef**} {**media** | **service** | **signaling** | **video**}} **Example:** `Router(config-telephony)# ip qos dscp af43 video`	Sets the DSCP priority levels for different types of traffic.
Step 10	**end** **Example:** `Router(config-telephony)# end`	Returns to privileged EXEC mode.

Examples

The following example shows different DSCP settings for media, signaling, video, and services enabled with the ip qos dscp command:

```
telephony-service
load 7960-7940 P00308000500
max-ephones 100
max-dn 240
ip source-address 10.10.10.1 port 2000
ip qos dscp af11 media
ip qos dscp cs2 signal
ip qos dscp af43 video
ip qos dscp 25 service
cnf-file location flash:
.
.
```

Set Date and Time Parameters for SCCP Phones

To specify the format of the date and time that appears on all SCCP phones in Cisco Unified CME, perform the following steps.



Note For certain phones, such as the Cisco Unified IP Phones 7906, 7911, 7931, 7941, 7942, 7945, 7961, 7962, 7965, 7970, 7971, and 7975, you must configure the **time-zone** command to ensure that the correct time stamp appears on the phone display. This command is not required for Cisco Unified IP Phone 7902G, 7905G, 7912G, 7920, 7921, 7935, 7936, 7940, 7960, or 7985G.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **telephony-service**
4. **date-format** {**dd-mm-yy** | **mm-dd-yy** |**yy-dd-mm** | **yy-mm-dd**}
5. **time-format** {**12** | **24**}
6. **time-zone** *number*
7. **end**

DETAILED STEPS

Procedure

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.

	Command or Action	Purpose
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**telephony-service** **Example:** `Router(config)# telephony-service`	Enters telephony-service configuration mode.
Step 4	**date-format** {**dd-mm-yy** | **mm-dd-yy** |**yy-dd-mm** | **yy-mm-dd**} **Example:** `Router(config-telephony)# date-format yy-mm-dd`	(Optional) Sets the date format for phone display. • Default: **mm-dd-yy**.
Step 5	**time-format** {**12** | **24**} **Example:** `Router(config-telephony)# time-format 24`	(Optional) Selects a 12-hour or 24-hour clock for the time display format on phone display. • Default: **12**.
Step 6	**time-zone** *number* **Example:** `Router(config-telephony)# time-zone 2`	Sets time zone for SCCP phones. • Not required for Cisco Unified IP Phone 7902G, 7905G, 7912G, 7920, 7921, 7935, 7936, 7940, 7960, or 7985G. • Default: 5, Pacific Standard/Daylight Time (-480).
Step 7	**end** **Example:** `Router(config-telephony)# end`	Returns to privileged EXEC mode.

Block Automatic Registration for SCCP Phones

Before you begin

Cisco Unified CME 4.0 or a later version.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **telephony-service**
4. **no auto-reg-ephone**
5. **end**

DETAILED STEPS

Procedure

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**telephony-service** **Example:** `Router(config)# telephony-service`	Enters telephony-service configuration mode.
Step 4	**no auto-reg-ephone** **Example:** `Router(config-telephony)# no auto-reg-ephone`	Disables automatic registration of Cisco Unified IP phones that are running SCCP but are not explicitly configured in Cisco Unified CME. • Default: Enabled.
Step 5	**end** **Example:** `Router(config-telephony)# end`	Returns to privileged EXEC mode.

Define Per-Phone Configuration Files and Alternate Location for SCCP Phones

Restriction

- TFTP does not support file deletion. When configuration files are updated, they overwrite any existing configuration files with the same name. If you change the configuration file location, files are not deleted from the TFTP server.
- Generating configuration files on flash memory or slot 0 memory can take up to a minute, depending on the number of files being generated.
- For smaller routers such as the Cisco 2600 series routers, you must manually enter the **squeeze** command to erase files after changing the configuration file location or entering any commands that trigger the deletion of configuration files. Unless you use the **squeeze** command, the space used by the moved or deleted configuration files is not usable by other files.
- If VRF Support on Cisco Unified CME is configured and the **cnf-file location** command is configured for system:, the per phone or per phone type file for an ephone in a VRF group is created in *system:/its/vrf<group-tag>/*. The vrf directory is automatically created and appended to the TFTP path. No action is required on your part. Locale files are still created in system:/its/.
- If VRF Support on Cisco Unified CME is configured and the **cnf-file location** command is configured as **flash:** or **slot0:**, the per phone or per phone type file for an ephone in a VRF group is named *flash:/its/vrf<group-tag>_<filename> or slot0:/its/vrf<group-tag>_filename>*. The vrf directory is automatically created and appended to the TFTP path. No action is required on your part. The location of the locale files is not changed.

To define a location other than system:/its for storing configuration files for per-phone and per-phone type configuration files, perform the following steps.

Before you begin

- Cisco Unified CME 4.0 or a later version.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **telephony-service**
4. **cnf-file location** {**flash:** | **slot0:** | **tftp** *tftp-url*}
5. **cnf-file** {**perphonetype** | **perphone**}
6. **end**

DETAILED STEPS

Procedure

	Command or Action	Purpose
Step 1	**enable**	Enables privileged EXEC mode.

	Command or Action	Purpose
	Example: `Router> enable`	• Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**telephony-service** **Example:** `Router(config)# telephony-service`	Enters telephony-service configuration mode.
Step 4	**cnf-file location** {**flash:** | **slot0:** | **tftp** *tftp-url*} **Example:** `Router(config-telephony)# cnf-file location flash:`	Specifies a location other than system:/its for storing phone configuration files. • Required for per-phone or per-phone type configuration files.
Step 5	**cnf-file** {**perphonetype** | **perphone**} **Example:** `Router(config-telephony)# cnf-file perphone`	Specifies whether to use a separate file for each type of phone or for each individual phone. • Required if you configured the **cnf-file location** command.
Step 6	**end** **Example:** `Router(config-telephony)# end`	Returns to privileged EXEC mode.

Example

The following example selects flash memory as the configuration file storage location and per-phone as the type of configuration files that the system generates:

```
telephony-service
 cnf-file location flash:
 cnf-file perphone
```

What to do next

If you changed the configuration file storage location, use the **option 150 ip** command to update the address. See Change the TFTP Address on a DHCP Server.

Modify Defaults for Timeouts for SCCP Phones

To configure values for system-level intervals for which default values are typically adequate, perform the following steps.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **telephony-service**
4. **timeouts busy** *seconds*
5. **timeouts interdigit** *seconds*
6. **timeouts ringing** *seconds*
7. **keepalive** *seconds*
8. **end**

DETAILED STEPS

Procedure

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**telephony-service** **Example:** `Router(config)# telephony-service`	Enters telephony-service configuration mode.
Step 4	**timeouts busy** *seconds* **Example:** `Router(config-telephony)# timeouts busy 20`	(Optional) Sets the length of time after which calls that are transferred to busy destinations are disconnected. • *seconds*—Number of seconds. Range is 0 to 30. Default is 10.
Step 5	**timeouts interdigit** *seconds* **Example:** `Router(config-telephony)# timeouts interdigit 30`	(Optional) Configures the interdigit timeout value for all Cisco Unified IP phones attached to the router. • *seconds*—Number of seconds before the interdigit timer expires. Range is 2 to 120. Default is 10.
Step 6	**timeouts ringing** *seconds* **Example:** `Router(config-telephony)# timeouts ringing 30`	(Optional) Sets the duration, in seconds, for which the Cisco Unified CME system allows ringing to continue if a call is not answered. Range is 5 to 60000. Default is 180.
Step 7	**keepalive** *seconds* **Example:** `Router(config-telephony)# keepalive 45`	(Optional) Sets the time interval, in seconds, between keepalive messages that are sent to the router by Cisco Unified IP phones.

	Command or Action	Purpose
		• The default is usually adequate. If the interval is set too large, it is possible for notification to be delayed when a system goes down. • Range: 10 to 65535. Default: 0.
Step 8	**end** **Example:** `Router(config-telephony)# end`	Returns to privileged EXEC mode.

Configure Redundant Router for SCCP Phones

Before you begin

- Cisco Unified CME 4.0 or a later version.
- The secondary router's running configuration must be identical to that of the primary router.
- The physical configuration of the secondary router must be as described in Redundant Cisco Unified CME Router for SCCP Phones, on page 11.
- Phones that use this feature must be configured with the **type** command, which guarantees that the appropriate phone configuration file will be present.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **telephony-service**
4. **ip source-address** *ip-address* [**port** *port*] [**secondary** *ip-address* [**rehome** *seconds*]] [**any-match** | **strict-match**]
5. **exit**
6. **voice-port** *slot-number* / *port*
7. **signal ground-start**
8. **incoming alerting ring-only**
9. **ring number** *number*
10. **end**

DETAILED STEPS

Procedure

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.

	Command or Action	Purpose
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**telephony-service** **Example:** `Router(config)# telephony-service`	Enters telephony-service configuration mode.
Step 4	**ip source-address** *ip-address* [**port** *port*] [**secondary** *ip-address* [**rehome** *seconds*]] [**any-match** | **strict-match**] **Example:** `Router(config-telephony)# ip source-address 10.0.0.1 port 2000 secondary 10.2.2.25`	Identifies the IP address and port number that the primary Unified CME router uses for IP phone registration. • *ip-address*—Address of the primary Unified CME router. • **port** *port*—(Optional) TCP/IP port number to use for SCCP. Range is 2000 to 9999. Default is 2000. • **secondary** *ip-address*—Indicates a backup Unified CME router. • **rehome** *seconds*—Not used by Unified CME. Used only by phones registered to Cisco Unified SRST. • **any-match**—(Optional) Disables strict IP address checking for registration. This is the default. • **strict-match**—(Optional) Router rejects IP phone registration attempts if the IP server address used by the phone does not exactly match the source address.
Step 5	**exit** **Example:** `Router(config-telephony)# exit`	Exits telephony-service configuration mode.
Step 6	**voice-port** *slot-number* / *port* **Example:** `Router(config)# voice-port 2/0`	Enters voice-port configuration mode for the FXO voice port for DID calls from the PSTN.
Step 7	**signal ground-start** **Example:** `Router(config-voiceport)# signal ground-start`	Specifies ground-start signaling for a voice port.
Step 8	**incoming alerting ring-only** **Example:** `Router(config-voiceport)# incoming alerting ring-only`	Instructs the FXO ground-start voice port to detect incoming calls by detecting incoming ring signals.
Step 9	**ring number** *number* **Example:** `Router(config-voiceport)# ring number 3`	(Required only for the secondary router) Sets the maximum number of rings to be detected before answering an incoming call over an FXO voice port. • *number*—Number of rings detected before answering the call. Range is 1 to 10. Default is 1.

	Command or Action	Purpose
		Note For an incoming FXO voice port on a secondary Cisco Unified CME router, set this value higher than is set on the primary router. We recommend setting this value to 3 on the secondary router.
Step 10	**end** **Example:** `Router(config-voiceport)# end`	Returns to privileged EXEC mode.

Configure Redundant Router for SIP Phones

Before you begin

- Cisco Unified CME 11.6 or a later version.
- Auto-register configuration is recommended only on the primary router.
- XML interface for secondary backup router is configured. See Configure the XML Interface for the Secondary Backup Router, on page 40.

Note It is recommended to configure the XML interface for a seamless failover from primary to secondary Cisco Unified CME. Else, there is delay in the phones getting registered to secondary Cisco Unified CME due to mismatch in the configuration version timestamp.

- Ensure that you configure version stamp synchronization on the primary router. See Configure Version Stamp Synchronization on the Primary Router, on page 39.

Note It is recommended to configure version stamp synchronization for a seamless failover from primary to secondary Cisco Unified CME. Else, there is delay in the phones getting registered to secondary Cisco Unified CME.

Restriction

- Active calls are not supported when switchover happens from primary router to the secondary router.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register global**
4. **source-address** *ip-address* [**port** *port*] [**secondary** *ip-address*]

5. **keepalive** *seconds*
6. **end**

DETAILED STEPS

Procedure

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice register global** **Example:** `Router(config)# voice register global`	Enters voice register global configuration mode.
Step 4	**source-address** *ip-address* [**port** *port*] [**secondary** *ip-address*] **Example:** `Router(config-register-global)# source-address 10.6.21.4 port 6000 secondary 10.6.50.6`	Identifies the IP address and port number that the Cisco Unified CME router uses for IP phone registration. • *ip-address*—Address of the primary Cisco Unified CME router. • **port** *port*—(Optional) TCP/IP port number to use for SIP. Range is 2000 to 9999. Default is 5060 for SIP. • **secondary** *ip-address*—Indicates a backup Cisco Unified CME router.
Step 5	**keepalive** *seconds* **Example:** `Router(config-register-global)# keepalive 200`	Sets the length of the time interval between successive keepalive messages from the SIP phones to Cisco Unified CME router. Default is 120 seconds.
Step 6	**end** **Example:** `Router(config-register-global)# end`	Returns to privileged EXEC mode.

Configure Version Stamp Synchronization on the Primary Router

To configure the primary router to enable automatic synchronization of 'version stamp' with secondary backup router, perform the following steps.

Tip All phone-related configurations are tagged with a 'version stamp' that indicates when the last configuration change was made.

Before you begin

- XML interface for secondary backup router is configured. See Configure the XML Interface for the Secondary Backup Router, on page 40.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **telephony-service**
4. **standby username** *username* **password** *password*
5. **end**

DETAILED STEPS

Procedure

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**telephony-service** **Example:** `Router(config)# telephony-service`	Enters telephony service configuration mode.
Step 4	**standby username** *username* **password** *password* **Example:** `Router(config-telephony)# standby username user23 password 3Rs92uzQ`	Defines an authorized user. • Same username and password that is defined in Configure the XML Interface for the Secondary Backup Router, on page 40.
Step 5	**end** **Example:** `Router(config-telephony)# end`	Returns to privileged EXEC mode.

Configure the XML Interface for the Secondary Backup Router

To configure the secondary backup router to activate the XML interface required to receive "version stamp" configuration change information from the primary router, perform the following steps.

Restriction

- Automatic synchronization for new or replacement routers is not supported.

Before you begin

- The XML interface, provided through the Cisco IOS XML Infrastructure (IXI), must be configured. See Configuring the XML API.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **telephony-service**
4. **xml user** *user-name* **password** *password privilege-level*
5. **end**

DETAILED STEPS

Procedure

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**telephony-service** **Example:** `Router(config)# telephony-service`	Enters telephony service configuration mode.
Step 4	**xml user** *user-name* **password** *password privilege-level* **Example:** `Router(config-telephony)# xml user user23 password 3Rs92uzQ 15`	Defines an authorized user. • *user-name*—Username of the authorized user. • *password*—Password to use for access. • *privilege-level*—Level of access to Cisco IOS commands to be granted to this user. Only the commands with the same or a lower level can be executed via XML. Range is 0 to 15.
Step 5	**end** **Example:**	Returns to privileged EXEC mode.

	Command or Action	Purpose
	`Router(config-telephony)# end`	

Configure Overlap Dialing on SCCP IP Phones

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **telephony-service**
4. **overlap-signal**
5. **exit**
6. **ephone** *phone-tag*
7. **overlap-signal**
8. **exit**
9. **ephone-template** *template-tag*
10. **overlap-signal**
11. **end**

DETAILED STEPS

Procedure

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**telephony-service** **Example:** `Router(config)telephony-service`	Enters telephony-service configuration mode.
Step 4	**overlap-signal** **Example:** `Router(config-telephony)#overlap-signal`	Allows to configure overlap signaling support for SCCP IP phones.
Step 5	**exit** **Example:** `Router(config-telephony)#exit`	Exits telephony-service configuration mode.

	Command or Action	Purpose
Step 6	**ephone** *phone-tag* **Example:** `Router(config)ephone 10`	Enters ephone configuration mode.
Step 7	**overlap-signal** **Example:** `Router(config-ephone)overlap-signal`	Applies overlap signaling support for ephone.
Step 8	**exit** **Example:** `Router(config-ephone)exit`	Exits ephone configuration mode.
Step 9	**ephone-template** *template-tag* **Example:** `Router(config)ephone-template 10`	Enters ephone-template configuration mode.
Step 10	**overlap-signal** **Example:** `Router(config-ephone-template)#overlap-signal`	Applies overlap signaling support to ephone template.
Step 11	**end** **Example:** `Router(config-ephone-template)# end`	Returns to privileged EXEC mode.

Set Up Cisco Unified CME for SIP Phones

To identify filenames and location of phone firmware for phone types to be connected, to specify the port for phone registration, and to specify the number of phones and directory numbers to be supported, perform the following steps.

Note If your Cisco Unified CME system supports SCCP and SIP phones, do not connect your SIP phones to your network until after you have verified the configuration profile for the SIP phone.

Note From Cisco IOS XE Amsterdam 17.2.1r onwards, **cme-app** mode was added for ISR4321 routers. This mode allows configuration of up to 200 phones for routers that are dedicated to CME use only. The **cme** or **cme-app** modes configure SIP phones and features for standalone call control use.

Restriction

- SIP endpoints are not supported on H.323 trunks. SIP endpoints are supported on SIP trunks only.
- Certain Cisco Unified IP phones, such as the Cisco Unified IP Phones 7911G, 7941G, 7941GE, 7961G, 7961GE, 7970G, and 7971GE, are supported only in Cisco Unified CME 4.1 and later versions.
- DSCP requires Cisco Unified CME 7.1 or a later version. If DSCP is configured for the gateway interface using the **service-policy** command or for the dial peer using the **ip qos dscp** command, the value set with those commands takes precedence over the DSCP value configured in this procedure.

Before you begin

Cisco CME 3.4 or a later version.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register global**
4. **mode** [**cme** | **cme-app**]
5. **source-address** *ip-address* [**port** *port*]
6. **load** *phone-type firmware-file*
7. **tftp-path** {**flash**: | **slot0:** | **tftp:**//*url*}
8. **max-pool** *max-phones*
9. **max-dn** *max-directory-numbers*
10. **authenticate** [**all**][**realm** *string*]
11. **ip qos dscp** {{*number* | *af* | *cs* | **default** | **ef**} {**media** | **service** | **signaling** | **video**}}
12. **end**

DETAILED STEPS

Procedure

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice register global** **Example:** `Router(config)# voice register global`	Enters voice register global configuration mode to set parameters for all supported SIP phones in Cisco Unified CME.

	Command or Action	Purpose
Step 4	**mode** [**cme** | **cme-app**] **Example:** `Router(config-register-global)# mode cme`	Enables mode for provisioning SIP phones in Cisco Unified CME.
Step 5	**source-address** *ip-address* [**port** *port*] **Example:** `Router(config-register-global)# source-address 10.6.21.4`	Enables the Cisco Unified CME router to receive messages from SIP phones through the specified IP address and port. • **port** *port*—(Optional) TCP/IP port number. Range: 2000 to 9999. Default: 2000.
Step 6	**load** *phone-type firmware-file* **Example:** `Router(config-register-global)# load 7960-7940 P0S3-07-3-00`	Associates a phone type with a phone firmware file. • A separate load command is required for each phone type.
Step 7	**tftp-path** {**flash**: | **slot0:** | **tftp:**//*url*} **Example:** `Router(config-register-global)# tftp-path http://mycompany.com/files`	(Optional) Defines a location, other than system memory, from which the SIP phones will download configuration profile files. • Default: system memory (system:/cme/sipphone/).
Step 8	**max-pool** *max-phones* **Example:** `Router(config-register-global)# max-pool 10`	Sets maximum number of SIP phones to be supported by the Cisco Unified CME router. • Version- and platform-dependent; type ? for range. • In Cisco CME 3.4 to Cisco Unified CME 7.0: Default is maximum number supported by platform. • In Cisco Unified CME 7.0(1) and later versions: Default is 0.
Step 9	**max-dn** *max-directory-numbers* **Example:** `Router(config-register-global)# max-dn 20`	(Optional) Sets maximum number of directory numbers for SIP phones to be supported by the Cisco Unified CME router. • Required for Cisco Unified CME 7.0(1) and later versions. • In Cisco Unified CME 7.0(1) and later versions: Default is 0. Range is 1 to maximum number supported by platform. Type **?** for range. • In Cisco CME 3.4 to Cisco Unified CME 7.0: Default is 150 or maximum allowed on platform. Type ? for value.
Step 10	**authenticate** [**all**][**realm** *string*] **Example:** `Router(config-register-global)# authenticate all realm company.com`	(Optional) Enables authentication for registration requests in which the MAC address of the SIP phone cannot be identified by using other methods.

	Command or Action	Purpose
Step 11	**ip qos dscp** {{*number* | *af* | *cs* | **default** | **ef**} {**media** | **service** | **signaling** | **video**}} **Example:** `Router(config-register-global)# ip qos dscp af43 video`	Sets the DSCP priority levels for different types of traffic.
Step 12	**end** **Example:** `Router(config-register-global)# end`	Exits voice register global configuration mode and enters privileged EXEC mode.

Set Up Cisco Unified CME for SIP Phones

To identify filenames and location of phone firmware for phone types to be connected, to specify the port for phone registration, and to specify the number of phones and directory numbers to be supported, perform the following steps.

Note If your Cisco Unified CME system supports SCCP and SIP phones, do not connect your SIP phones to your network until after you have verified the configuration profile for the SIP phone.

Note From Cisco IOS XE Amsterdam 17.2.1r onwards, **cme-app** mode was added for ISR4321 routers. This mode allows configuration of up to 200 phones for routers that are dedicated to CME use only. The **cme** or **cme-app** modes configure SIP phones and features for standalone call control use.

Restriction

- SIP endpoints are not supported on H.323 trunks. SIP endpoints are supported on SIP trunks only.
- Certain Cisco Unified IP phones, such as the Cisco Unified IP Phones 7911G, 7941G, 7941GE, 7961G, 7961GE, 7970G, and 7971GE, are supported only in Cisco Unified CME 4.1 and later versions.
- DSCP requires Cisco Unified CME 7.1 or a later version. If DSCP is configured for the gateway interface using the **service-policy** command or for the dial peer using the **ip qos dscp** command, the value set with those commands takes precedence over the DSCP value configured in this procedure.

Before you begin

Cisco CME 3.4 or a later version.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register global**

4. **mode** [**cme** | **cme-app**]
5. **source-address** *ip-address* [**port** *port*]
6. **load** *phone-type firmware-file*
7. **tftp-path** {**flash**: | **slot0:** | **tftp://***url*}
8. **max-pool** *max-phones*
9. **max-dn** *max-directory-numbers*
10. **authenticate** [**all**][**realm** *string*]
11. **ip qos dscp** {{*number* | *af* | *cs* | **default** | **ef**} {**media** | **service** | **signaling** | **video**}}
12. **end**

DETAILED STEPS

Procedure

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice register global** **Example:** `Router(config)# voice register global`	Enters voice register global configuration mode to set parameters for all supported SIP phones in Cisco Unified CME.
Step 4	**mode** [**cme**	**cme-app**] **Example:** `Router(config-register-global)# mode cme`	Enables mode for provisioning SIP phones in Cisco Unified CME.
Step 5	**source-address** *ip-address* [**port** *port*] **Example:** `Router(config-register-global)# source-address 10.6.21.4`	Enables the Cisco Unified CME router to receive messages from SIP phones through the specified IP address and port. • **port** *port*—(Optional) TCP/IP port number. Range: 2000 to 9999. Default: 2000.
Step 6	**load** *phone-type firmware-file* **Example:** `Router(config-register-global)# load 7960-7940 P0S3-07-3-00`	Associates a phone type with a phone firmware file. • A separate load command is required for each phone type.
Step 7	**tftp-path** {**flash**:	**slot0:**	**tftp://***url*} **Example:** `Router(config-register-global)# tftp-path http://mycompany.com/files`	(Optional) Defines a location, other than system memory, from which the SIP phones will download configuration profile files. • Default: system memory (system:/cme/sipphone/).

	Command or Action	Purpose
Step 8	**max-pool** *max-phones* **Example:** `Router(config-register-global)# max-pool 10`	Sets maximum number of SIP phones to be supported by the Cisco Unified CME router. • Version- and platform-dependent; type ? for range. • In Cisco CME 3.4 to Cisco Unified CME 7.0: Default is maximum number supported by platform. • In Cisco Unified CME 7.0(1) and later versions: Default is 0.
Step 9	**max-dn** *max-directory-numbers* **Example:** `Router(config-register-global)# max-dn 20`	(Optional) Sets maximum number of directory numbers for SIP phones to be supported by the Cisco Unified CME router. • Required for Cisco Unified CME 7.0(1) and later versions. • In Cisco Unified CME 7.0(1) and later versions: Default is 0. Range is 1 to maximum number supported by platform. Type **?** for range. • In Cisco CME 3.4 to Cisco Unified CME 7.0: Default is 150 or maximum allowed on platform. Type ? for value.
Step 10	**authenticate** [**all**][**realm** *string*] **Example:** `Router(config-register-global)# authenticate all realm company.com`	(Optional) Enables authentication for registration requests in which the MAC address of the SIP phone cannot be identified by using other methods.
Step 11	**ip qos dscp** {{*number* | *af* | *cs* | **default** | **ef**} {**media** | **service** | **signaling** | **video**}} **Example:** `Router(config-register-global)# ip qos dscp af43 video`	Sets the DSCP priority levels for different types of traffic.
Step 12	**end** **Example:** `Router(config-register-global)# end`	Exits voice register global configuration mode and enters privileged EXEC mode.

Set Date and Time Parameters for SIP Phones

Before you begin

- Cisco CME 3.4 or a later version.
- **mode cme** command is enabled.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register global**
4. **timezone** *number*
5. **date-format** [**d/m/y** | **m/d/y** | **y-d-m** |**y/d/m** | **y/m/d** | **yy-m-d**]
6. **time-format** {**12** | **24**}
7. **dst auto-adjust**
8. **dst** {**start** | **stop**} *month* [**day** day-of-month | **week** week-number | **day** day-of-week] **time** *hour:minutes*
9. **end**

DETAILED STEPS

Procedure

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice register global** **Example:** `Router(config)# voice register global`	Enters voice register global configuration mode to set parameters for all supported SIP phones in Cisco Unified CME.
Step 4	**timezone** *number* **Example:** `Router(config-register-global)# timezone 8`	Selects the time zone used for SIP phones in Cisco Unified CME. • Default: **5**, Pacific Standard/Daylight Time. Type **?** to display a list of time zones.
Step 5	**date-format** [**d/m/y** | **m/d/y** | **y-d-m** |**y/d/m** | **y/m/d** | **yy-m-d**] **Example:** `Router(config-register-global)# date-format yy-m-d`	(Optional) Selects the date display format on SIP phones in Cisco Unified CME. • Default: **m/d/y**.
Step 6	**time-format** {**12** | **24**} **Example:** `Router(config-register-global)# time-format 24`	(Optional) Selects the time display format on SIP phones in Cisco Unified CME. • Default: **12**.
Step 7	**dst auto-adjust** **Example:**	(Optional) Enables automatic adjustment of Daylight Saving Time on SIP phones in Cisco Unified CME.

	Command or Action	Purpose
	`Router(config-register-global)# dst auto-adjust`	• To modify start and stop times for daylight savings time, use the **dst** command.

Step 8	**dst** {**start** | **stop**} *month* [**day** day-of-month | **week** week-number | **day** day-of-week] **time** *hour:minutes*
	Example:
	`Router(config-register-global)# dst start jan day 1 time 00:00`
	`Router(config-register-global)# dst stop mar day 31 time 23:59`
		(Optional) Sets the time period for Daylight Saving Time on SIP phones in Cisco Unified CME.
		• Required if automatic adjustment of Daylight Saving Time is enabled by using the **dst auto-adjust** command.
		• Default is Start: First week of April, Sunday, 2:00 a.m. Stop: Last week of October, Sunday 2:00 a.m.

Step 9	**end** **Example:** `Router(config-register-global)# end`	Returns to privileged EXEC mode.

Set Network Time Protocol for SIP Phones

To enable Network Time Protocol (NTP) for certain phones, such as the Cisco Unified IP Phones 7911G, 7941G, 7941GE, 7961G, 7961GE, 7970G, and 7971GE, connected to Cisco Unified CME running SIP, perform the following steps.

Before you begin

- Cisco Unified CME 4.1 or a later version.
- The firmware load 8.2(1) or a later version is installed for SIP phones to download. For upgrade information, see Upgrade or Downgrade SIP Phone Firmware.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register global**
4. **ntp-server** *ip-address* [**mode** {**anycast** | **directedbroadcast** | **multicast** | **unicast**}]
5. **end**

DETAILED STEPS

Procedure

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.

	Command or Action	Purpose
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice register global** **Example:** `Router(config)# voice register global`	Enters voice register global configuration mode to set global parameters for all supported SIP phones in a Cisco Unified CME environment.
Step 4	**ntp-server** *ip-address* [**mode** {**anycast** | **directedbroadcast** | **multicast** | **unicast**}] **Example:** `Router(config-register-global)# ntp-server 10.1.2.3`	Synchronizes clock on this router with the specified NTP server.
Step 5	**end** **Example:** `Router(config-register-global)# end`	Returns to privileged EXEC mode.

Enable HFS Download Service for SIP Phones

Restriction

- Only Cisco Unified 8951, 9951, and 9971 SIP IP Phones are supported.
- No IPv6 support for the HFS download service.

Before you begin

Cisco Unified CME 8.8 or a later version.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **ip http server**
4. **ip http port** *number*
5. **voice register global**
6. **mode cme**
7. **load** *phone-type firmware-file*
8. **create profile**
9. **exit**
10. **telephony-service**
11. **hfs enable** [**port** *port-number*]
12. **end**

DETAILED STEPS

Procedure

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**ip http server** **Example:** `Router(config)# ip http server`	Enables the underlying IOS HTTP server of the HFS infrastructure.
Step 4	**ip http port** *number* **Example:** `Router(config)# ip http port 60`	(Optional) Specifies the port where the HTTP service is run.
Step 5	**voice register global** **Example:** `Router(config)# voice register global`	Enters voice register global configuration mode to set global parameters for all supported Cisco SIP IP phones in a Cisco Unified CME.
Step 6	**mode cme** **Example:** `Router(config-register-global)# mode cme`	Enables the mode for configuring SIP IP phones in a Cisco Unified CME system.
Step 7	**load** *phone-type firmware-file* **Example:** `Router(config-register-global)# load 3951 SIP51.9.2.1S`	Associates a type of SIP IP phone with a phone firmware file.
Step 8	**create profile** **Example:** `Router(config-register-global)# create profile`	Generates the configuration profile files required for SIP IP phones.
Step 9	**exit** **Example:** `Router(config-register-global)# exit`	Exits voice register global configuration mode.
Step 10	**telephony-service** **Example:** `Router (config)# telephony-service`	Enters telephony-service configuration mode for configuring Cisco Unified CME.

	Command or Action	Purpose
Step 11	**hfs enable** [**port** *port-number*] **Example:** `Router(config-telephony)# hfs enable port 5678`	Enables the HFS download service on a specified port. • **port** *port-number*—(Optional) Specifies the port where the HFS download service is enabled. Range is from 1024 to 65535. Port 80 is the default port. Port 6970 is the custom port. **Note** If the entered custom HFS port clashes with the underlying IP HTTP port, an error message is displayed and the command is disallowed.
Step 12	**end** **Example:** `Router(config-telephonyl)# end`	Exits to privileged EXEC mode.

Troubleshooting HFS Download Service

The **debug cme-hfs** command can be used to troubleshoot an attempt to download Cisco Unified SIP IP phone configuration and firmware files using the HFS service.

Configure HFS Home Path for SIP Phone Firmware Files

To configure a home path where any requested Cisco Unified SIP IP Phone firmware file that has no explicit binding can be searched and fetched using the HFS download service, perform the following steps.

Restriction

- Only Cisco 8951, 9951, and 9971 SIP IP Phones are supported.
- No IPv6 support for the HFS download service.

Before you begin

Cisco Unified CME 8.8 or a later version.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **ip http server**
4. **ip http port** *number*
5. **telephony-service**
6. **hfs enable** [**port** *port-number*]
7. **hfs home-path** *path*
8. **end**

DETAILED STEPS

Procedure

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**ip http server** **Example:** `Router(config)# ip http server`	Enables the underlying IOS HTTP server of the HFS infrastructure.
Step 4	**ip http port** *number* **Example:** `Router(config)# ip http port 1234`	Specifies the port where the HTTP service is run.
Step 5	**telephony-service** **Example:** `Router (config)# telephony-service`	Enters telephony-service configuration mode for configuring Cisco Unified CME.
Step 6	**hfs enable** [**port** *port-number*] **Example:** `Router(config-telephony)# hfs enable port 6970`	Enables the HFS download service on a specified port.
Step 7	**hfs home-path** *path* **Example:** `Router(config-telephony)# hfs home-path flash:/cme/loads/`	Sets a home path directory for Cisco Unified SIP IP phone firmware files that can be searched and fetched using the HFS download service. **Note** The administrator must store the phone firmware files at the location set as the home path directory.
Step 8	**end** **Example:** `Router(config-telephony)# end`	Exits to privileged EXEC mode.

Change Session-Level Application for SIP Phones

Before you begin

Cisco CME 3.4 or a later version.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register global**
4. **application** *application-name*
5. **end**

DETAILED STEPS

Procedure

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice register global** **Example:** `Router(config)# voice register global`	Enters voice register global configuration mode to set parameters for all supported SIP phones in Cisco Unified CME.
Step 4	**application** *application-name* **Example:** `Router(config-register-global)# application sipapp2`	(Optional) Changes the default application for all dial peers associated with the SIP phones in Cisco Unified CME to the specified application. **Note** This command can also be configured in voice register pool configuration mode. The value set in voice register pool configuration mode has priority over the value set in voice register global mode.
Step 5	**end** **Example:** `Router(config-register-global)# end`	Exits voice register global configuration mode and enters privileged EXEC mode.

Enable Media Flow Mode on SIP Trunks

Restriction

- If any media service (like transcoding and conferencing) is needed for SIP to SIP trunk call, at least one of the SIP trunks must be placed in flow through mode.
- If media needs to flow through Cisco Unified CME for voicemail calls, the SIP trunk going towards the voicemail must be in flow through mode.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **media** [**flow around** | **flow through**]
5. **exit**
6. **dial-peer voice** *tag* **voip**
7. **media** {[**flow-around** | **flow-through**] **forking**}
8. **exit**
9. **voice class media** tag
10. **media** {[**flow-around** | **flow-through**] **forking**}
11. **end**

DETAILED STEPS

Procedure

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice service voip** **Example:** `Router(config)#voice service voip`	Enters voice service voip configuration mode.
Step 4	**media** [**flow around**	**flow through**] **Example:** `Router(conf-voi-serv)#media flow-around`	Enables global media setting for VoIP calls. • **flow around**—Allows the media to flow around the gateway. • **flow through**—Allows the media to flow through the gateway.
Step 5	**exit** **Example:** `Router(config-voi-ser)#exit`	Exits voice service voip configuration mode.
Step 6	**dial-peer voice** *tag* **voip** **Example:** `Router(config)#dial-peer voice 222 voip`	Enters dial-peer configuration mode to define a VoIP dial peer for the voice-mail system. • *tag*—Defines the dial peer being configured. Range is 1 to 1073741823.

	Command or Action	Purpose
Step 7	**media** {[**flow-around** | **flow-through**] **forking**} **Example:** `Router(config-dial-peer)#media flow-around`	Enables media settings for voice dial-peer. • **flow-around**—Allows the media to flow around the gateway. • **flow-through**—Allows the media to flow through the gateway. • **forking**—Enables media forking.
Step 8	**exit** **Example:** `Router(config-ephone)exit`	Exits voip dial-peer configuration mode.
Step 9	**voice class media** tag **Example:** `Router(config)#voice class media 10`	Enters voice class media configuration mode. • *tag*— Defines the voice class media tag being configured. Range is from 1 to 10000.
Step 10	**media** {[**flow-around** | **flow-through**] **forking**} **Example:** `Router(config-class)#media flow-around`	Enables media settings for voice dial-peer. • **flow-around**—Allows the media to flow around the gateway. • **flow-through**—Allows the media to flow through the gateway. • **forking**—Enables media forking.
Step 11	**end** **Example:** `Router(config-class)# end`	Returns to privileged EXEC mode.

Configure Overlap Dialing on SIP Phones

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register global**
4. **overlap-signal**
5. **exit**
6. **voice register pool** *pool-tag*
7. **overlap-signal**
8. **exit**
9. **voice register template** *template tag*
10. **overlap-signal**
11. **end**

DETAILED STEPS

Procedure

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice register global** **Example:** `Router(config)voice register global`	Enters voice register global configuration mode to set parameters for all supported SIP phones in Cisco Unified CME.
Step 4	**overlap-signal** **Example:** `Router(config-register-pool)overlap-signal`	Allows to configure overlap signaling support for SIP IP phones.
Step 5	**exit** **Example:** `Router(config-register-pool)exit`	Exits voice register pool configuration mode.
Step 6	**voice register pool** *pool-tag* **Example:** `Router(config)voice register pool 10`	Enters voice register pool configuration mode to set phone-specific parameters for a SIP phone.
Step 7	**overlap-signal** **Example:** `Router(config-register-global)overlap-signal`	Enables overlap signaling support for voice register global.
Step 8	**exit** **Example:** `Router(config-register-global)exit`	Exits voice register-template configuration mode.
Step 9	**voice register template** *template tag* **Example:** `Router(config)voice register template 5`	Enters voice register-template configuration mode to create an ephone template. • *template-tag*—Unique identifier for the ephone template that is being created. Range: 1 to 10.
Step 10	**overlap-signal** **Example:** `Router(config-register-temp) overlap-signal`	Applies overlap signaling support for voice register-template.

	Command or Action	Purpose
Step 11	**end** **Example:** `Router(config-register-temp)# end`	Returns to privileged EXEC mode.

Configuration Examples for System-Level Parameters

Example for Bulk Registration Support for SIP Phones

The following example shows TCP and UDP configured for various phones. Notice that in Bulk Registration (TCP), only the primary directory number is displayed, while in Line Registration (UDP), all directory numbers are displayed.

```
Router# show sip-ua status registrar
Line          destination      expires(sec)  contact
transport     call-id
              peer
============================================================
1001          21.1.1.138       112           21.1.1.138
TCP           239665429027943@21.1.1.138
              40015
1009          21.1.1.138       118           21.1.1.138
UDP           23967173002794 5@21.1.1.138
              40019
1010          21.1.1.138       118           21.1.1.138
UDP           23967174512794 5@21.1.1.138
              40021
```

Example for IPv6 Support on Cisco Unified CME

```
!
ip source-route
!
!ip cef
no ip dhcp use vrf connected
ip dhcp excluded-address 10.10.10.1 10.10.10.9
ip dhcp excluded-address 192.168.2.1
ipv6 unicast-routing
ipv6 cef
ntp server 223.255.254.254
```

```
multilink bundle-name authenticated
isdn switch-type primary-5ess
!
voice service voip
allow-connections h323 to h323
allow-connections h323 to sip
allow-connections sip to h323
allow-connections sip to sip
fax protocol cisco
sip
registrar server expires max 1200 min 300
!
!
!
voice register dn  1
number 2016
allow watch
name SIP-7961GE
label SIP2016
!
voice register dn  2
number 2017
!
!
voice logout-profile 1
!
voice logout-profile 2
number 2001 type normal
speed-dial 1 2004 label "7960-1"
!
interface GigabitEthernet0/0
ip address 10.10.10.2 255.255.255.0
duplex auto
speed auto
ipv6 address 2000:A0A:201:0:F:35FF:FF2C:697D/64
ipv6 enable
interface GigabitEthernet0/1
ip address 40.10.30.1 255.255.255.0
shutdown
duplex auto
```

```
speed auto
ipv6 address 2000::1/64
ipv6 address 2000::2/64
ipv6 address 2000::A/64
ipv6 address 3000::1/64
ipv6 address 4000::1/64
ipv6 address 9000::1/64
ipv6 address F000::1/64
ipv6 enable
!
i!
!
!
ip http server
!
ipv6 route 2001:20:20:20::/64 2000:A0A:201:0:F:35FF:FF2C:5
ipv6 route 2001:50:50:50::/64 2000:A0A:201:0:F:35FF:FF2C:5
!
tftp-server flash:P00308000500.bin
tftp-server flash:P00308000500.loads
p-server flash:cvm70sccp.8-5-2FT1-18.sbn
!
!
voice-port 0/0/0:23
!
!
mgcp fax t38 ecm
!
sccp local GigabitEthernet0/0
sccp ccm 10.10.10.2 identifier 1 version 7.0
sccp ccm 2000:A0A:201:0:F:35FF:FF2C:697D identifier 2 version 7.0
sccp
!
!
gateway
timer receive-rtp 1200
!
sip-ua
protocol mode dual-stack preference ipv6
!
```

```
!
telephony-service
protocol mode dual-stack preference ipv6
sdspfarm conference mute-on 111 mute-off 222
sdspfarm units 2
sdspfarm transcode sessions 20
sdspfarm tag 1 xcoder
sdspfarm tag 2 conference
conference hardware
no auto-reg-ephone
em logout 0:0 0:0 0:0
max-ephones 52
max-dn 192
ip source-address 10.10.10.2 port 2000
ip source-address 2000:A0A:201:0:F:35FF:FF2C:697D
service phone settingsAccess 1
service phone spanTOPCPort 0
timeouts transfer-recall 15
system message MOTO-CME1
url directories http://10.10.10.2:80/localdirectory
cnf-file location flash:
cnf-file perphone
load 7914 S00103020003
load 7911 SCCP11.8-5-2FT1-18S
load 7970 SCCP70.8-5-2FT1-18S
time-zone 5
max-conferences 4 gain -6
call-forward pattern .T
web admin system name cisco password cisco
web admin customer name admin password admin
transfer-system full-consult
```

Example for System-Level Parameters

The following example shows the system-level configuration for a Cisco Unified CME that can support up to 500 directory numbers on 100 phones. It sets up TFTP file sharing for phone firmware files for Cisco Unified IP Phones 7905, 7912, 7914, 7920, 7940, and 7960 and it loads those files.

```
tftp-server flash:ATA030100SCCP040211A.zup
! ATA 186/188 firmware
tftp-server flash:CP7902080001SCCP051117A.sbin
! 7902 firmware
```

```
tftp-server flash:CP7905080001SCCP051117A.sbin
! 7905 firmware
tftp-server flash:CP7912080001SCCP051117A.sbin
! 7912 firmware
tftp-server flash:cmterm_7920.4.0-02-00.bin
! 7914 firmware
tftp-server flash:P00503010100.bin
! 7920 firmware
tftp-server flash:S00104000100.sbn
! 7935 firmware
tftp-server flash:cmterm_7936.3-3-5-0.bin
! 7936 firmware
tftp-server flash:P0030702T023.bin
tftp-server flash:P0030702T023.loads
tftp-server flash:P0030702T023.sb2
! 7960/40 firmware
!
telephony-service
 max-ephones 100
 max-dn 500
 load ata ATA030100SCCP040211A
 load 7902 CP7902080001SCCP051117A
 load 7905 CP7905080001SCCP051117A
 load 7912 CP7912080001SCCP051117A
 load 7914 S00104000100
 load 7920 cmterm_7920.4.0-02-00
 load 7935 P00503010100
 load 7936 cmterm_7936.3-3-5-0
 load 7960-7940 P0030702T023
 ip source-address 10.16.32.144 port 2000
 create cnf-files version-stamp Jan 01 2002 00:00:00
 transfer-system full-consult
```

Cisco Unified IP Phone 7911, 7941, 7941-GE, 7961, 7961-GE, 7970, and 7971 require multiple files to be shared using TFTP. The following configuration example adds support for these phones.

```
tftp-server flash:SCCP11.7-2-1-0S.loads
tftp-server flash:term11.default.loads
tftp-server flash:apps11.1-0-0-72.sbn
tftp-server flash:cnu11.3-0-0-81.sbn
tftp-server flash:cvm11.7-2-0-66.sbn
```

```
tftp-server flash:dsp11.1-0-0-73.sbn
tftp-server flash:jar11.7-2-0-66.sbn
! 7911 firmware
!
tftp-server flash:TERM41.7-0-3-0S.loads
tftp-server flash:TERM41.DEFAULT.loads
tftp-server flash:TERM61.DEFAULT.loads
tftp-server flash:CVM41.2-0-2-26.sbn
tftp-server flash:cnu41.2-7-6-26.sbn
tftp-server flash:Jar41.2-9-2-26.sbn
! 7941/41-GE, 7961/61-GE firmware
!
tftp-server flash:TERM70.7-0-1-0s.LOADS
tftp-server flash:TERM70.DEFAULT.loads
tftp-server flash:TERM71.DEFAULT.loads
tftp-server flash:CVM70.2-0-2-26.sbn
tftp-server flash:cnu70.2-7-6-26.sbn
tftp-server flash:Jar70.2-9-2-26.sbn
! 7970/71 firmware
!
telephony-service
 load 7911 SCCP11.7-2-1-0S
 load 7941 TERM41.7-0-3-0S
 load 7961 TERM41.7-0-3-0S
 load 7941GE TERM41.7-0-3-0S
 load 7961GE TERM41.7-0-3-0S
 load 7970 TERM70.7-0-1-0s
 load 7971 TERM70.7-0-1-0s
 create cnf-files version-stamp Jan 01 2002 00:00:00
.
.
.
```

Example for Blocking Automatic Registration

The following example shows how to disable automatic ephone registration, display a log of attempted registrations, and then clear the log:

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

```
Router(config)# exit

Router# show ephone attempted-registrations


Attempting Mac address:


Num Mac Address DateTime DeviceType

---------------------------------------------------------------------------


1 C863.8475.5417 22:52:05 UTC Thu Apr 28 2005 SCCP Gateway (AN)

2 C863.8475.5408 22:52:05 UTC Thu Apr 28 2005 SCCP Gateway (AN)

.....

25 000D.28D7.7222 22:26:32 UTC Thu Apr 28 2005 Telecaster 7960

26 000D.BDB7.A9EA 22:25:59 UTC Thu Apr 28 2005 Telecaster 7960

...

47 C863.94A8.D40F 22:52:17 UTC Thu Apr 28 2005 SCCP Gateway (AN)

48 C863.94A8.D411 22:52:18 UTC Thu Apr 28 2005 SCCP Gateway (AN)


49 C863.94A8.D400 22:52:15 UTC Thu Apr 28 2005 SCCP Gateway (AN)


Router# clear telephony-service ephone-attempted-registrations
```

Example for Enabling the HFS Download Service for Cisco Unified SIP IP Phone

The following example shows how to enable the HFS download service:

```
Router(config)# ip http server
Router(config)# ip http port 1234
Router (config)# telephony-service
Router(config-telephony)# hfs enable port 65500
```

Example for Configuring an HFS Home Path for Cisco Unified SIP IP Phone Firmware Files

The following example shows how a new directory called phone-load can be created under the root directory of the flash memory and set as the hfs home-path:

```
cassini-c2801#mkdir flash:phone-loads
Create directory filename [phone-loads]?
Created dir flash:phone-loads
cassini-c2801#sh flash:
-#- --length-- -----date/time------ path
```

```
1     13932728 Mar 22 2007 15:57:38 +00:00 c2801-ipbase-mz.124-1c.bin
2     33510140 Sep 18 2010 01:21:56 +00:00 rootfs9951.9-0-3.sebn
3       143604 Sep 18 2010 01:22:20 +00:00 sboot9951.111909R1-9-0-3.sebn
4         1249 Sep 18 2010 01:22:40 +00:00 sip9951.9-0-3.loads
5        66996 Sep 18 2010 01:23:00 +00:00 skern9951.022809R2-9-0-3.sebn
6        10724 Sep 18 2010 00:59:48 +00:00 dkern9951.100609R2-9-0-3.sebn
7      1507064 Sep 18 2010 01:00:24 +00:00 kern9951.9-0-3.sebn
8            0 Jan 5 2011 02:03:46 +00:00 phone-loads
14819328 bytes available (49192960 bytes used)
cassini-c2801#conf t
Enter configuration commands, one per line.  End with CNTL/Z.
cassini-c2801(config)#tele
cassini-c2801(config)#telephony-service
cassini-c2801(config-telephony)#hfs hom
cassini-c2801(config-telephony)#hfs home-path flash:?
WORD
cassini-c2801(config-telephony)#hfs home-path flash:phone-loads
cassini-c2801(config-telephony)#
```

Example for Verifying the HFS File Bindings of Cisco Unified SIP IP Phone Configuration and Firmware Files

The following is a sample output from the **show voice register hfs** command:

```
Router(config)#show voice register hfs
Fetch Service Enabled = Y
   App enabled port = 6970
   Use default port = N
   Registered session-id = 19

Default home path = flash:/
   Ongoing fetches from home = 0

HTTP File Server Bindings
   No. of bindings = 11
   No. of url table entries = 9
```

No. of alias table entries = 9

Example for Redundant Router for SCCP Phones

The following example is configured on the primary Cisco Unified CME router. It establishes the router at 10.5.2.78 as a secondary router. The voice port 3/0/0 is the FXO port for incoming calls from the PSTN. It is set to use ground-start signaling and to detect incoming calls by counting incoming ring signals.

```
telephony-service
 ip source-address 10.0.0.1 port 2000 secondary 10.5.2.78


voice-port 3/0/0
 signal ground-start
 incoming alerting ring-only
```

The secondary Cisco Unified CME router is configured with the same commands, except that the ring number command is set to 3 instead of using the default of 1.

```
telephony-service
 ip source-address 10.0.0.1 port 2000 secondary 10.5.2.78


voice-port 3/0/0
 signal ground-start
 incoming alerting ring-only
 ring number 3
```

Example for Redundant Router for SIP Phones

The following example is configured on the primary Cisco Unified CME router. It establishes the router at 10.6.50.6 as a secondary router with keepalive value set to 200 seconds.



Note For the synchronization to happen, additional configurations are needed. These configurations such as IXI, HTTP, and telephony-service are provided in the output.

```
voice register global
 source-address 10.6.21.4 port 6000 secondary 10.6.50.6
 keepalive 200

ip http server

ixi transport http
 response size 8
 no shutdown
 request outstanding 2
 request timeout 30

ixi application cme
 no shutdown
 response timeout -1

telephony-service
 ip source-address 10.6.21.4 secondary 10.6.50.6
 standby user cisco password cisco123
```

The secondary Cisco Unified CME router is configured with the same commands:

```
voice register global
 source-address 10.6.21.4 port 6000 secondary 10.6.50.6
 keepalive 200

ip http server

ixi transport http
 response size 8
 no shutdown
 request outstanding 2
 request timeout 30

ixi application cme
 no shutdown
 response timeout -1

telephony-service
```

```
 ip source-address 10.6.50.6
 xml user cisco password cisco123 15
```

Example for Media Flow Around Mode for SIP Trunks

The following example shows media flow-around enabled in voice service voip, voice class media, and dial peer configuration modes:

```
Router# show running config

!

!

voice service voip

ip address trusted list

ipv4 20.20.20.1

media flow-around

allow-connections sip to sip

vpn-group 1

vpn-gateway 1 https://9.10.60.254/SSLVPNphone

vpn-trustpoint 1 trustpoint cme_cert root

vpn-hash-algorithm sha-1

vpn-profile 1

keepalive 50

auto-network-detect enable

host-id-check disable

vpn-profile 2

mtu 1300

authen-method both

password-persistent enable

host-id-check enable

vpn-profile 4

fail-connect-time 50

sip

!

voice class media 10

media flow-around

!
```

```
!
!
dspfarm profile 1 conference
codec g711ulaw
maximum sessions 2
associate application SCCP
!
dial-peer voice 222 voip
media flow-around
!
dial-peer voice 10 voip
media flow-around
!
dial-peer voice 101 voip
end
```

Example for Configuring Overlap Dialing for SCCP IP Phones

The following example shows the **overlap-signal** command configured in telephony-service configuration mode, ephone template 10, and ephone 10:

```
The following example shows the overlap-signal command configured in telephony-service
configuration mode, ephone template 10, and ephone 10:
Router# show running config
!
!
telephony-service
max-ephones 25
max-dn 15
load 7906 SCCP11.8-5-3S.loads
load 7911 SCCP11.8-5-3S.loads
load 7921 CP7921G-1.3.3.LOADS
load 7941 SCCP41.8-5-3S.loads
load 7942 SCCP42.8-5-3S.loads
load 7961 SCCP41.8-5-3S.loads
```

```
load 7962 SCCP42.8-5-3S.loads
max-conferences 12 gain -6
web admin system name cisco password cisco
transfer-system full-consult
create cnf-files version-stamp Jan 01 2002 00:00:00
overlap-signal
!
ephone-template 1
button-layout 1 line
button-layout 3-6 blf-speed-dial
!
ephone-template 9
feature-button 1 Endcall
feature-button 3 Mobility
!
!
ephone-template 10
feature-button 1 Park
feature-button 2 MeetMe
feature-button 3 CallBack
button-layout 1 line
button-layout 2-4 speed-dial
button-layout 5-6 blf-speed-dial
overlap-signal
!
ephone 10
device-security-mode none
mac-address 02EA.EAEA.0010
overlap-signal
```

Example for Configuring Overlap Dialing for SIP IP Phones

The following example shows the **overlap-signal** configured in voice register global configuration mode and voice register pool 10:

```
Router# show running config
!
!
!
voice service voip
ip address trusted list
ipv4 20.20.20.1
media flow-around
allow-connections sip to sip
!
voice class media 10
media flow-around
!
!
voice register global
max-pool 10
overlap-signal
!
voice register pool 5
overlap-signal
!
!
!
```

Where to Go Next

After configuring system-level parameters, you are ready to configure phones for making basic calls in Cisco Unified CME.

- To use Extension Assigner to assign extension numbers to the phones in your Cisco Unified CME, see Create Phone Configurations Using Extension Assigner.
- Otherwise, see Configure Phones to Make Basic Call.

Feature Information for System-Level Parameters

The following table provides release information about the feature or features described in this module. This table lists only the software release that introduced support for a given feature in a given software release train. Unless noted otherwise, subsequent releases of that software release train also support that feature.

Use Cisco Feature Navigator to find information about platform support and Cisco software image support. To access Cisco Feature Navigator, go to https://cfnng.cisco.com/. An account on Cisco.com is not required.

Table 4: Feature Information for System-Level Parameters

Feature Name	Cisco Unified CME Versions	Feature Information
Redundant Router for SIP Phones	11.6	Introduces redundant router support for SIP phones.
Unsolicited Notify for Shared Line and Presence Events for Cisco Unified SIP IP Phones	9.0	Allows the Unsolicited Notify mechanism to reduce network traffic during Cisco Unified SIP IP phone registration using the bulk registration method.
HFS Download Support for IP Phone Firmware and Configuration Files	8.8	Provides download support for SIP and SCCP IP phone firmware, scripts, midlets, and configuration files using the HTTP File-Fetch Server (HFS) infrastructure.
Bulk Registration	8.6/3.4	Introduces bulk registration support for SIP phones. Introduces bulk registration for registering a block of phone numbers with an external registrar.
Media Flow Around for SIP-SIP Trunks	8.5	Introduces the media flow around feature, which eliminates the need to terminate RTP and re-originate on Cisco Unified CME, reducing media switching latency and increasing the call handling capacity for Cisco Unified CME SIP trunk.
Overlap Dialing for SCCP and SIP Phones	8.5	Allows the dialed digits from the SIP or SCCP IP phones to pass across the PRI/BRI trunks as overlap digits and not as enbloc digits, enabling overlap dialing on the PRI/BRI trunks.
DSCP	7.1	Supports DSCP packet marking for Cisco Unified IP Phones to specify the class of service for each packet.
Maximum Ephones	7.0/4.3	The **max-ephones** command sets the maximum number of SCCP phones that can register to Cisco Unified CME, without limiting the number that can be configured. Maximum number of phones that can be configured is 1000.
Network Time Protocol for SIP Phones	4.1	Allows SIP phones to synchronize to an NTP server.

Feature Name	Cisco Unified CME Versions	Feature Information
Blocking Automatic Registration	4.0	Blocks IP phones that are not explicitly configured in Cisco Unified CME from registering.
Per-Phone Configuration Files and Alternate Location	4.0	Defines a location other than system for storing configuration files and specifies the type of configuration files to generate.
Redundant Router for SCCP Phones	4.0	Introduces redundant router capability.
SIP phones in Cisco Unified CME	3.4	Introduces support for SIP endpoints directly connected to Cisco Unified CME.