

Call Transfer and Forward

- Information About Call Transfer and Forward, on page 1
- Configure Call Transfer and Forwarding, on page 28
- Configuration Examples for Call Transfer and Forwarding, on page 72
- Where to Go Next, on page 84
- Feature Information for Call Transfer and Forwarding, on page 85

Information About Call Transfer and Forward

Call Forward

Call forward feature diverts calls to a specified number under one or more of the following conditions:

- All calls—When all-call call forwarding is activated by a phone user, all incoming calls are diverted. The target destination for diverted calls can be specified in the router configuration or by the phone user with a soft key or feature access code. The most recently entered destination is recognized by Cisco Unified CME, regardless of how it was entered.
- No answer—Incoming calls are diverted when the extension does not answer before the timeout expires. The target destination for diverted calls is specified in the router configuration.
- Busy—Incoming calls are diverted when the extension is busy and call waiting is not active. The target destination for diverted calls is specified in the router configuration.
- Night service—All incoming calls are automatically diverted during night-service hours. The target destination for diverted calls is specified in the router configuration.

A directory number can have all four types of call forwarding defined at the same time with a different forwarding destination defined for each type of call forwarding. If more than one type of call forwarding is active at one time, the order for evaluating the different types is as follows:

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

H.450.3 capabilities are enabled globally on the router by default, and can be disabled either globally or for individual dial peers. You can configure incoming patterns for using the H.450.3 standard. Calling-party numbers that do not match the patterns defined with this command are forwarded using Cisco-proprietary call forwarding for backward compatibility. For information about configuring H.450.3 on a Cisco Unified CME system, see Enable Call Forwarding for a Directory Number, on page 35.

Selective Call Forward

You can apply call forward to a busy or no-answer directory number based on the number that is dialed to reach the directory number: the primary number, the secondary number, or either of those numbers expanded by a dial-plan pattern.

Cisco Unified CME automatically creates one POTS dial peer for each ephone-dn when it is assigned a primary number. If the ephone-dn is assigned a secondary number, it creates a second POTS dial peer. If the **dialplan-pattern** command is used to expand the primary and secondary numbers for ephone-dns, it creates two more dial peers, resulting in the creation of the following four dial peers for the ephone-dn:

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

Call forwarding is normally applied to all dial peers created for an ephone-dn. Selective call forwarding allows you to apply call forwarding for busy or no-answer calls only for the dial peers you have specified, based on the called number that was used to route the call to the ephone-dn.

```
For example, the following commands set up a single ephone-dn (ephone-dn 5) with four dial
 peers:

telephony-service

 dialplan-pattern 1 40855501.. extension-length 4 extension-pattern 50..


ephone-dn 5

 number 5066 secondary 5067
```

In this example, selective call forwarding can be applied so that calls are forwarded when:

- callers dial the primary number 5066.
- when callers dial the secondary number 5067.
- when callers dial the expanded numbers 4085550166 or 4085550167.

For configuration information, see Enable Call Forwarding for a Directory Number, on page 35.

Call Forward Unregistered

The Call Forward Unregistered (CFU) feature allows you to forward a call to a different number if the directory number (DN) is not associated with a phone or if the associated phone is not registered to Cisco Unified CME. The CFU feature is very useful for wireless phone users when the wireless phone is out of the access point or phone shuts down automatically because of an automatic shutdown feature. The service is not available and the call can be forwarded to the CFU destination. Any unregistered or floating DN can be forwarded using the CFU feature.

An unregistered DN indicates that none of its associated phones are registered to the Cisco Unified CME. A registered phone will become unregistered when the Cisco Unified CME sends an unregistration request or

responses to a phone's unregistration request. Cisco Unified CME sends an unregistration request under the following circumstances:

- When the keepalive timer expires.
- When a user issues a reset or restart command on the phone.
- When an extension mobility (EM) user logs into the phone. (All DNs configured under the logout-profile are unregistered except for the shared ones that are associated with other registered phones.)
- When an EM user logs out of the phone. (All DNs configured under the user-profile are unregistered except for the shared ones that are associated with other registered phones.)

There is always a gap between the time the phone loses its connection with Cisco Unified CME and the time when Cisco Unified CME claims the phone is unregistered. The length of the gap depends on the keepalive timer. Cisco Unified CME considers the phone as registered and tries to associate DNs until the keepalive timer expires. You can configure the expiration for the keepalive timer using the registrar server expires max <seconds> min <seconds> command under sip in voice service voip mode for SIP IP phones. For more information, see Example for Configuring Keepalive Timer Expiration in SIP Phones, on page 84.

Cisco Unified CME 8.6 supports the CFU feature on SIP IP phones using the call-forward b2bua unregistered command under voice register dn tag. The CFU feature supports overlap dialing and en-bloc dialing. A call to a floating DN is forwarded to its CFU destination, if configured. Calls to a DN out of service point or phones losing connection are not forwarded to a CFU number until the phone becomes unregistered. For more information on configuring call-forward unregistered, see Example for Configuring Call Forward Unregistered for SIP IP Phones, on page 83.



Note In earlier versions of Cisco Unified CME, a busy tone was played for callers when the callers are unable to reach the SCCP phone number. In Cisco Unified CME 8.6 and later versions, a fast busy tone is played instead of a busy tone for callers who are unable to reach the phone.

B2BUA Call Forward for SIP Devices

Cisco Unified CME 3.4 an d later versions acts as both UA server and UA client; that is, as a B2BUA. Calls into a SIP phone can be forwarded to other SIP or SCCP devices (including Cisco Unity or Cisco Unity Express, third-party voice mail systems, an auto attendant or an IVR system, such as Cisco Unified IPCC and Cisco Unified IPCC Express). In addition, SCCP phones can be forwarded to SIP phones.

Cisco Unity or other voice-messaging systems connected by a SIP trunk or SIP user agent are able to pass an MWI to a SIP phone when a call is forwarded. The SIP phone then displays the MWI when indicated by the voice-messaging system.

The call-forward busy response is triggered when a call is sent to a SIP phone using a VoIP dial peer and a busy response is received back from the phone. SIP-to-SIP call forwarding is invoked only if the phone is dialed directly. Call forwarding is not invoked when the phone number is called through a sequential, longest-idle, or peer hunt group.

You can configure call forwarding for an individual directory number, or for every number on a SIP phone. If the information is configured in both, the information under voice register dn takes precedence over the information configured under voice register pool.

For configuration information, see Configure SIP-to-SIP Phone Call Forwarding, on page 63.

Call Forward All Synchronization for SIP Phones

The Call Forward All feature allows users to forward all incoming calls to a phone number that they specify. This feature is supported on all SIP phones and can be provisioned from either Cisco Unified CME or the individual SIP phone. Before Cisco Unified CME 4.1, there was no method for exchanging the Call Forward All configuration between Cisco Unified CME and the SIP phone. If Call Forward All was enabled on the phone, the configuration in Cisco Unified CME was not updated; conversely, the configuration in Cisco Unified CME was not sent to the phone.

In Cisco Unified CME 4.1 and later, the following enhancements are supported for the Cisco Unified IP Phone 7911G, 7941G, 7941GE, 7961G, 7961GE, 7970G, and 7971GE to keep the configuration consistent between Cisco Unified CME and the SIP phone:

- When Call Forward All is configured on Cisco Unified CME with the **call-forward b2bua all** command, the configuration is sent to the phone which updates the CfwdAll soft key to indicate that Call forward All is enabled. Because Call Forward All is configured on a per line basis, the CfwdAll soft key is updated only when Call Forward All is enabled for the primary line.
- When a user enables Call Forward All on a phone using the CfwdAll soft key, the uniform resource identifier (URI) for the service (defined with the **call-feature-uri** command) and the call forward number (unless Call Forward All is disabled) is sent to Cisco Unified CME. It updates its voice register pool and voice register dn configuration with the **call-forward b2bua all** command to be consistent with the phone configuration.
- Call Forward All supports KPML so that a user does not need to press the Dial or # key, or wait for the interdigit timeout, to configure the Call Forward All number. Cisco Unified CME collects the Call Forward All digits until it finds a match in the dial peers.

For configuration information, see Configure Call-Forwarding-All Softkey URI on SIP Phones, on page 68.

Call Transfer

When you are connected to another party, call transfer allows you to shift the connection of the other party to a different number. Call transfer methods must inter-operate with systems in the other networks with which you interface. Cisco CME 3.2 and later versions provide full call-transfer and call-forwarding interoperability with call processing systems that support H.450.2, H.450.3, and H.450.12 standards. For call processing systems that do not support H.450 standards, Cisco CME 3.2 and later versions provide VoIP-to-VoIP hairpin call routing.

Call transfers can be blind or consultative. A blind transfer is one in which the transferring extension connects the caller to a destination extension before ringback begins. A consultative transfer is one in which the transferring party either connects the caller to a ringing phone (ringback heard) or speaks with the third party before connecting the caller to the third party.

You can configure blind or consultative transfer on a system-wide basis or for individual extensions. For example, in a system that is set up for consultative transfer, a specific extension with an auto-attendant that automatically transfers incoming calls to specific extension numbers can be set to use blind transfer, because auto-attendants do not use consultative transfer.

Call Transfer Blocking

Transfers to all numbers except those on local phones are automatically blocked by default. During configuration, you can allow transfers to nonlocal numbers. In Cisco Unified CME 4.0 and later versions, you can prevent individual phones from transferring calls to numbers that are globally enabled for transfer. This ensures that individual phones do not incur toll charges by transferring calls outside the Cisco Unified CME

system. Call transfer blocking can be configured for individual phones or configured as part of a template that is applied to a set of phones.

Another way to eliminate toll charges on call transfers is to limit the number of digits that phone users can dial when transferring calls. For example, if you specify a maximum of eight digits in the configuration, users who are transferring calls can dial one digit for external access and seven digits more, which is generally enough for a local number but not a long-distance number. In most locations, this plan will limit transfers to nontoll destinations. Long-distance calls, which typically require ten digits or more, will not be allowed. This configuration is only necessary when global transfer to numbers outside the Cisco Unified CME system has been enabled using the **transfer-pattern** (telephony-service) command. Transfers to numbers outside the Cisco Unified CME system are not permitted by default.

For configuration information, see Configure Call Transfer Options for SCCP Phones, on page 40.

Trunk-to-Trunk Transfer Blocking for Toll Fraud Prevention on Cisco Unified SIP IP Phones

In Cisco Unified CME 4.0 trunk-to-trunk transfer blocking for toll bypass fraud prevention is supported on Cisco Unified Skinny Client Control Protocol (SCCP) IP phones.

In Cisco Unified CME 9.5, trunk-to-trunk transfer blocking for toll bypass fraud prevention is also supported on Cisco Unified Session Initiation Protocol (SIP) IP phones.

In Cisco Unified CME 10.5, trunk-to-trunk conference blocking is also supported on Cisco Unified Skinny Client Control Protocol (SCCP) and Cisco Unified Session Initiation Protocol (SIP) IP phones.

Table 1: Configuration Modes for Transfer-Blocking Commands, on page 5 lists the transfer-blocking commands and the appropriate configuration modes for Cisco Unified CME and Cisco Unified SRST.

Table 1: Configuration Modes for Transfer-Blocking Commands

Commands	Cisco Unified CME
transfer-pattern	telephony-service
transfer max-length	voice register pool or voice register template
transfer-pattern blocked	voice register pool or voice register template
conference transfer-pattern	telephony-service
conference max-length	ephone ephone-template voice register pool voice register template

Commands	Cisco Unified CME
conference-pattern blocked	ephone ephone-template voice register pool voice register template

Note The call transfer and conference restrictions apply when transfers or conferences are initiated toward external parties, like a PSTN trunk, a SIP trunk, or an H.323 trunk. The restrictions do not apply to transfers to local extensions.

Transfer Pattern

The **transfer-pattern** command for Cisco Unified SCCP IP phones is extended to Cisco Unified SIP IP phones.

The **transfer-pattern** command specifies the directory numbers for call transfer. The command can be configured up to 32 times using the following command syntax:

transfer-pattern *transfer-pattern* [blind]

Note The **blind** keyword in the **transfer-pattern** command applies to Cisco Unified SCCP IP phones only and does not apply to Cisco Unified SIP IP phones.

With the **transfer-pattern** command configured, only call transfers to numbers that match the configured transfer pattern are allowed to take place. With the transfer pattern configured, all or a subset of transfer numbers can be dialed and the transfer to a remote party can be initiated.

Note In Cisco Unified CME 9.5 and later versions, Cisco Unified SIP IP phones and Cisco Unified SCCP IP phones registered to the same Cisco Unified CME are considered local and do not require transfer-pattern configuration.

The following are examples of configurable transfer patterns:

- .T—This configuration allows call transfers to any destinations with one or more digits, like 123, 877656, or 76548765.
- 919.....—This configuration only allows call transfers to remote numbers beginning with “919” and followed by eight digits, like 91912345678. However, call transfers to 9191234 or 919123456789 are not allowed.

Backward Compatibility

To maintain backward compatibility, all call transfers from Cisco Unified SIP IP phones to any number (local or over trunk) are allowed when no transfer patterns are configured through the **transfer-pattern**, **transfer-pattern blocked**, or **transfer max-length** commands.

For Cisco Unified SCCP IP phones, call transfers over trunk continue to be blocked when no transfer patterns are configured.

Dial Plans

Whatever dial plan is used for external calls, the same numbers should be configured as specific numbers using the **transfer-pattern** command.

If a dial plan requires "9" to be dialed before an external call is made, then "9" should be a prefix of the transfer-pattern number. For example, 12345678 is an external number that requires "9" to be dialed before the external call can be made so the transfer-pattern number should be 912345678.



Note In Cisco Unified CME 9.5 and later versions, once transfer patterns are configured in telephony-service configuration mode, the transfer patterns apply to both Cisco Unified SCCP IP phones and Cisco Unified SIP IP phones.

Transfer Max-Length

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

For example, the maximum length is configured as 5, then only call transfers from Cisco Unified SIP IP phones up to a five-digit directory number are allowed. All call transfers to directory numbers with more than five digits are blocked.



Note If only transfer max length is configured and conference max-length is not configured, then transfer max-length takes effect for transfers and conferences.

Conference Max-Length

Conference calls are allowed when:

- both **conference transfer-pattern** and **transfer-pattern** commands are configured
- dialed digits match the configured transfer pattern

When conference max-length command is configured, the Cisco Unified CME will allow the conferences only if the dialed digits are within the max-length limit.

If configured, the conference max-length command does not impact call transfers.



Note If both **conference max-length** and **transfer max-length** commands are configured, the conference **max-length** command takes precedence for conferences.

Conference-Pattern Blocked

The conference-pattern blocked command is used to prevent extensions on an ephone or a voice register pool from initiating conferences.

The following table summarizes the behavior of the **conference-pattern blocked** command in relation to **no conference-pattern blocked**, **conference max-length**, **no conference max-length**, and **transfer max-length** commands.

	conference max-length	**no conference max-length**
No conference-pattern blocked (default case)	Allowing/Blocking of conference call depends on configured conference max-length	Allowing/Blocking of conference call depends on configured transfer max-length
conference-pattern blocked	No conference calls allowed for SIP and SCCP phones.	

	Max-length <= allowed max-length		**Max-length > allowed max-length**	
	Transfer	Conference	Transfer	Conference
Transfer max-length + No Conference max-length (use transfer max-length for conference cases too, as conference max-length not configured)	Y	Y	N	N
No transfer max-length + Conference max-length (conference max-length has precedence over transfer max-length for conference)	Y	Y	Y	N
No transfer max-length + Conference max-length (conference max-length has precedence over transfer max-length for conference)	Y	Y	N	N

No transfer max-length + No conference max-length	All transfer and conference calls are allowed.

Configure the Maximum Number of Digits for a Conference Call

Before you begin

- Cisco Unified CME 10.5 or a later version.
- The conference transfer-pattern command must be configured.
- The transfer-pattern command must be configured.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. Enter one of the following commands:
   - **voice register pool** *pool-tag*
   - **voice register template** *template-tag*
   - **ephone** *phone-tag*
   - **ephone template** *template-tag*
4. **conference max-length** value
5. **exit**

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	Enter one of the following commands: • **voice register pool** *pool-tag* • **voice register template** *template-tag* • **ephone** *phone-tag* • **ephone template** *template-tag*	Enters voice register pool configuration mode and creates a pool configuration for a Cisco Unified SIP IP phone in Cisco Unified CME. • *pool-tag*—Unique number assigned to the pool. Range is 1 to 100.

	Command or Action	Purpose
	Example: `Router(config)# voice register pool 25`	or Enters voice register template configuration mode and defines a template of common parameters for Cisco Unified SIP IP phones. • *template-tag*—Declares a template tag. Range is 1 to 10. or Enters ephone configuration mode. • *phone-tag*—Unique sequence number that identifies this ephone during configuration tasks. The maximum number of ephones is version and platform-specific. Type? to display range.
Step 4	**conference max-length** value **Example:** `Router(config-register-pool)# conference max-length 6`	Allows the conference calls from Cisco IP phones to specified directory numbers of phones. • conference max-length—Specifies the maximum number of digits while making a conference call. Range is 3 to 16.
Step 5	**exit** **Example:** `Router(config-register-pool)# exit`	Exits voice register pool configuration mode and enters global configuration mode.

Configure Conference Blocking Options for Phones

To prevent extensions from making conference calls to directory numbers that are otherwise allowed globally.

Before you begin

- Cisco Unified CME 10.5 or a later version.
- The conference transfer-pattern command must be configured.
- The transfer-pattern command must be configured.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. Enter one of the following commands:
   - **voice register pool** *pool-tag* or
   - **voice register template** *template-tag*
   - **ephone** *phone-tag*
   - **ephone template** *template-tag*

4. **conference-pattern blocked**
5. **exit**

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	Enter one of the following commands: • **voice register pool** *pool-tag* or • **voice register template** *template-tag* • **ephone** *phone-tag* • **ephone template** *template-tag* **Example:** `Router(config)# voice register pool 25`	Enters voice register pool configuration mode and creates a pool configuration for a Cisco Unified SIP IP phone in Cisco Unified CME or for a set of Cisco Unified SIP IP phones in Cisco Unified SIP SRST. • pool-tag—Unique number assigned to the pool. Range is 1 to 100. or Enters voice register template configuration mode and defines a template of common parameters for Cisco Unified SIP IP phones. • template-tag—Declares a template tag. Range is 1 to 10. or Enters ephone configuration mode. • phone-tag—Unique sequence number that identifies this ephone during configuration tasks. The maximum number of ephones is version and platform-specific. Type? to display range.
Step 4	**conference-pattern blocked** **Example:** `Router(config-register-pool)#`	Blocks conference calls to external numbers. • conference-pattern block—Prevents extensions on an ephone or a voice register pool from initiating conferences.
Step 5	**exit** **Example:** `Router(config-register-pool)# exit`	Exits voice register pool configuration mode.

Transfer-Pattern Blocked

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

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

Table 2: Behaviors of Cisco Unified IP Phones for Specific Configurations, on page 12 compares the behaviors of Cisco Unified SCCP and SIP IP phones for specific configurations.

Table 2: Behaviors of Cisco Unified IP Phones for Specific Configurations

Configuration	Cisco Unified SCCP IP Phones	Cisco Unified SIP IP Phones
No transfer patterns are configured.	All non-local call transfers are blocked.	All non-local call transfers are allowed for backward compatibility.
Specific transfer patterns are configured.	Call transfers to specific external entities are allowed.	Call transfers to specific external entities are allowed.
The **transfer-pattern** blocked command is configured.	All non-local call transfers are blocked. **Note** The configuration reverts to the default, where no transfer patterns are configured.	All non-local call transfers are blocked. **Note** The configuration unconditionally blocks all non-local call transfers. It does not return to the default, where all non-local call transfers are allowed.

Conference Transfer-Pattern

When both the **transfer-pattern** and **conference transfer-pattern** commands are configured and the dialed digits match the configured transfer pattern, conference calls are allowed. However, when the dialed digits do not match any of the configured transfer pattern, the conference call is blocked.

For configuration information, see Specify Transfer Patterns for Trunk-to-Trunk Calls and Conferences for SIP, on page 42 and Conference-Pattern Blocked, on page 8 and Conference Max-Length, on page 7.

For configuration examples, see Example for Configuring Conference Transfer Patterns, on page 73, Example for Configuring Maximum Length of Transfer Number, on page 73, Example for Configuring Transfer Patterns, on page 73, and Example for Blocking All Call Transfers, on page 74.

Call Transfer Recall on SCCP Phones

The Call Transfer Recall feature in Cisco Unified CME 4.3 and later versions returns a transferred call to the phone that initiated the transfer if the destination is busy or does not answer. After a phone user completes a transfer to a directory number on a local phone, if the transfer-to party does not answer before the configured recall timer expires, the call is directed back to the transferor phone. The message “Transfer Recall From *xxxx*” displays on the transferor phone.

The transfer-to directory number cannot have Call Forward Busy enabled, or it cannot be a hunt group pilot number. If the transfer-to directory number has Call Forward No Answer (CFNA) enabled, Cisco Unified CME recalls the call only if the transfer-recall timeout is set to less than the CFNA timeout. If the transfer-recall

timeout is set to more than the CFNA timeout, the call is forwarded to the CFNA target number after the transfer-to party does not answer.

If the transferor phone is busy, Cisco Unified CME attempts the recall again after the transfer-recall timeout value expires. Cisco Unified CME attempts a recall up to three times. If the transferor phone remains busy, the call is disconnected after the third recall attempt.

The transferor phone and transfer-to phone must be registered to the same Cisco Unified CME, however the transferee phone can be remote.

For configuration information, see Enable Call Transfer and Forwarding on SCCP Phones at System-Level, on page 28.

Call Transfer Recall on SIP Phones

From Unified CME 11.6 onwards, Call Transfer Recall feature is supported on SIP phones. This feature returns a transferred call to the phone that initiated the transfer if the destination is busy or does not answer. After a phone user completes a transfer to a directory number on a local SIP phone, and if the transfer-to party does not answer before the configured recall timer expires, the call is directed back to the transferor phone. The message "Transfer Recall From *xxxx* " displays on the transferor phone.

The Call Transfer Recall in SIP phones is achieved using the CLI **timeouts transfer-recall** command in voice register dn or voice register global configuration modes.

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

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

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

When Single Number Reach (SNR) is configured in Cisco Unified CME, the desk IP Phone rings first. If the desk IP Phone does not answer within the configured SNR timer expiry value, the configured remote number (mobile) starts ringing while continuing to ring the desk IP Phone. If both the extensions does not answer the call, transfer recall is directed back to the transferor phone. Transfer recall does not happen if the desk IP Phone or remote phone (mobile) is busy. Also, transfer recall does not happen if one of the SNR extensions answers the call.

For configuration information, see Enable Call-Transfer Recall on SIP Phones at System-Level, on page 34.

From Cisco Unified CME release 11.6 onwards, call transfer recall feature supports mixed deployment of SCCP and SIP phones. In a mixed deployment scenario, you can have a SIP phone as transferor and with an SCCP phone being transfer-to or vice versa.

In mixed mode, if the transfer recall is performed with multiple SIP or SCCP transferors and a single transfer-to SCCP phone, transfer recall display messages are displayed on both the transferors. Here, transfer recall

happens for all the calls when the destination is busy or does not answer the call. In the case of single transfer-to SIP phones, only the first phone call is recalled even if dual-line is configured.

Consultative-Transfer Enhancements in Cisco Unified CME 4.3 and Later Versions

Cisco Unified CME 4.3 modifies the digit-collection process for consultative call transfers. After a phone user presses the Transfer soft key to make a consultative transfer, a new consultative call leg is created and the Transfer soft key is not displayed again until the dialed digits of the transfer-to number are matched to a transfer pattern and the consultative call leg is in the alerting state.

Transfer-to digits dialed by the phone user are no longer buffered. The dialed digits, except the call park FAC code, are collected on the seized consultative call-leg until the digits match a pattern for consultative transfer, blind transfer, park-slot transfer, park-slot transfer blocking, or PSTN transfer blocking. The existing pattern matching process is unchanged, and you have the option of using this new transfer digit-collection method or reverting to the former method.

Before Cisco Unified CME 4.3, the consultative transfer feature collects dialed digits on the original call leg until the digits either match a transfer pattern or blocking pattern. When the transfer-to number is matched, and PSTN blocking is not enabled, the original call is put on hold and an idle line or channel is seized to send the dialed digits from the buffer.

The method of matching a pattern for consultative transfer, blind transfer, park-slot transfer, park-slot transfer blocking, PSTN transfer blocking, and after-hours blocking remain the same. When the transfer-to number matches the pattern for a blind transfer or park-slot transfer, Cisco Unified CME terminates the consultative call leg and transfers the call.

After the transfer-to digits are collected, if the transfer is not committed before the transfer-timeout expires in 30 seconds, the consultation call leg is disconnected.

These enhancements are supported only if:

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

Cisco Unified CME 4.3 and later versions enable these transfer enhancements by default.

To revert to the digit-collection method used in previous versions of Cisco Unified CME, see Enable Call Transfer and Forwarding on SCCP Phones at System-Level, on page 28.

Consultative Transfer With Direct Station Select

Direct Station Select (DSS) is a feature that allows a multi-button phone user to transfer calls to an idle monitored line by pressing the Transfer key and the appropriate monitored line button. A monitored line is one that appears on two phones; one phone can use the line to make and receive calls and the other phone simply monitors whether the line is in use. For Cisco CME 3.2 and later versions, consultative transfers can occur during Direct Station Select (transferring calls to idle monitored lines).

If the person sharing the monitored line does not want to accept the call, the person announcing the call can reconnect to the incoming call by pressing the EndCall soft key to terminate the announcement call and pressing the Resume soft key to reconnect to the original caller.

Direct Station Select consultative transfer is enabled with the **transfer-system full-consult dss** command, which defines the call transfer method for all lines served by the router. The **transfer-system full-consult dss** command supports the **keep-conference** command. See Configure Hardware Conferencing.

H.450.2 and H.450.3 Support

H.450.2 is a standard protocol for exchanging call-transfer information across a network, and H.450.3 is a standard protocol for exchanging call-forwarding information across a network. Cisco CME 3.0 and later versions support the H.450.2 call-transfer standards and the H.450.3 call-forwarding standards that were introduced in Cisco ITS V2.1. Using the H.450.2 and H.450.3 standards to manage call transfer and forwarding in a VoIP network provides the following benefits:

- The final call path from the transferred party to the transfer destination is optimal, with no hairpinned routes or excessive use of resources.
- Call parameters (for example, codec) can be different for the different call legs.
- This solution is scalable.
- There is no limit to the number of times a call can be transferred.

Considerations for using the H.450.2 and H.450.3 standards include the following:

- Cisco IOS Release 12.2(15)T or a later release is required on all voice gateways in the network.
- Support of H.450.2 and H.450.3 is required on all voice gateways in the network. H.450.2 and H.450.3 are used regardless of whether the transfer-to or forward-to target is on the same Cisco Unified CME system as the transferring party or the forwarding party, so the transferred party must also support H.450.2 and the forwarded party must also support H.450.3. The exception is calls that can be reoriginated through hairpin call routing or through the use of an H.450 tandem gateway.
- Call forwarding over SIP networks uses the *302 Moved Temporarily* SIP response, which works in a manner similar to the way in which the H.450.3 standard is used for H.323 networks. To enable call forwarding, you must specify a pattern that matches the calling-party numbers of the calls that you want to be able to forward.
- Cisco Unified CME supports all SIP Refer method call transfer scenarios, but you must ensure that call transfer is enabled using H.450.2 standards.
- H.450 standards are not supported by Cisco Unified Communications Manager, Cisco BTS, or Cisco PGW, although hairpin call routing or an H.450 tandem gateway can be set up to handle calls to and from those types of systems.

The following series of figures depicts a call being transferred using H.450.2 standards. Figure 1: Call Transfer Using H.450.2: A Calls B, on page 16 shows A calling B. Figure 2: Call Transfer Using H.450.2: B Consults with C, on page 16 shows B consulting with C and putting A on hold. Figure 3: Call Transfer Using H.450.2: B Transfers A to C, on page 16 shows that B has connected A and C, and Figure 4: Call Transfer Using H.450.2: A and C Are Connected, on page 16 shows A and C directly connected, with B no longer involved in the call.

Figure 1: Call Transfer Using H.450.2: A Calls B

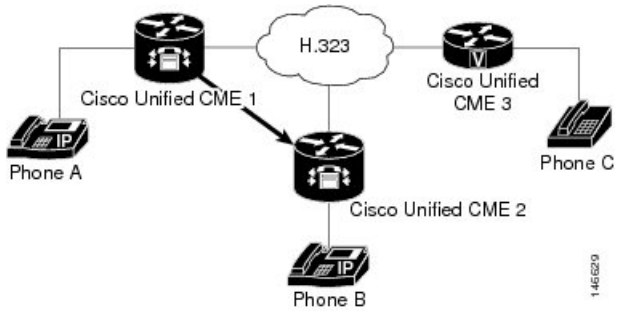


Figure 2: Call Transfer Using H.450.2: B Consults with C

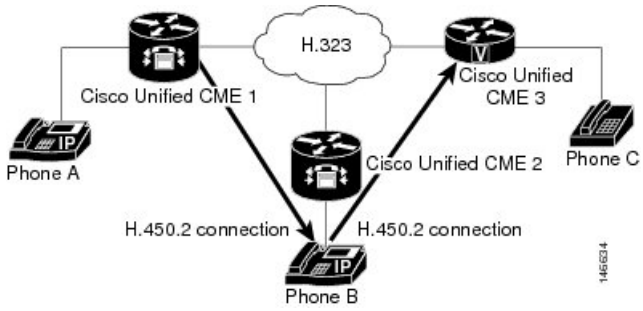


Figure 3: Call Transfer Using H.450.2: B Transfers A to C

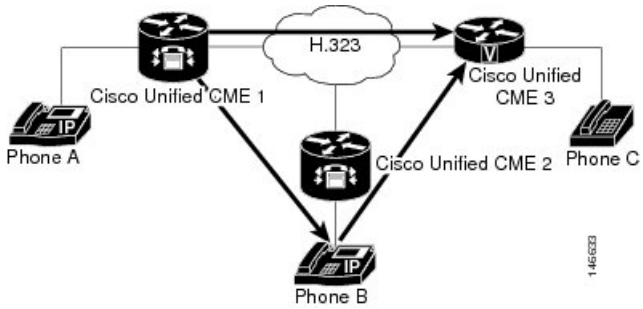
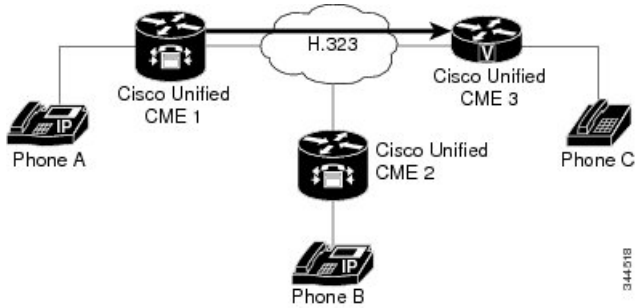


Figure 4: Call Transfer Using H.450.2: A and C Are Connected



Tips for Using H.450 Standards

Use H.450 standards when a network meets the following conditions:

- The router that you are configuring uses Cisco CME 3.0 or a later version, or Cisco ITS V2.1.
- For Cisco CME 3.0 or Cisco ITS V2.1 systems, all endpoints in the network must support H.450.2 and H.450.3 standards. For Cisco CME 3.1 or later systems, if some of the endpoints do not support H.450 standards (for example, Cisco Unified Communications Manager, Cisco BTS, or Cisco PGW), you can use hairpin call routing or an H.450 tandem gateway to handle transfers and forwards with those endpoints. Also, either you must explicitly disable H.450.2 and H.450.3 on the dial peers that handle those calls or you must enable H.450.12 capability to automatically detect the calls that support H.450.2 and H.450.3 and those calls that do not.

Support for the H.450.2 standard and the H.450.3 standard is enabled by default and can be disabled globally or for individual dial peers. For configuration information, see Enable Call Transfer and Forwarding on SCCP Phones at System-Level, on page 28.

Transfer Method Recommendations by Cisco Unified CME Version

You must specify the method to use for call transfers: H.450.2 standard signaling or Cisco proprietary signaling, and whether transfers should be blind or allow consultation. Table 3: Transfer Method Recommendations, on page 17 summarizes transfer method recommendations for all Cisco Unified CME versions.

Table 3: Transfer Method Recommendations

Cisco Unified CME Version	transfer-system Command Default	transfer-system Keyword to Use	Transfer Method Recommendation
4.0 and later	**full-consult**	**full-consult** or **full-blind**	Use H.450.2 for call transfer, which is the default for this version. You do not need to use the **transfer-system** command unless you want to use the **full-blind** or **dss** keyword. Optionally, you can use the proprietary Cisco method by using the **transfer-system** command with the **blind** or **local-consult** keyword. Use H.450.7 for call transfer using QSIG supplementary services
3.0 to 3.3	**blind**	**full-consult** or **full-blind**	Use H.450.2 for call transfer. You must explicitly configure the **transfer-system** command with the **full-consult** or **full-blind** keyword because H.450.2 is not the default for this version. Optionally, you can use the proprietary Cisco method by using the **transfer-system** command with the **blind** or **local-consult** keyword.

Cisco Unified CME Version	transfer-system Command Default	transfer-system Keyword to Use	Transfer Method Recommendation
2.1	**blind**	**blind** or **local-consult**	Use the Cisco proprietary method, which is the default for this version. You do not need to use the **transfer-system** command unless you want to use the **local-consult** keyword. Optionally, you can use the **transfer-system** command with the **full-consult** or **full-blind** keyword. You must also configure the router with a Tcl script that is contained in the app-h450-transfer.x.x.x.x.zip file. This file is available from the Cisco Unified CME software download website at: Download Software.
Earlier than 2.1	**blind**	**blind**	Use the Cisco proprietary method, which is the default for this version. You do not need to use the **transfer-system** command unless you want to use the **local-consult** keyword.

H.450.12 Support

Cisco CME 3.1 and later versions support the H.450.12 call capabilities standard, which provides a means to advertise and dynamically discover H.450.2 and H.450.3 capabilities in voice gateway endpoints on a call-by-call basis. When discovered, the calls associated with non-H.450 endpoints can be directed to use non-H.450 methods for transfer and forwarding, such as hairpin call routing or H.450 tandem gateway.

When H.450.12 is enabled, H.450.2 and H.450.3 services are disabled for call transfers and call forwards unless a positive H.450.12 indication is received from all other VoIP endpoints involved in the call. If a positive H.450.12 indication is received, the router uses the H.450.2 standard for call transfers and the H.450.3 standard for call forwarding. If a positive H.450.12 indication is not received, the router uses the alternative method that you have configured for call transfers and forwards, either hairpin call routing or an H.450 tandem gateway.

You can have either of the following situations in your network:

- All gateway endpoints support H.450.2 and H.450.3 standards. In this situation, no special configuration is required because support for H.450.2 and H.450.3 standards is enabled on the Cisco CME 3.1 or later router by default. H.450.12 capability is disabled by default, but it is not required because all calls can use H.450.2 and H.450.3 standards.
- Not all gateway endpoints support H.450.2 and H.450.3 standards. Therefore, specify how non-H.450 calls are to be handled by choosing one of the following options:
  - Enable the H.450.12 capability in Cisco CME 3.1 and later to dynamically determine, on a call-by-call basis, whether each call has H.450.2 and H.450.3 support. If H.450.12 is enabled and a call is determined to have H.450 support, the call is transferred using H.450.2 standards or forwarded using H.450.3 standards. See Enable H.450.12 Capabilities, on page 46.

    Support for the H.450.12 standard is disabled by default and can be enabled globally or for individual dial peers.

If the call does not have H.450 support, it can be handled by a VoIP-to-VoIP connection that you configure using dial peers and Enable H.323-to-H.323 Connection Capabilities, on page 48. The connection can be used for hairpin call routing or routing to an H.450 tandem gateway.

- Explicitly disable H.450.2 and H.450.3 capability on a global basis or by individual dial peer, which forces all calls to be handled by a VoIP-to-VoIP connection that you configure using dial peers and the Enable H.323-to-H.323 Connection Capabilities, on page 48. This connection can be used for hairpin call routing or routing to an H.450 tandem gateway.

Hairpin Call Routing

Cisco CME 3.1 and later supports hairpin call routing using a VoIP-to-VoIP connection to transfer and forward calls that cannot use H.450 standards. When a call that originally terminated on a voice gateway is transferred or forwarded by a phone or other application attached to the gateway, the gateway reoriginates the call and routes the call as appropriate, making a VoIP-to-VoIP, or hairpin, connection. This approach avoids any protocol dependency on the far-end transferred-party endpoint or transfer-destination endpoint. Hairpin routing of transferred and forwarded calls also causes the generation of separate billing records for each call leg, so that the transferred or forwarded call leg is typically billed to the user who initiates the transfer or forward.

In Cisco CME 3.2 and later versions, transcoding between G.711 and G.729 is supported when one leg of a VoIP-to-VoIP hairpin call uses G.711 and the other leg uses G.729.

Hairpin call routing provides the following benefits:

- Call transfer and forwarding is provided to non-H.450 endpoints, such as Cisco Unified Communications Manager, Cisco BTS, or Cisco PGW.
- The network can also contain Cisco CME 3.0 or Cisco ITS 2.1 systems.

Hairpin call routing has the following disadvantages:

- End-to-end signaling and media delay are increased significantly.
- A single hairpinned call uses as much WAN bandwidth as two directly connected calls.

VoIP-to-VoIP hairpin connections can be made using dial peers if the **allow-connections h323 to h323** command is enabled and at least one of the following is true:

- H.450.12 is used to detect calls on which H.450.2 or H.450.3 is not supported by the remote system.
- H.450.2 or H.450.3 is explicitly disabled.
- Cisco Unified CME automatically detects that the remote system is a Cisco Unified Communications Manager.

Figure 5: Hairpin with H.323: A Calls B, on page 20 shows a call that is made from A to B. Figure 6: Hairpin with H.323: Call is Forwarded to C, on page 20 shows that B has forwarded all calls to C. Figure 7: Hairpin with H.323: A is Connected to C via B, on page 20 shows that A and C are connected by an H.323 hairpin.

Figure 5: Hairpin with H.323: A Calls B

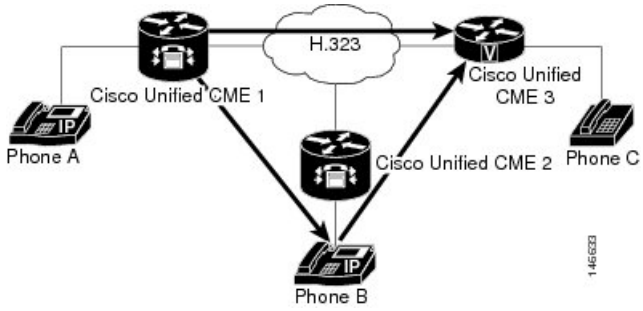


Figure 6: Hairpin with H.323: Call is Forwarded to C

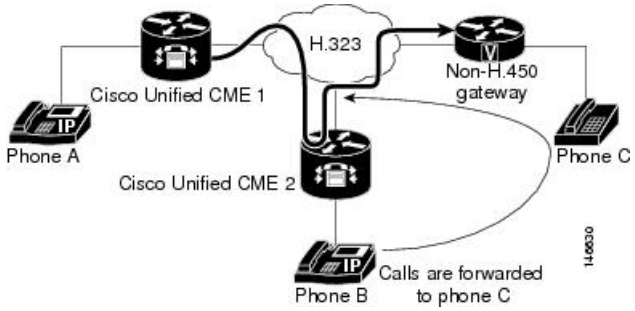
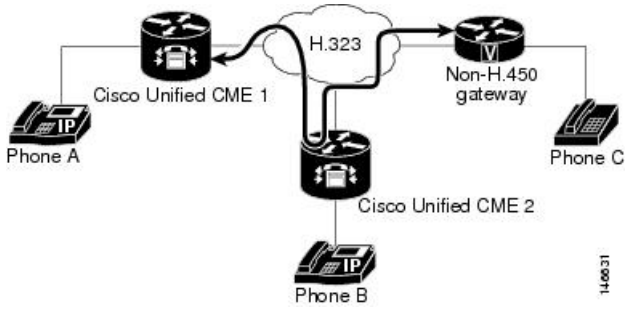


Figure 7: Hairpin with H.323: A is Connected to C via B



Tips for Using Hairpin Call Routing

Use hairpin call routing when a network meets the following three conditions:

- The router that you are configuring uses Cisco CME 3.1 or a later version.
- Some or all calls require VoIP-to-VoIP routing because they cannot use H.450 standards, which can happen for any of the following reasons:
  - H.450 capabilities have been explicitly disabled on the router.
  - H.450 capabilities do not exist in the network.
  - H.450 capabilities are supported on some endpoints and not supported on other endpoints, including those handled by Cisco Unified Communications Manager, Cisco BTS, and Cisco PGW. When some endpoints support H.450 and others do not, you must enable H.450.12 capabilities on the router to detect which endpoints are H.450-capable or designate some dial peers as H.450-capable.

For more information about enabling H.450.12 capabilities, see Enable H.450.12 Capabilities, on page 46.

- No voice gateway is available to act as an H.450 tandem gateway.

For information about configuring Cisco Unified CME to forward calls using local hairpin routing, see Forward Calls Using Local Hairpin Routing, on page 50.

Support for VoIP-to-VoIP connections is disabled by default and can be enabled globally. For configuration information, see Enable H.323-to-H.323 Connection Capabilities, on page 48.

Calling Number Local

In a scenario where calls are forwarded using local hairpin call routing, you can use the Calling Number Local feature. Calling Number Local replaces a calling-party number and name with the forwarding-party number and name (the local number and name). For ephone-dns, the CLI command **calling-number local** is configured under telephony-service configuration to enable the feature. For more information, see Cisco Unified Communications Manager Express Command Reference.

From Cisco Unified CME Release 12.0 onwards, calling number local feature is supported for voice register DNs as well. For voice register DNs, the CLI command **calling-number local** is configured in voice register global configuration mode. For more information, see Cisco Unified Communications Manager Express Command Reference.

When the CLI command **calling-number local** is enabled, the calling number is replaced with the forwarding party's number. If the forwarded number is over a trunk, toll charges may be applied on the forwarding number.

H.450 Tandem Gateways

H.450 tandem gateways address the limitations of hairpin call routing using a manner similar to hairpin call routing but without the double WAN link traversal created by hairpin connections. An H.450 tandem gateway is an additional voice gateway that serves as a “front-end” for a call processor that does not support the H.450 standards, such as Cisco Unified Communications Manager, Cisco BTS Softswitch (Cisco BTS), or Cisco PSTN Gateway (Cisco PGW). Transferred and forwarded calls that are intended for non-H.450 endpoints are terminated instead on the H.450 tandem gateway and reoriginated there for delivery to the non-H.450 endpoints. The H.450 tandem gateway can also serve as a PSTN gateway.

An H.450 tandem gateway is configured with a dial peer that points to the Cisco Unified Communications Manager or other system for which the H.450 tandem gateway is serving as a front end. The H.450 tandem voice gateway is also configured with dial peers that point to all the Cisco Unified CME systems in the private H.450 network. In this way, Cisco Unified CME and the Cisco Unified Communications Manager are not directly linked to each other, but are instead both linked to an H.450 tandem gateway that provides H.450 services to the non-H.450 platform.

An H.450 tandem gateway can also work as a PSTN gateway for remote Cisco Unified CME systems and for Cisco Unified Communications Manager (or other non-H.450 system). Use different inbound dial peers to separate Cisco Unified Communications Manager-to-PSTN G.711 calls from tandem gateway-to-Cisco Unified CME G.729 calls.

Note An H.450 tandem gateway that is used in a network to support non-H.450-capable call processing systems requires the Integrated Voice and Video Services feature license. This feature license, which was introduced in March 2004, includes functionality for H.323 gatekeeper, IP-to-IP Gateway, and H.450 tandem gateway. With Cisco IOS Release 12.3(7)T, an H.323 gatekeeper feature license is required with a JSX Cisco IOS image on the selected router. Consult your Cisco Unified CME SE regarding the required feature license. With Cisco IOS Release 12.3(7)T, you cannot use Cisco Unified CME and H.450 tandem gateway functionality on the same router.

VoIP-to-VoIP connections can be made for an H.450 tandem gateway if the **allow-connections h323 to h323** command is enabled and one or more of the following is true:

- H.450.12 is used to dynamically detect calls on which H.450.2 or H.450.3 is not supported by the remote VoIP system.
- H.450.2 or H.450.3 is explicitly disabled.
- Cisco CME 3.1 or later automatically detects that the remote system is a Cisco Unified Communications Manager.

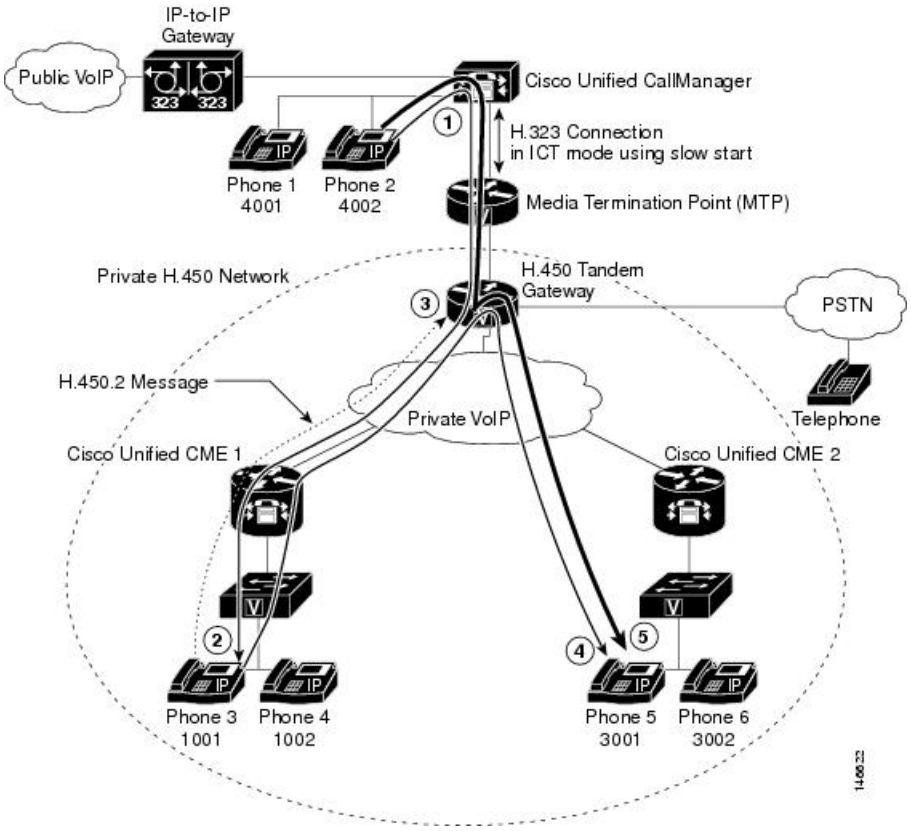
For Cisco CME 3.1 and earlier, the only type of VoIP-to-VoIP connection supported by Cisco Unified CME is H.323-to-H.323. For Cisco CME 3.2 and later versions, H.323-to-SIP connections are allowed only for Cisco Unified CME systems running Cisco Unity Express.

Figure 8: H.450 Tandem Gateway, on page 23 shows a tandem voice gateway that is located between the central hub of the network of a CPE-based Cisco CME 3.1 or later network and a Cisco Unified Communications Manager network. This topology would work equally well with a Cisco BTS or Cisco PGW in place of the Cisco Unified Communications Manager.

In the network topology in Figure 8: H.450 Tandem Gateway, on page 23, the following events occur (refer to the event numbers on the illustration):

1. A call is generated from extension 4002 on phone 2, which is connected to a Cisco Unified Communications Manager. The H.450 tandem gateway receives the H.323 call and, acting as the H.323 endpoint, the H.450 tandem gateway handles the call connection to a Cisco Unified IP phone in a CPE-based Cisco CME 3.1 or later network.
2. The call is received by extension 1001 on phone 3, which is connected to Cisco Unified CME 1. Extension 1001 performs a consultation transfer to extension 2001 on phone 5, which is connected to Cisco Unified CME 2.
3. When extension 1001 transfers the call, the H.450 tandem gateway receives an H.450.2 message from extension 1001.
4. The H.450 tandem gateway terminates the call leg from extension 1001 and reoriginates a call leg to extension 2001, which is connected to Cisco Unified CME 2.
5. Extension 4002 is connected with extension 2001.

Figure 8: H.450 Tandem Gateway



Tips for Using H.450 Tandem Gateways

Use this procedure when a network meets the following conditions:

- The router that you are configuring uses Cisco CME 3.1 or a later version.
- Some endpoints in the network are not H.450-capable, including those handled by Cisco Unified Communications Manager, Cisco BTS, and Cisco PGW.

Support for VoIP-to-VoIP connections is disabled by default and can be enabled globally. For more information, see Enable H.323-to-H.323 Connection Capabilities, on page 48.

Use dial peers to set up an H.450 tandem gateway. See Dial Peers, on page 23.

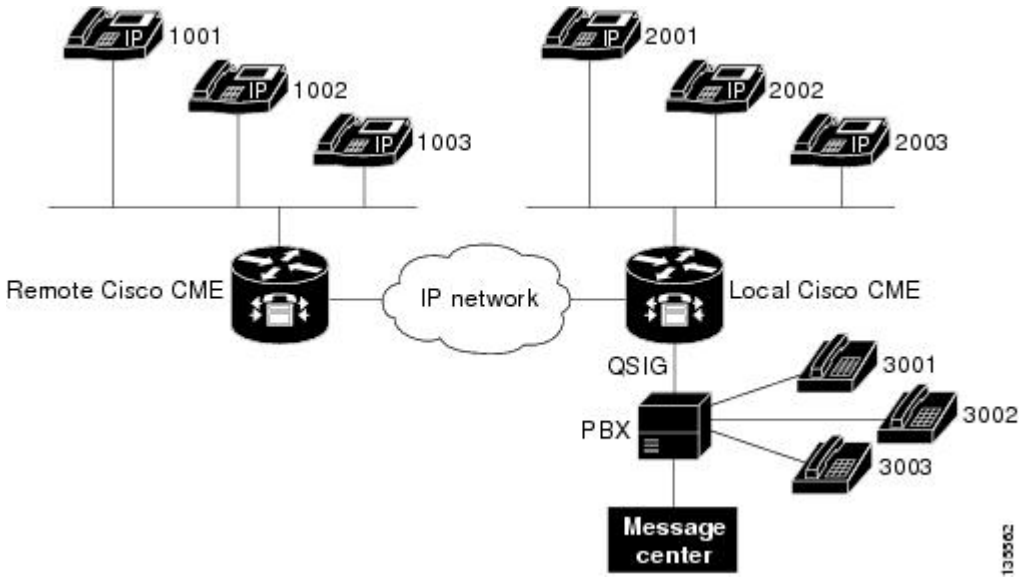
Dial Peers

Dial peers describe the virtual interfaces to or from which a call is established. All voice technologies use dial peers to define the characteristics associated with a call leg. Attributes applied to a call leg include specific quality of service (QoS) features, compression/decompression (codec), voice activity detection (VAD), and fax rate. Dial peers are also used to establish the routing paths in your network, including special routing paths such as hairpins and H.450 tandem gateways. Dial peer settings override the global settings for call forward and call transfer.

Q Signaling Supplementary Services

Q Signaling (QSIG) is an intelligent inter-PBX signaling system widely adopted by PBX vendors. It supports a range of basic services, generic functional procedures, and supplementary services. Cisco Unified CME 4.0 introduces supplementary services features that allow Cisco Unified CME phones to seamlessly interwork using QSIG with phones connected to a PBX. One benefit is that IP phones can use a PBX message center with proper MWI notifications. Figure 9: Cisco Unified CME System with PBX, on page 24 illustrates a topology for a Cisco Unified CME system with some phones under the control of a PBX.

Figure 9: Cisco Unified CME System with PBX



The following QSIG supplementary service features are supported in Cisco Unified CME systems. They follow the standards from the European Computer Manufacturers Association (ECMA) and the International Organization for Standardization (ISO) on PRI and BRI interfaces.

- Basic calls between IP phones and PBX phones.
- Calling Line/Name identification (CLIP/CNIP) presented on an IP phone when called by a PBX phone; in the reverse direction, such information is provided to the called endpoint.
- Connected Line/Name identification (COLP/CONP) information provided when a PBX phone calls an IP phone and is connected; in the reverse direction, such information presented on an IP phone.
- Call Forward using QSIG and H.450.3 to support any combination of IP phone and PBX phone, including an IP phone in the Cisco Unified CME system that is connected to a PBX or an IP phone in another Cisco Unified CME system across an H.323 network.
- Call forward to the PBX message center according to the configured policy. The other two endpoints can be a mixture of IP phone and PBX phones.
- Hairpin call transfer, which interworks with a PBX in transfer-by-join mode. Note that Cisco Unified CME does not support the actual signaling specified for this transfer mode (including the involved FACILITY message service APDUs) which are intended for an informative purpose only and not for the transfer functionality itself. As a transferrer (XOR) host, Cisco Unified CME simply hairpins two call legs to create a connection; as a transferee (XEE) or transfer-to (XTO) host, it will not be aware of a transfer that is taking place on an existing leg. As a result, the final endpoint may not be updated with the accurate identity of its peer. Both blind transfer and consult transfer are supported.

- Message-waiting indicator (MWI) activation or deactivation requests are processed from the PBX message center.
- The PBX message center can be interrogated for the MWI status of a particular ephone-dn.
- A user can retrieve voice messages from a PBX message center by making a normal call to the message center access number.

For information about enabling QSIG supplementary services, see Enable H.450.7 and QSIG Supplementary Services at System-Level, on page 52 and Enable H.450.7 and QSIG Supplementary Services on a Dial Peer, on page 53.

Disable SIP Supplementary Services for Call Forward and Call Transfer

If a destination gateway does not support supplementary services, you can disable REFER messages for call transfers and the redirect responses for call forwarding from being sent by Cisco Unified CME.

For configuration information, see Disable SIP Supplementary Services for Call Forward and Call Transfer, on page 55.

Typical Network Scenarios for Call Transfer and Call Forwarding

In a mixed network that involves two or more types of call agents or call-control systems, there can be communication protocol discrepancies and dependencies, and therefore the opportunity for interoperability errors. These discrepancies show up most often when a call is being transferred or forwarded. This section provides descriptions of the specific mixed-network scenarios you might encounter when configuring a router running Cisco CME 3.1 or a later version. Each of the following sections point to the configuration instructions necessary to ensure call transfer and forwarding capabilities throughout the network.



Note Cisco Communications Manager Express 3.2 (Cisco CME 3.2) and later versions provide full call-transfer and call-forwarding with call processing systems on the network that support H.450.2, H.450.3, and H.450.12 standards. For interoperability with call processing systems that do not support H.450 standards, Cisco CME 3.2 and later versions provide VoIP-to-VoIP hairpin call routing without requiring the special Tool Command Language (Tcl) script that was needed in earlier versions of Cisco Unified CME.

Cisco CME 3.1 or Later and Cisco IOS Gateways

In a network with Cisco CME 3.1 or a later version and Cisco IOS gateways, all systems that might participate in calls that involve call transfer and call forwarding are capable of supporting the H.450.2, H.450.3, and H.450.12 standards. This is the simplest environment for operating the Cisco CME 3.1 or later features.

Configuration for this type of network consists of:

1. Setting up call-transfer and call-forwarding parameters for transfers and forwards that are initiated on this router (H.450.2 and H.450.3 capabilities for transferred parties, transfer destinations, forwarded parties, and forwarding destinations are enabled by default). See Enable Call Transfer and Forwarding on SCCP Phones at System-Level, on page 28.
2. Enabling H.450.12 globally to detect any calls on which H.450.2 and H.450.3 standards are not supported. Although this step is optional, we recommend it. See Enable H.450.12 Capabilities, on page 46.
3. Optionally setting up VoIP-to-VoIP connections (hairpin call routing or H.450 tandem gateway) to route calls that do not support H.450.2 or H.450.3 standards. See Enable H.323-to-H.323 Connection Capabilities, on page 48.
4. Setting up dial peers to manage call legs within the network.

Cisco CME 3.0 or an Earlier Version and Cisco IOS Gateways

Before Cisco CME 3.1, H.450.2 and H.450.3 standards are used for all calls by default and routers do not support the H.450.12 standard.

Configuration for this type of network consists of:

- Setting up call-transfer and call-forwarding parameters for transfers and forwards that are initiated on this router (H.450.2 and H.450.3 capabilities for transferred parties, transfer destinations, forwarded parties, and forwarding destinations are enabled by default). See Enable Call Transfer and Forwarding on SCCP Phones at System-Level, on page 28
- Enabling H.450.12 in advertise-only mode on Cisco CME 3.1 or later systems. As each Cisco CME 3.0 system is upgraded to Cisco CME 3.1 or later, enable H.450.12 in advertise-only mode. Note that no checking for H.450.2 or H.450.3 support is done in advertise-only mode. When all Cisco CME 3.0 systems in the network have been upgraded to Cisco CME 3.1 or later, remove the advertise-only restriction. See Enable H.450.12 Capabilities, on page 46
- Optionally setting up VoIP-to-VoIP connections (hairpin call routing or H.450 tandem gateway) to route calls that cannot use H.450.2 or H.450.3 standards. See Enable H.323-to-H.323 Connection Capabilities, on page 48
- Setting up dial peers to manage call legs within the network.

Cisco CME 3.1 or Later, Non-H.450 Gateways, and Cisco IOS Gateways

In a network with Cisco CME 3.1 or later, non-H.450 gateways, and Cisco IOS gateways, the H.450.2 and H.450.3 services are provided only to calling endpoints that use H.450.12 to explicitly indicate that they are capable of H.450.2 and H.450.3 operations. Because the Cisco BTS and Cisco PGW do not support the H.450.12 standard, calls to and from these systems that involve call transfer or forwarding are handled using H.323-to-H.323 hairpin call routing.

Configuration for this type of network consists of:

1. Setting up call-transfer and call-forwarding parameters for transfers and forwards that are initiated on this router (H.450.2 and H.450.3 capabilities for transferred parties, transfer destinations, forwarded parties, and forwarding destinations are enabled by default). Optionally disable H.450.2 and H.450.3 capabilities on dial peers that point to non-H.450-capable systems such as Cisco Unified Communications Manager, Cisco BTS, or Cisco PGW. See Enable Call Transfer and Forwarding on SCCP Phones at System-Level, on page 28.
2. Enabling H.450.12 to detect any calls on which H.450.2 and H.450.3 standards are not supported, either globally or for specific dial peers. See Enable H.450.12 Capabilities, on page 46.
3. Setting up VoIP-to-VoIP connections (hairpin call routing or H.450 tandem gateway) to route calls that do not support H.450.2 or H.450.3 standards. See Enable H.323-to-H.323 Connection Capabilities, on page 48.
4. Setting up dial peers to manage call legs within the network.



Note If your network contains a Cisco Unified Communications Manager, also see the instructions in the Enable Interworking with Cisco Unified Communications Manager, on page 57.

Cisco Unified CME, Non-H.450 Gateways, and Cisco IOS Gateways

Note Cisco CME 3.0 and Cisco ITS V2.1 systems do not have H.450.12 capabilities.

In a network that contains a mix of Cisco Unified CME versions and at least one non-H.450 gateway, the simplest configuration approach is to globally disable all H.450.2 and H.450.3 services and force H.323-to-H.323 hairpin call routing for all transferred and forwarded calls. In this case, you would enable H.450.12 detection capabilities globally. Alternatively, you could select to enable H.450.12 capability for specific dial peers. In this case, you would not configure H.450.12 capability globally; you would leave it in its default disabled state.

Configuration for this type of network consists of:

1. Setting up call-transfer and call-forwarding parameters for transfers and forwards that are initiated on this router (H.450.2 and H.450.3 capabilities for transferred parties, transfer destinations, forwarded parties, and forwarding destinations are enabled by default). See Enable Call Transfer and Forwarding on SCCP Phones at System-Level, on page 28.
2. Enabling H.450.12 to detect any calls on which H.450.2 and H.450.3 standards are not supported, either globally or on specific dial peers. See Enable H.450.12 Capabilities, on page 46
3. Setting up VoIP-to-VoIP connections (hairpin call routing or H.450 tandem gateway) to route all transferred and forwarded calls. See Enable H.323-to-H.323 Connection Capabilities, on page 48.
4. Setting up dial peers to manage call legs within the network.

Note If your network contains a Cisco Unified Communications Manager, also see the instructions in the Enable Interworking with Cisco Unified Communications Manager, on page 57.

Cisco CME 3.1 or Later, Cisco Unified Communications Manager, and Cisco IOS Gateways

In a network with Cisco CME 3.1 or later, Cisco Unified Communications Manager, and Cisco IOS gateways, Cisco CME 3.1 and later versions support automatic detection of calls to and from Cisco Unified Communications Manager using proprietary signaling elements that are included with the standard H.323 message exchanges. The Cisco CME 3.1 or later system uses these detection results to determine the H.450.2 and H.450.3 capabilities of calls rather than using H.450.12 supplementary services capabilities exchange, which Cisco Unified Communications Manager does not support. If a call is detected to be coming from or going to a Cisco Unified Communications Manager endpoint, the call is treated as a non-H.450 call. All other calls in this type of network are treated as though they support H.450 standards. Therefore, this type of network should contain only Cisco CME 3.1 or later and Cisco Unified Communications Manager call-processing systems.

Configuration for this type of network consists of:

1. Setting up call-transfer and call-forwarding parameters for transfers and forwards that are initiated on this router (H.450.2 and H.450.3 capabilities for transferred parties, transfer destinations, forwarded parties, and forwarding destinations are enabled by default). See Enable Call Transfer and Forwarding on SCCP Phones at System-Level, on page 28
2. Enabling H.450.12 to detect any calls on which H.450.2 and H.450.3 standards are not supported, either globally or on specific dial peers. See Enable H.450.12 Capabilities, on page 46

3. Setting up VoIP-to-VoIP connections (hairpin call routing or H.450 tandem gateway) to route all transferred and forwarded calls that are detected as being to or from Cisco Unified Communications Manager. SeeEnable H.323-to-H.323 Connection Capabilities, on page 48
4. Setting up specific parameters for Cisco Unified Communications Manager. SeeEnable Cisco Unified Communications Manager to Interwork with Cisco Unified CME, on page 61
5. Setting up dial peers to manage call legs within the network.

Cisco CME 3.0 or an Earlier Version, Cisco Unified Communications Manager, and Cisco IOS Gateways

Calls between the Cisco Unified Communications Manager and the older Cisco CME 3.0 or Cisco ITS V2.1 networks need special consideration. Because Cisco CME 3.0 and Cisco ITS V2.1 systems do not support automatic Cisco Unified Communications Manager detection and also do not natively support H.323-to-H.323 call routing, alternative arrangements are required for these systems.

To configure call transfer and forwarding on the Cisco CME 3.0 router, you can select from the following three options:

- Use a Tcl script to handle call transfer and forwarding by invoking Tcl-script-based H.323-to-H.323 hairpin call routing (app-h450-transfer.2.0.0.9.tcl or a later version). Enable this script on all VoIP dial peers and also under telephony-service mode, and set the local-hairpin script parameter to 1.
- Use a loopback-dn mechanism.
- Configure a loopback call path using router physical voice ports.

All three options force use of H.323-to-H.323 hairpin call routing for all calls regardless of whether the call is from a Cisco Unified Communications Manager or other H.323 endpoint (including Cisco CME 3.1 or later).

Configure Call Transfer and Forwarding

Enable Call Transfer and Forwarding on SCCP Phones at System-Level

To enable H.450 call transfers and forwards for transferring or forwarding parties; that is, to allow transfers and forwards to be initiated from a Cisco Unified CME system, perform the following steps.

Note H.450.2 and H.450.3 capabilities are enabled by default for transferred or forwarded parties and transfer-destination or forward-destination parties. Dial peer settings override the global setting.

Restriction

- Call transfers are handled differently depending on the Cisco Unified CME version. See Transfer Method Recommendations by Cisco Unified CME Version, on page 17 for recommendations on selecting a transfer method for your Cisco Unified CME version.
- The **transfer-system local-consult** command is not supported if the transfer-to destination is on the Cisco ATA, Cisco VG224, or a SCCP-controlled FXS port.
- The H.450.2 and H.450.3 standards are not supported by Cisco Unified Communications Manager, Cisco BTS, or Cisco PGW.
- In versions earlier than Cisco Unified CME 4.2, the caller ID displays correctly only after connect; caller ID does not display correctly at Call Transfer or Call Forward.

Call-Transfer Recall

- Requires Cisco Unified CME 4.3 or a later version.
- Transferor and transfer-to party must be on the same Cisco Unified CME router; transferee party can be remote to the Cisco Unified CME router.
- Transfer recall is not supported if the transfer-to party has Call Forward Busy enabled, or if the transfer-to party is a hunt group pilot number.
- If the transfer-to party has Call Forward No Answer enabled, Cisco Unified CME recalls a transferred call only if the transfer-recall timeout is set to less than the timeout value set with the **call-forward noan** command.
- Recall timer for trunk-line directory number has precedence (set on transferor using **trunk** command with **transfer-timeout** keyword) over the transfer-recall timer. Transfer recall is not initiated for hairpin transfers.

Before you begin

Cisco CME 3.0 or a later version, or Cisco ITS V2.1.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **telephony-service**
4. **transfer-system** {**blind** | **full-blind** | **full-consult** [**dss**] | **local-consult** }
5. **transfer-pattern** *transfer-pattern* [**blind**]
6. **call-forward pattern** *pattern*
7. **timeouts transfer-recall** *seconds*
8. **transfer-digit-collect** {**new-call** | **orig-call**}
9. **exit**
10. **voice service voip**
11. **supplementary-service h450.2**
12. **supplementary-service h450.3**
13. **exit**

14. **dial-peer voice** *tag* **voip**
15. **supplementary-service h450.2**
16. **supplementary-service h450.3**
17. **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	**transfer-system** { **blind** | **full-blind** | **full-consult** [**dss**] | **local-consult** } **Example:** `Router(config-telephony)# transfer-system full-consult`	Specifies the call transfer method. • **blind**—Calls are transferred without consultation using the Cisco proprietary method and a single phone line. This is the default in versions earlier than Cisco Unified CME 4.0. • **full-blind**—Calls are transferred without consultation using H.450.2 standard methods. • **full-consult**—Calls are transferred with consultation using H.450.2 standard methods and a second phone line if available. Calls fall back to full-blind if the second line is unavailable. This is the default in Cisco Unified CME 4.0 and later versions. Transfer-system needs to be set at full-consult for the “transfer by directory” to work. Transfer by directory is supported by full-consult or blind transfer. If you want to transfer using directory/placed/missed/received calls, the transfer-system needs to be set at full-consult for this to work appropriately. When changed to full-consult, you can do "blind transfer" by selecting the number from the directory and when the other phone rings, you can press the softkey "Transfer" and the call will be transferred to the number selected and then you can hang up.

	Command or Action	Purpose
		• **dss**—(Optional) Calls are transferred with consultation to idle monitored lines. All other call-transfer behavior is identical to full-consult. • **local-consult**—Calls are transferred with local consultation using a second phone line if available. The calls fall back to blind for nonlocal consultation or nonlocal transfer target. Not supported if transfer-to destination is on the Cisco ATA, Cisco VG224, or a SCCP-controlled FXS port. • Cisco CME 3.0 and later versions—Use only the **full-blind** or **full-consult** keyword. • Before Cisco CME 3.0—Use the **local-consult** or **blind** keyword. (Cisco ITS 2.1 can use the **full-blind** or **full-consult** keyword by also using the Tcl script in the file called app-h450-transfer.x.x.x.x.zip.)
Step 5	**transfer-pattern** *transfer-pattern* [**blind**] **Example:** `Router(config-telephony)# transfer-pattern .T`	Allows transfer of telephone calls by Cisco Unified IP phones to specified phone number patterns. If no transfer pattern is set, the default is that transfers are permitted only to other local IP phones. • *transfer-pattern*—String of digits for permitted call transfers. Wildcards are allowed. A pattern of .T transfers all calling parties using the H.450.2 standard. • **blind**—(Optional) When H.450.2 consultative call transfer is configured, forces transfers that match the pattern specified in this command to be executed as blind transfers. Overrides settings made using the **transfer-system** and **transfer-mode** commands. **Note** For transfers to nonlocal numbers, transfer-pattern digit matching is performed before translation-rule operations. Therefore, you should specify in this command the digits actually entered by phone users before they are translated.
Step 6	**call-forward pattern** *pattern* **Example:** `Router(config-telephony)# call-forward pattern .T`	Specifies the H.450.3 standard for call forwarding. • *pattern*—Digits to match for call forwarding using the H.450.3 standard. If an incoming calling-party number matches the pattern, it can be forwarded using the H.450.3 standard. A pattern of .T forwards all calling parties using the H.450.3 standard. Calling-party numbers that do not match the patterns defined with this command are forwarded using Cisco proprietary call forwarding for backward compatibility. **Note**

	Command or Action	Purpose
		For forwarding to nonlocal numbers, pattern matching is performed before translation-rule operations. Therefore, you should specify in this command the digits actually entered by phone users before they are translated.
Step 7	**timeouts transfer-recall** *seconds* **Example:** `Router(config-telephony)# timeouts transfer-recall 30`	(Optional) Enables Cisco Unified CME to recall a transferred call if the transfer-to party is busy or does not answer. • *seconds*—Duration, in seconds, to wait before recalling a transferred call. Range: 1 to 1800. Default: 0 (disabled). This command is supported in Cisco Unified CME 4.3 and later versions. This command can also be configured in ephone-dn and ephone-dn-template configuration mode.
Step 8	**transfer-digit-collect** {**new-call** | **orig-call**} **Example:** `Router(config-telephony)# transfer-digit-collect orig-call`	(Optional) Selects the digit-collection method used for consultative call transfers. • **new-call**—Digits are collected from the new call leg. Default value in Cisco Unified CME 4.3 and later versions. • **orig-call**—Digits are collected from original call-leg. Default behavior in versions earlier than Cisco Unified CME 4.3. This command is supported in Cisco Unified CME 4.3 and later versions.
Step 9	**exit** **Example:** `Router(config-telephony)# exit`	Exits telephony-service configuration mode.
Step 10	**voice service voip** **Example:** `Router(config)# voice service voip`	(Optional) Enters voice-service configuration mode to establish global call transfer and forwarding parameters.
Step 11	**supplementary-service h450.2** **Example:** `Router(conf-voi-serv)# supplementary-service h450.2`	(Optional) Enables H.450.2 supplementary services capabilities globally. Default is enabled. Use the **no** form of this command to disable H.450.2 capabilities globally. You can also use this command in dial-peer configuration mode to enable H.450.2 services for a single dial peer.
Step 12	**supplementary-service h450.3** **Example:**	(Optional) Enables H.450.3 supplementary services capabilities globally.

	Command or Action	Purpose
	`Router(conf-voi-serv)# supplementary-service h450.3`	Default is enabled. Use the **no** form of this command to disable H.450.3 capabilities globally. You can also use this command in dial-peer configuration mode to enable H.450.3 services for a single dial peer.
Step 13	**exit** **Example:** `Router(conf-voi-serv)# exit`	(Optional) Exits voice-service configuration mode.
Step 14	**dial-peer voice** *tag* **voip** **Example:** `Router(config)# dial-peer voice 1 voip`	(Optional) Enters dial-peer configuration mode.
Step 15	**supplementary-service h450.2** **Example:** `Router(config-dial-peer)# no supplementary-service h450.2`	(Optional) Enables H.450.2 supplementary services capabilities for an individual dial peer. Default is enabled. You can also use this command in voice-service configuration mode to enable H.450.2 services globally. • If this command is enabled globally and enabled on a dial peer, the functionality is enabled for the dial peer. This is the default. • If this command is enabled globally and disabled on a dial peer, the functionality is disabled for the dial peer. • If this command is disabled globally and either enabled or disabled on a dial peer, the functionality is disabled for the dial peer.
Step 16	**supplementary-service h450.3** **Example:** `Router(config-dial-peer)# no supplementary-service h450.3`	(Optional) Enables H.450.3 supplementary services capabilities exchange for an individual dial peer. Default is enabled. You can also use this command in voice-service configuration mode to enable H.450.3 services globally. • If this command is enabled globally and enabled on a dial peer, the functionality is enabled for the dial peer. This is the default configuration. • If this command is enabled globally and disabled on a dial peer, the functionality is disabled for the dial peer. • If this command is disabled globally and either enabled or disabled on a dial peer, the functionality is disabled for the dial peer.

	Command or Action	Purpose
Step 17	**end** **Example:** `Router(config-dial-peer)# end`	Returns to privileged EXEC mode.

Enable Call-Transfer Recall on SIP Phones at System-Level

To enable call-transfer recalls to be initiated from a Cisco Unified CME system, perform the following steps.



Note

- Transferor and transfer-to party must be on the same Cisco Unified CME router; transferee party can be remote to the Cisco Unified CME router.
- Transfer recall is not supported if the transfer-to party has Call Forward Busy enabled, or if the transfer-to party is a hunt group pilot number.

Before you begin

Cisco Unified CME 11.6 or a later version.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register global**
4. **timeouts transfer-recall** *seconds*
5. **exit**
6. **voice service voip**
7. **no supplementary-service sip refer**
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.

	Command or Action	Purpose
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	**timeouts transfer-recall** *seconds*
	Example:
	`Router(config-register-global)# timeouts transfer-recall 30`
	`Router(config-register-dn)# timeouts transfer-recall 30`
		Enables Cisco Unified CME to recall a transferred call if the transfer-to party is busy or does not answer in the voice register global configuration mode. You can also recall a transferred call in the voice register dn configuration mode.
		• *seconds*—Duration, in seconds, to wait before recalling a transferred call. Range: 1 to 1800. Default: 0 (disabled).
		• This command is supported in Cisco Unified CME 11.6 and later versions.
		• This command can also be configured in voice register dn or voice register global configuration modes.

Step 5	**exit** **Example:** `Router(config-register-global)# exit`	Exits voice register global configuration mode.
Step 6	**voice service voip** **Example:** `Router(config)# voice service voip`	(Optional) Enters voice-service configuration mode.
Step 7	**no supplementary-service sip refer** **Example:** `Router(config-voi-serv)# no supplementary-service sip refer`	Prevents the router from forwarding a REFER message to the destination for call-transfer recalls.
Step 8	**end** **Example:** `Router(config-voi-serv)# end`	Returns to privileged EXEC mode.

Enable Call Forwarding for a Directory Number

To define the conditions and target numbers for call forwarding for individual ephone-dns, and set other restrictions for call forwarding, perform the following steps.

Note When defining call forwarding to nonlocal numbers, it is important to note that pattern digit matching is performed before translation-rule operations. Therefore, you should specify in this command the digits actually entered by phone users before they are translated.

Restriction

- Call forwarding is invoked only if that phone is dialed directly. Call forwarding is not invoked when the phone number is called through a sequential, longest-idle, or peer hunt group.
- If call forwarding is configured for hunt group member, call forward is ignored by the hunt group.
- Calls from an internal extension to an extension which is busy, is forwarded to the SNR destination even if no forward local-calls is configured under the Directory Number.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **telephony-service**
4. **call-forward pattern** *pattern*
5. **exit**
6. **ephone-dn** *dn-tag* [**dual-line**]
7. **number** *number* [**secondary** *number*] [**no-reg** [**both** | **primary**]]
8. **call-forward all** *target-number*
9. **call-forward busy** *target-number* [**primary** | **secondary**] [**dialplan-pattern**]
10. **call-forward noan** *target-number* **timeout** *seconds* [**primary** | **secondary**] [**dialplan-pattern**]
11. **call-forward night-service** *target-number*
12. **call-forward max-length** *length*
13. **no forward local-calls**
14. **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)#`	Enters telephony-service configuration mode.
Step 4	**call-forward pattern** *pattern* **Example:** `Router(config-telephony)# call-forward pattern .T`	Specifies the H.450.3 standard for call forwarding. Calling-party numbers that do not match the patterns defined with this command are forwarded using

	Command or Action	Purpose
		Cisco-proprietary call forwarding for backward compatibility. • *pattern*—Digits to match for call forwarding using the H.450.3 standard. If an incoming calling-party number matches the pattern, it is forwarded using the H.450.3 standard. A pattern of .T forwards all calling parties using the H.450.3 standard.
Step 5	**exit** **Example:** `Router(config-telephony)# exit`	Exits telephony-service configuration mode.
Step 6	**ephone-dn** *dn-tag* [**dual-line**] **Example:** `Router(config)# ephone-dn 20`	Enters ephone-dn configuration mode, creates an ephone-dn, and optionally assigns it dual-line status. • **dual-line**—(Optional) Enables an ephone-dn with one voice port and two voice channels, which supports features such as call waiting, call transfer, and conferencing with a single ephone-dn.
Step 7	**number** *number* [**secondary** *number*] [**no-reg** [**both** | **primary**]] **Example:** `Router(config-ephone-dn)# number 2777 secondary 2778`	Configures a valid extension number for this ephone-dn instance.
Step 8	**call-forward all** *target-number* **Example:** `Router(config-ephone-dn)# call-forward all 2411`	Forwards all calls for this extension to the specified number. • *target-number*—Phone number to which calls are forwarded. **Note** After you use this command to specify a target number, the phone user can activate and cancel the call-forward-all state from the phone using the CFwdAll soft key or a feature access code (FAC).
Step 9	**call-forward busy** *target-number* [**primary** | **secondary**] [**dialplan-pattern**] **Example:** `Router(config-ephone-dn)# call-forward busy 2513`	Forwards calls for a busy extension to the specified number.
Step 10	**call-forward noan** *target-number* **timeout** *seconds* [**primary** | **secondary**] [**dialplan-pattern**] **Example:** `Router(config-ephone-dn)# call-forward noan 2513 timeout 45`	Forwards calls for an extension that does not answer.

	Command or Action	Purpose
Step 11	**call-forward night-service** *target-number* **Example:** `Router(config-ephone-dn)# call-forward night-service 2879`	Automatically forwards incoming calls to the specified number when night service is active. • *target-number*—Phone number to which calls are forwarded. **Note** Night service must also be configured. See Configure Call Coverage Features.
Step 12	**call-forward max-length** *length* **Example:** `Router(config-ephone-dn)# call-forward max-length 5`	(Optional) Limits the number of digits that can be entered for a target number when using the CfwdAll soft key on an IP phone. • *length*—Number of digits that can be entered using the CfwdAll soft key on an IP phone.
Step 13	**no forward local-calls** **Example:** `Router(config-ephone-dn)# no forward local-calls`	(Optional) Specifies that local calls (calls from ephone-dns on the same Cisco Unified CME system) will not be forwarded from this extension. • If this extension is busy, an internal caller hears a busy signal. • If this extension does not answer, the internal caller hears ringback.
Step 14	**end** **Example:** `Router(config-ephone-dn)# end`	Returns to privileged EXEC mode.

Call Transfer for a Directory Number

To enable call transfer for a specific directory number, perform the following steps. This procedure overrides the global setting for blind or consultative transfer for individual directory numbers.

Before you begin

Call transfer must be enabled globally. See Enable Call Transfer and Forwarding on SCCP Phones at System-Level, on page 28.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **ephone-dn** *dn-tag* [**dual-line**]
4. **transfer-mode** {**blind** | **consult**}
5. **timeouts transfer-recall** *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	**ephone-dn** *dn-tag* [**dual-line**] **Example:** `Router(config)# ephone-dn 20`	Enters ephone-dn configuration mode, creates an ephone-dn, and optionally assigns it dual-line status. • **dual-line**—(Optional) Enables an ephone-dn with one voice port and two voice channels, which supports features such as call waiting, call transfer, and conferencing with a single ephone-dn.
Step 4	**transfer-mode** {**blind** | **consult**} **Example:** `Router(config-ephone-dn)# transfer-mode blind`	Specifies the type of call transfer for an individual directory number using the H.450.2 standard, allowing you to override the global setting. • Default: system-level value set with the **transfer-system** command.
Step 5	**timeouts transfer-recall** *seconds* **Example:** `Router(config-ephone-dn)# timeouts transfer-recall 30`	(Optional) Enables call-transfer recall and sets the number of seconds that Cisco Unified CME waits before recalling a transferred call if the transfer-to party does not answer or is busy. • *seconds*—Duration, in seconds, to wait before recalling a transferred call. Range: 1 to 1800. Default: 0 (disabled). • This command is supported in Cisco Unified CME 4.3 and later versions. • This command can also be configured in ephone-dn-template and telephony-service configuration mode.
Step 6	**end** **Example:** `Router(config-ephone-dn)# end`	Returns to privileged EXEC mode.

Configure Call Transfer Options for SCCP Phones

To specify a maximum number of digits for transfer destinations or block transfers to external destinations by individual phones, perform the following steps.

Before you begin

- Transfers made to speed-dial numbers are not blocked when the **transfer-pattern blocked** command is used.
- Transfers made using speed-dial are not blocked by the **after-hours block pattern** command.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **ephone-template** *template-tag*
4. **transfer-pattern blocked**
5. **transfer max-length** *digit-length*
6. **exit**
7. **ephone** *phone-tag*
8. **ephone-template** *template-tag*
9. **restart**
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	**ephone-template** *template-tag* **Example:** `Router(config)# ephone-template 1`	Enters ephone-template configuration mode. • *template-tag*—Unique number that identifies this template during configuration tasks. Range: 1 to 20.
Step 4	**transfer-pattern blocked** **Example:** `Router(config-ephone-template)# transfer-pattern blocked`	(Optional) Prevents directory numbers on the phone to which this template is applied from transferring calls to patterns specified in the **transfer-pattern** (telephony-service) command. **Note**

	Command or Action	Purpose
		This command is also available in ephone configuration mode to block external transfers from individual phones without using a template.
Step 5	**transfer max-length** *digit-length* **Example:** `Router(config-ephone-template)# transfer max-length 8`	(Optional) Specifies the maximum number of digits the user can dial when transferring a call. • *digit-length*—Number of digits allowed in a number to which a call is being transferred. Range: 3 to 16. Default: 16.
Step 6	**exit** **Example:** `Router(config-ephone-template)# exit`	Exits ephone-template configuration mode.
Step 7	**ephone** *phone-tag* **Example:** `Router(config)# ephone 25`	Enters ephone configuration mode.
Step 8	**ephone-template** *template-tag* **Example:** `Router(config-ephone)# ephone-template 1`	Applies a template to a phone. • *template-tag*—Template number that you want to apply to this phone.
Step 9	**restart** **Example:** `Router(config-ephone)# restart`	Performs a fast reboot of this phone without contacting the DHCP server for updated information. Repeat Step 6 to Step 9 for each phone on which you want to limit transfer capabilities.
Step 10	**end** **Example:** `Router(config-ephone)# end`	Exits to privileged EXEC mode.

Verify Call Transfer for SCCP Phones

Procedure

Step 1 Use the **show running-config** command to verify your configuration. Transfer method and patterns are listed in the telephony-service portion of the output. You can also use the **show telephony-service** command to display this information.

Example:

```
Router# show running-config
!
telephony-service
fxo hook-flash
load 7910 P00403020214
```

```
load 7960-7940 P00305000600
load 7914 S00103020002
load 7905 CP7905040000SCCP040701A
max-ephones 100
max-dn 500
ip source-address 10.115.33.177 port 2000
max-redirect 20
no service directed-pickup
timeouts ringing 10
voicemail 7189
max-conferences 8 gain -6
moh music-on-hold.au
web admin system name cisco password cisco
dn-webedit
time-webedit
transfer-system full-consult
transfer-pattern 92......
transfer-pattern 91..........
transfer-pattern 93......
transfer-pattern 94......
transfer-pattern 95......
transfer-pattern 96......
transfer-pattern 97......
transfer-pattern 98......
transfer-pattern 99......
transfer-pattern .T
secondary-dialtone 9
!
create cnf-files version-stamp 7960 Jul 13 2004 03:39:28
```

Step 2 If you have used the **transfer-mode** command to override the global transfer mode for an individual ephone-dn, use the **show running-config** or **show telephony-service ephone-dn** command to verify that setting.

Example:

```
Router# show running-config
!
ephone-dn 40 dual-line
number 451
description Main Number
huntstop channel
no huntstop
transfer-mode blind
```

Step 3 Use the **show telephony-service ephone-template** command to view ephone-template configurations.

Specify Transfer Patterns for Trunk-to-Trunk Calls and Conferences for SIP



Restriction Call transfer and conference restrictions apply when transfers or conferences are initiated toward external parties, like a PSTN trunk, a SIP trunk, or an H.323 trunk. The restrictions do not apply to transfers to local extensions.

Before you begin

Cisco Unified CME 9.5 or a later version.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **telephony-service**
4. **transfer-pattern** *transfer-pattern*
5. **exit**
6. Enter one of the following commands:
   - **voice register pool** *pool-tag*
   - **voice register template** *template-tag*
   - **ephone** *phone tag*
   - **ephone-template** *template-tag*
7. **transfer max-length** *max-length*
8. **exit**
9. **telephony-service**
10. **conference transfer-pattern**
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 for configuring Cisco Unified CME.
Step 4	**transfer-pattern** *transfer-pattern* **Example:** `Router(config-telephony)# transfer-pattern 1234...Router(config-telephony)# transfer-pattern 2468..`	Allows the transfer of calls from Cisco IP phones to specified directory numbers of phones other than Cisco IP phones. • *transfer-pattern*—String of digits for permitted call transfers. Wildcards are allowed. A maximum of 32 transfer patterns can be entered, using a separate command for each one.
Step 5	**exit** **Example:**	Exits telephony-service configuration mode and enters global configuration mode.

	Command or Action	Purpose
	`Router(config-telephony)# exit`	
Step 6	Enter one of the following commands: • **voice register pool** *pool-tag* • **voice register template** *template-tag* • **ephone** *phone tag* • **ephone-template** *template-tag* **Example:** `Router(config)# voice register pool 25`	Enters voice register pool configuration mode and creates a pool configuration for a Cisco Unified SIP IP phone in Cisco Unified CME or for a set of Cisco Unified SIP IP phones in Cisco Unified SIP SRST. • *pool-tag*—Unique number assigned to the pool. Range is 1 to 100. or Enters voice register template configuration mode and defines a template of common parameters for Cisco Unified SIP IP phones. • *template-tag*—Declares a template tag. Range is 1 to 10. or Enters ephone configuration mode. • phone-tag—Unique sequence number that identifies this ephone during configuration tasks. The maximum number of ephones is version and platform-specific. Type ? to display range.
Step 7	**transfer max-length** *max-length* **Example:** `Router(config-register-pool)# transfer max-length 7`	(Optional) Specifies the maximum length of the transfer number. • *max-length*—Maximum length of the transfer number. Range is 3 to 16.
Step 8	**exit** **Example:** `Router(config-register-pool)# exit`	Enters global configuration mode.
Step 9	**telephony-service** **Example:** `Router(config)# telephony-service`	Enters telephony-service configuration mode for configuring Cisco Unified CME.
Step 10	**conference transfer-pattern** **Example:** `Router(config-telephony)# conference transfer-pattern`	Enables a Cisco Unified CME system to apply transfer patterns to a conference call using conference softkeys or feature buttons.
Step 11	**end** **Example:** `Router(config-telephony)# end`	Exits telephony-service configuration mode and enters privileged EXEC mode.

Conference Max-Length

Conference calls are allowed when:

- both **conference transfer-pattern** and **transfer-pattern** commands are configured
- dialed digits match the configured transfer pattern

When conference max-length command is configured, the Cisco Unified CME will allow the conferences only if the dialed digits are within the max-length limit.

If configured, the conference max-length command does not impact call transfers.

Note If both **conference max-length** and **transfer max-length** commands are configured, the conference **max-length** command takes precedence for conferences.

Block Trunk-to-Trunk Call Transfers for SIP

To block call transfers to external destinations, perform the following steps.

Restriction Call transfer restrictions apply when transfers are initiated toward external parties, like a PSTN trunk, a SIP trunk, or an H.323 trunk. The restrictions do not apply to transfers to local extensions.

Before you begin

Cisco Unified CME 9.5 or a later version.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. Enter one of the following commands:
   - **voice register pool** *pool-tag*
   - **voice register template** *template-tag*
4. **transfer-pattern blocked**
5. **end**

DETAILED STEPS

Procedure

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

	Command or Action	Purpose
	`Router> enable`	
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	Enter one of the following commands: • **voice register pool** *pool-tag* • **voice register template** *template-tag* **Example:** `Router(config)# voice register template 5`	Enters voice register pool configuration mode and creates a pool configuration for a Cisco Unified SIP IP phone in Cisco Unified CME or for a set of Cisco Unified SIP IP phones in Cisco Unified SIP SRST. • *pool-tag*—Unique number assigned to the pool. Range is 1 to 100. Enters voice register template configuration mode and defines a template of common parameters for Cisco Unified SIP IP phones. • *template-tag*—Declares a template tag. Range is 1 to 10.
Step 4	**transfer-pattern blocked** **Example:** `Router(config-register-temp)# transfer-pattern blocked`	Blocks all call transfers for a specific Cisco Unified SIP IP phone or a set of Cisco Unified SIP IP phone.
Step 5	**end** **Example:** `Router(config-register-temp)# end`	Exits voice register template configuration mode and enters privileged EXEC mode.

Enable H.450.12 Capabilities

To enable H.450.12 capabilities globally or by individual dial peer when not all gateway endpoints in your network support H.450.2 and H.450.3 standards, perform the following steps. H.450.12 capabilities are disabled by default to minimize the risk of compatibility issues with other types of H.323 systems. Settings for individual dial peers override the global setting.

Restriction Cisco CME 3.0 and earlier versions do not support H.450.12.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **supplementary-service h450.12** [**advertise-only**]
5. **exit**

6. **dial-peer voice** *tag* **voip**
7. **supplementary-service h450.12**
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	**voice service voip** **Example:** `Router(config)# voice service voip`	(Optional) Enters voice service configuration mode to establish global call transfer and forwarding parameters.
Step 4	**supplementary-service h450.12** [**advertise-only**] **Example:** `Router(conf-voi-serv)# supplementary-service h450.12`	(Optional) Enables H.450.12 supplementary services capabilities globally for VoIP endpoints. • This command enables call-by-call detection of H.450 capabilities when some endpoints in your mixed network are H.450-capable and other endpoints are not. This command is disabled by default. • **advertise-only**—(Optional) Advertises H.450 capabilities to the remote end but does not require H.450.12 responses. Use this keyword on Cisco CME 3.1 or later systems if you have a mixed network containing Cisco CME 3.0 systems. This command is also used in dial-peer configuration mode to affect an individual dial peer.
Step 5	**exit** **Example:** `Router(conf-voi-serv)# exit`	(Optional) Exits voice-service configuration mode.
Step 6	**dial-peer voice** *tag* **voip** **Example:** `Router(config)# dial-peer voice 1 voip`	(Optional) Enters dial-peer configuration mode.

	Command or Action	Purpose
Step 7	**supplementary-service h450.12** **Example:** `Router(config-dial-peer)# supplementary-service h450.12`	(Optional) Enables H.450.12 supplementary services capabilities for an individual dial peer. This command is disabled by default. This command is also used in voice-service configuration mode to enable H.450.12 services globally. • If this command is enabled globally and enabled on a dial peer, the functionality is enabled for the dial peer. • If this command is enabled globally and disabled on a dial peer, the functionality is enabled for the dial peer. • If this command is disabled globally and enabled on a dial peer, the functionality is enabled for the dial peer. • If this command is disabled globally and disabled on a dial peer, the functionality is disabled for the dial peer. This is the default.
Step 8	**end** **Example:** `Router(config-dial-peer)# end`	Returns to privileged EXEC mode.

Enable H.323-to-H.323 Connection Capabilities

Vo IP-to-VoIP c onnections permit the termination and reorigination of transferred and forwarded calls over the VoIP network. VoIP-to-VoIP connections are used for hairpin call routing and for H.450 tandem gateways. The only type of VoIP-to-VoIP connection that is supported by Cisco CME 3.1 or a later version is H.323-to-H.323 connection.

VoIP-to-VoIP connections are disabled on the router by default, and they must be explicitly enabled to make use of hairpin call routing or an H.450 tandem gateway. In addition, you must configure a mechanism to direct transferred or forwarded calls to the hairpin or the H.450 tandem gateway, using one of the following methods:

- Enable H.450.12 capabilities globally or on the routes that your transfers and forwards take. See Enable H.450.12 Capabilities, on page 46.
- Explicitly disable H.450.2 and H.450.3 capabilities globally or on the routes that your transfers and forwards take. See Enable Call Transfer and Forwarding on SCCP Phones at System-Level, on page 28.

Restriction

- Codecs on all the VoIP dial peers of the H.450 tandem gateway must be the same.
- Only one codec type is supported in the VoIP network at a time, and there are only two codec choices: G.711 (A-law or mu-law) or G.729.
- Transcoding is not supported.
- Codec renegotiation is not supported. For example, if an H.323 call that uses a G.729 codec is received by a Cisco Unified CME system and is forwarded to a voice-mail system that requires a G.711 codec, the codec cannot be renegotiated from G.729 to G.711.
- H.323-to-SIP hairpin call routing is supported only with Cisco Unity Express. For more information, see Integrating Cisco CallManager Express with Cisco Unity Express.
- Cisco Unified Communications Manager must use a media termination point (MTP), intercluster trunk (ICT) mode, and slow start.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **allow-connections h323 to h323**
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 service voip** **Example:** `Router(config)# voice service voip`	Enters voice service configuration mode to establish global call transfer and forwarding parameters.
Step 4	**allow-connections h323 to h323** **Example:** `Router(conf-voi-serv)# allow-connections h323 to h323`	Enables VoIP-to-VoIP call connections. Use the **no** form of the command to disable VoIP-to-VoIP connections; this is the default.

	Command or Action	Purpose
Step 5	**end** **Example:** `Router(config-voi-serv)# end`	Returns to privileged EXEC mode.

Forward Calls Using Local Hairpin Routing

When Cisco Unified CME is used to forward calls that originate on phones that do not support the H.450.3 standard such as Cisco Unified Communications Manager phones, local hairpin routing must be used to forward the calls. For calling parties whose numbers match the pattern specified, the system automatically detects whether H.450.3 is supported and uses the appropriate method to forward calls.

To enable hairpin routing, you must denote the originating and terminating legs of the hairpin. To forward calls to Cisco Unity Express, connections must be allowed to a SIP trunk.

Optionally, you can disable the use of H.450.3 but this is not required because the system automatically detects calls on which H.450.3 is not supported and local hairpin routing is required when the calling-party numbers match the pattern specified.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **telephony-service**
4. **call-forward pattern** *pattern*
5. **calling-number local**
6. **exit**
7. **voice service voip**
8. **allow connections** *from-type* **to** *to-type*
9. **supplementary-service h450.3**
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.

	Command or Action	Purpose
Step 3	**telephony-service** **Example:** `Router(config)# telephony-service`	Enters telephony-service configuration mode.
Step 4	**call-forward pattern** *pattern* **Example:** `Router(config-telephony)# call-forward pattern 6000`	Specifies the calling-party numbers for which to allow call forwarding with automatic detection of whether H.450.3 is supported. If H.450.3 is supported, H.450.3 is used for the forward and, if not, local hairpin is used. • *pattern*—Digits to match for call forwarding. A pattern of .T forwards all calling parties.
Step 5	**calling-number local** **Example:** `Router(config-telephony)# calling-number local`	(Optional) Replaces a calling-party number and name with the forwarding-party (local) number and name for hairpin-forwarded calls only. • Before Cisco CME 3.3, this command must be used with Tool Command Language (Tcl) script app-h450-transfer.2.0.0.7 or a later version. The local-hairpin attribute-value (AV) pair must be set to 1.
Step 6	**exit** **Example:** `Router(config-telephony)# exit`	Exits telephony-service configuration mode.
Step 7	**voice service voip** **Example:** `Router(config)# voice service voip`	Enters voice-service configuration mode.
Step 8	**allow connections** *from-type* **to** *to-type* **Example:** `Router(conf-voi-serv)# allow connections h323 to sip`	Allows connections between specific types of endpoints in a network. • *from-type*—Originating endpoint type. Valid choices are **h323** and **sip**. • *to-type*—Terminating endpoint type. Valid choices are **h323** and **sip**.
Step 9	**supplementary-service h450.3** **Example:** `Router(conf-voi-serv)# no supplementary-service h450.3`	(Optional) Enables H.450.3 supplementary services capabilities exchange globally. This is the default. Use the **no** form of this command to disable H.450.3 capabilities globally. This command can also be used in dial-peer configuration mode to disable H.450.3 functionality for a single dial peer. **Note** If this command is disabled globally and either enabled or disabled on a dial peer, the functionality is disabled for the dial peer.

	Command or Action	Purpose
Step 10	**end** **Example:** `Router(config-voi-serv)# end`	Exits to privileged EXEC mode.

Enable H.450.7 and QSIG Supplementary Services at System-Level

To enable H.4350.7 capabilities and QSIG supplementary services on all dial peers, perform the following steps.

Restriction

- QSIG integration supports SCCP phones only.
- QSIG integration is exclusive; once QSIG integration is configured, QSIG transit node capability is disabled. There is no dial-peer control to enable either transit or originate/terminate capability on a call by call basis.
- If you enable QSIG supplementary services at a system-level, you cannot disable the capability on individual dial peers.

Before you begin

Cisco Unified CME 4.0 or a later version.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **supplementary-service h450.7**
5. **qsig decode**
6. **exit**
7. **voice service pots**
8. **supplementary-service qsig call-forward**
9. **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 service voip** **Example:** `Router(config)# voice service voip`	Enters VoIP voice-service configuration mode to define global call transfer and forwarding parameters.
Step 4	**supplementary-service h450.7** **Example:** `Router(config-voi-serv)# supplementary-service h450.7`	Enables H.450.7 supplementary services capabilities exchange at a system-level.
Step 5	**qsig decode** **Example:** `Router(config-voi-serv)# qsig decode`	Enables decoding for QSIG supplementary services.
Step 6	**exit** **Example:** `Router(config-voi-serv)# exit`	Exits VoIP voice-service configuration mode.
Step 7	**voice service pots** **Example:** `Router(config)# voice service pots`	Enters POTS voice-service configuration mode to define global call transfer and forwarding parameters.
Step 8	**supplementary-service qsig call-forward** **Example:** `Router(config-voi-serv)# supplementary-service qsig call-forward`	Enables QSIG call-forwarding supplementary services (ISO 13873) to forward calls to another number.
Step 9	**end** **Example:** `Router(config-voi-serv)# end`	Exits to privileged EXEC mode.

Enable H.450.7 and QSIG Supplementary Services on a Dial Peer

To enable H.4350.7 capabilities and QSIG supplementary services on an individual dial peer, perform the following steps.

Restriction

- QSIG integration supports SCCP phones only.
- QSIG integration is exclusive; once QSIG integration is configured, QSIG transit node capability is disabled. There is no dial-peer control to enable either transit or originate/terminate capability on a call by call basis.
- If you enable QSIG supplementary services at a system-level, you cannot enable or disable the capability on individual dial peers.

Before you begin

Cisco Unified CME 4.0 or a later version.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **qsig decode**
5. **exit**
6. **dial-peer voice** *tag* **voip**
7. **supplementary-service h450.7**
8. **exit**
9. **dial-peer voice** *tag* **pots**
10. **supplementary-service qsig call-forward**
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 VoIP voice-service configuration mode to define global call transfer and forwarding parameters.

	Command or Action	Purpose
Step 4	**qsig decode** **Example:** `Router(config-voi-serv)# qsig decode`	Enables decoding for QSIG supplementary services.
Step 5	**exit** **Example:** `Router(config-voi-serv)# exit`	Exits VoIP voice-service configuration mode.
Step 6	**dial-peer voice** *tag* **voip** **Example:** `Router(config)# dial-peer voice 1 voip`	Enters dial-peer configuration mode to define parameters for an individual dial peer.
Step 7	**supplementary-service h450.7** **Example:** `Router(config-dial-peer)# supplementary-service h450.7`	Enables H.450.7 supplementary services capabilities exchange on a single dial peer.
Step 8	**exit** **Example:** `Router(config-dial-peer)# exit`	Exits dial-peer configuration mode.
Step 9	**dial-peer voice** *tag* **pots** **Example:** `Router(config)# dial-peer voice 2 pots`	Enters dial-peer configuration mode to define parameters for an individual dial peer.
Step 10	**supplementary-service qsig call-forward** **Example:** `Router(config-dial-peer)# supplementary-service qsig call-forward`	Enables QSIG call-forwarding supplementary services (ISO 13873) to forward calls to another number.
Step 11	**end** **Example:** `Router(config-dial-peer)# end`	Exits to privileged EXEC mode.

Disable SIP Supplementary Services for Call Forward and Call Transfer

To disable REFER messages for call transfers or redirect responses for call forwarding from being sent to the destination by Cisco Unified CME, perform the following steps. You can disable these supplementary features if the destination gateway does not support them.

Restriction

- In Cisco Unified CME 4.2 and 4.3, when the **supplementary-service sip refer** command is enabled (default) and both the caller being transferred (transferee) and the phone making the transfer (transferor) are SIP, but the transfer-to phone is SCCP, Cisco Unified CME hairpins the call to the transfer-to phone after receiving the REFER request from transferor instead of sending the REFER request to the transferee.

Before you begin

Cisco Unified CME 4.1 or a later version.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. Enter one of the following commands:
   - **voice service voip**
   - **dial-peer voice** *tag* **voip**
4. **no supplementary-service sip moved-temporarily**
5. **no supplementary-service sip refer**
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	Enter one of the following commands:
	• **voice service voip**
	• **dial-peer voice** *tag* **voip**
	Example:
	`Router(config)# voice service voip or`
	`Router(config)# dial-peer voice 99 voip`
		Enters voice-service configuration mode to set global parameters for VoIP features.
		or
		Enters dial peer configuration mode to set parameters for a specific dial peer.

Step 4	**no supplementary-service sip moved-temporarily** **Example:**	Disables SIP redirect response for call forwarding either globally or for a dial peer.

	Command or Action	Purpose

	`Router(conf-voi-serv)# no supplementary-service sip moved-temporarily`
	`or`
	`Router(config-dial-peer)# no supplementary-service sip moved-temporarily`
		Sending redirect message to the destination is the default behavior.

Step 5	**no supplementary-service sip refer** **Example:** `Router(conf-voi-serv)# no supplementary-service sip refer or Router(config-dial-peer)# no supplementary-service sip refer`	Disables SIP REFER message for call transfers either globally or for a dial peer. Sending REFER message to the destination is the default behavior.
Step 6	**end** **Example:** `Router(config-voi-serv)# end or Router(config-dial-peer)# end`	Exits to privileged EXEC mode.

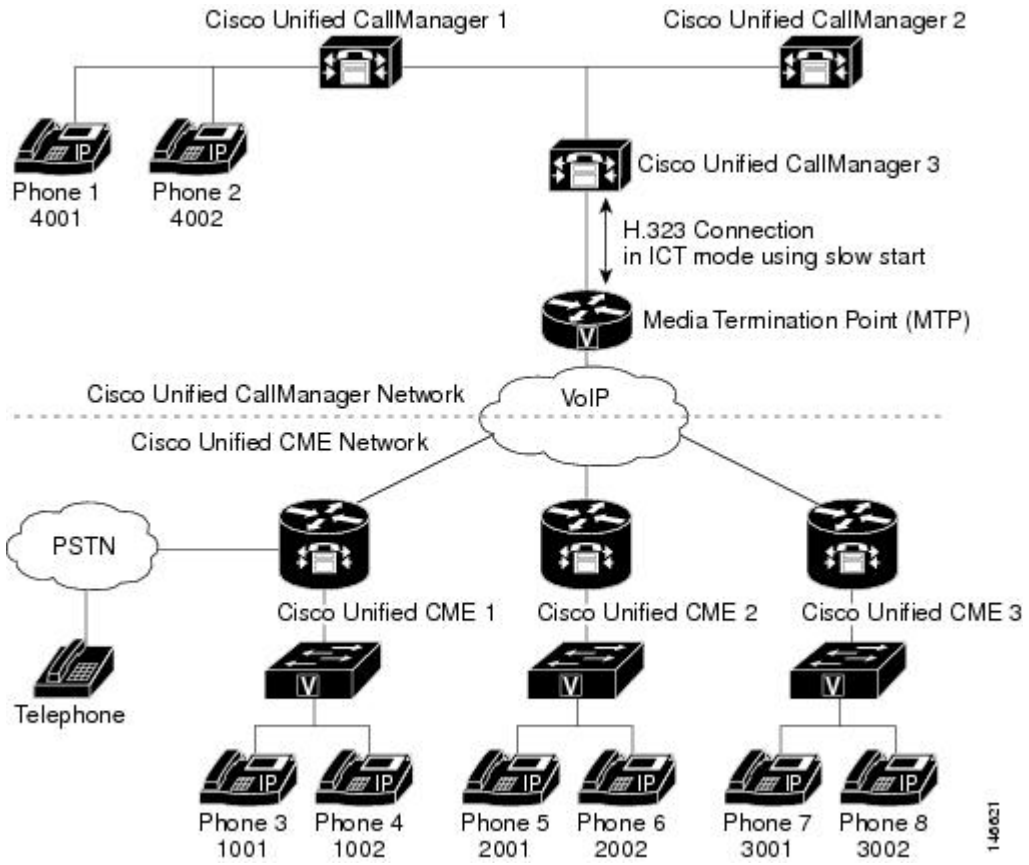
Enable Interworking with Cisco Unified Communications Manager

If Cisco CME 3.1 or later and Cisco Unified Communications Manager are used in the same network, some additional configuration is necessary, as described in the following sections:

- Configure Cisco CME 3.1 or Later to Interwork with Cisco Unified Communications Manager, on page 58
- Enable Cisco Unified Communications Manager to Interwork with Cisco Unified CME, on page 61
- Troubleshooting Call Transfer and Forward Configuration, on page 62

Figure 10: Network with Cisco Unified CME and Cisco Unified Communications Manager, on page 58 shows a network containing Cisco Unified CME and Cisco Unified Communications Manager systems.

Figure 10: Network with Cisco Unified CME and Cisco Unified Communications Manager



Prerequisites

- Cisco Unified CME must be configured to forward calls using local hairpin routing. For configuration information, see Forward Calls Using Local Hairpin Routing, on page 50.

Configure Cisco CME 3.1 or Later to Interwork with Cisco Unified Communications Manager

All of the commands in this section are optional because they are set by default to work with Cisco Unified Communications Manager. They are included here only to explain how to implement optional capabilities or return non default settings to their defaults.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **h323**
5. **telephony-service ccm-compatible**
6. **h225 h245-address on-connect**
7. **exit**
8. **supplementary-service h225-notify cid-update**

9. **exit**
10. **voice class h323** *tag*
11. **telephony-service ccm-compatible**
12. **h225 h245-address on-connect**
13. **exit**
14. **dial-peer voice** *tag* **voip**
15. **supplementary-service h225-notify cid-update**
16. **voice-class h323** *tag*
17. **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 configuration mode to establish global parameters.
Step 4	**h323** **Example:** `Router(conf-voi-serv)# h323`	Enters H.323 voice-service configuration mode.
Step 5	**telephony-service ccm-compatible** **Example:** `Router(conf-serv-h323)# telephony-service ccm-compatible`	(Optional) Globally enables a Cisco CME 3.1 or later system to detect Cisco Unified Communications Manager and exchange calls with it. This is the default configuration. • Use the **no** form of this command to disable Cisco Unified Communications Manager detection and exchange. We do not recommend using the **no** form of the command. • Using this command in an H.323 voice class definition allows you to specify this behavior for an individual dial peer.
Step 6	**h225 h245-address on-connect** **Example:**	(Optional) Globally enables a delay for the H.225 message exchange of an H.245 transport address until a call is connected. The delay allows

	Command or Action	Purpose
	`Router(conf-serv-h323)# h225 h245-address on-connect`	Cisco Unified Communications Manager to generate local ringback for calls to Cisco Unified CME phones. This is the default configuration. • The **no** form of this command disables the delay. We do not recommend using the **no** form of the command. • Using this command in an H.323 voice class definition allows you to specify this behavior for an individual dial peer.
Step 7	**exit** **Example:** `Router(conf-serv-h323)# exit`	Exits H.323 voice-service configuration mode.
Step 8	**supplementary-service h225-notify cid-update** **Example:** `Router(conf-voi-serv)# supplementary-service h225-notify cid-update`	(Optional) Globally enables H.225 messages with caller-ID updates to be sent to Cisco Unified Communications Manager. This is the default configuration. • The **no** form of the command disables caller-ID update. We do not recommend using the **no** form of the command. This command is also used in dial-peer configuration mode to affect a single dial peer. • If this command is enabled globally and enabled on a dial peer, the functionality is enabled for that dial peer. This is the default. • If this command is enabled globally and disabled on a dial peer, the functionality is disabled for that dial peer. • If this command is disabled globally and either enabled or disabled on a dial peer, the functionality is disabled for that dial peer.
Step 9	**exit** **Example:** `Router(config-voice-service)# exit`	Exits voice-service configuration mode.
Step 10	**voice class h323** *tag* **Example:** `Router(config)# voice class h323 48`	(Optional) Creates a voice class that contains commands to be applied to one or more dial peers.
Step 11	**telephony-service ccm-compatible** **Example:** `Router(config-voice-class)# telephony-service ccm-compatible`	(Optional) Enables the dial peer to exchange calls with a Cisco Unified Communications Manager system when this voice class is applied to a dial peer. This is the default configuration.

	Command or Action	Purpose
		• The **no** form of the command disables call exchange with Cisco Unified Communications Manager. We do not recommend using the **no** form of the command.
Step 12	**h225 h245-address on-connect** **Example:** `Router(config-voice-class)# h225 h245-address on-connect`	(Optional) Enables the calls that use this dial peer to delay the exchange of H.225 messages that contain the H.245 transport address until calls are connected, when this voice class is applied to a dial peer. The delay allows the playing of local ringback for calls from Cisco Unified Communications Manager. This is the default configuration. • The **no** form of this command disables the delay. We do not recommend using the **no** form of the command.
Step 13	**exit** **Example:** `Router(config-voice-class)# exit`	Exits voice-class configuration mode.
Step 14	**dial-peer voice** *tag* **voip** **Example:** `Router(config)# dial-peer voice 28 voip`	(Optional) Enters dial-peer configuration mode to set parameters for an individual dial peer.
Step 15	**supplementary-service h225-notify cid-update** **Example:** `Router(config-dial-peer)# no supplementary-service h225-notify cid-update`	(Optional) Enables H.225 messages with caller-ID updates to Cisco Unified Communications Manager for a specific dial peer. This is the default configuration. • The **no** form of the command disables caller-ID updates. We do not recommend using the **no** form of the command.
Step 16	**voice-class h323** *tag* **Example:** `Router(config-dial-peer)# voice-class h323 48`	(Optional) Applies the previously defined voice class with the specified tag number to this dial peer.
Step 17	**end** **Example:** `Router(config-dial-peer)# end`	Exits to privileged EXEC mode.

What to do next

Set up Cisco Unified Communications Manager using the configuration procedure in the Enable Cisco Unified Communications Manager to Interwork with Cisco Unified CME, on page 61.

Enable Cisco Unified Communications Manager to Interwork with Cisco Unified CME

To enable Cisco Unified Communications Manager to interwork with Cisco CME 3.1 or a later version, perform the following steps in addition to the normal Cisco Unified Communications Manager configuration.

Procedure

Step 1 Set Cisco Unified Communications Manager service parameters. From Cisco Unified Communications Manager Administration, choose Service Parameters. Choose the Cisco Unified Communications Manager service, and make the following settings:

- Set the H323 FastStart Inbound service parameter to False.
- Set the Send H225 User Info Message service parameter to H225 Info for Ring Back.

Step 2 Configure Cisco Unified CME as an ICT in the Cisco Unified Communications Manager network. For information about different intercluster trunk types and configuration instructions, see Cisco Unified Communications Manager documentation.

Step 3 Ensure that the Cisco Unified Communications Manager network uses an MTP. The MTP is required to provide DSP resources for transcoding and for sending and receiving G.729 calls to Cisco Unified CME. All media streams between Cisco Unified Communications Manager and Cisco Unified CME must pass through the MTP because Cisco CME 3.1 does not support transcoding. For more information, see Cisco Unified Communications Manager documentation.

Step 4 Set up dial peers to establish routing using the instructions in the Dial Peer Configuration on Voice Gateway Routers guide.

Troubleshooting Call Transfer and Forward Configuration

Procedure

Step 1 If you encounter lack of ringback on direct calls from a Cisco Unified Communications Manager phone to an IP phone on a Cisco Unified CME system, check the **show running-config** command output to ensure that the following two commands do *not* appear: **no h225 h245-address on-connect** and **no telephony-service ccm-compatible**. These commands should be enabled, which is their default state.

Step 2 Use the **debug h225 asn1** command to display the H.323 messages that are sent from the Cisco Unified CME system to the Cisco Unified Communications Manager system to see if the H.245 address is being sent too early.

Step 3 For calls that are routed using VoIP-to-VoIP connections, use the **show voip rtp connections detail** command to display the call identification number, IP addresses, and port numbers involved for all VoIP call legs. This command includes VoIP-to-POTS and VoIP-to-VoIP call legs. The following is sample output for this command:

```
Router# show voip rtp connections detail
VoIP RTP active connections :
No. CallId  dstCallId  LocalRTP   RmtRTP   LocalIP          RemoteIP
1   7       8          16586      22346   172.27.82.2     172.29.82.2
2   8       7          17010      16590   172.27.82.2     209.165.202.129

Found 2 active RTP connections
```

Step 4 Use the **show call prompt-mem-usage detail** command to see information on ringback tone generation that uses the interactive voice response (IVR) prompt playback mechanism. This ringback is needed for hairpin transfers that are committed during the alerting-of-the-transfer-destination phase of the call and for calls to destinations that do not provide in-band ringback tone, such as IP phones (FXS analog ports do provide in-band ringback tone). Ringback tone is played to the transferred party by the Cisco Unified CME system that performs the transfer (the system attached to the transferring

party). The system automatically generates tone prompts as needed based on the network-locale setting for the Cisco Unified CME system.

If you are not getting ringback tone when you should, use the **show call prompt-mem-usage** command to ensure that the correct prompt is loaded and playing. The following sample output indicates that a prompt is playing ("Number of prompts playing") and indicates the country code used for the prompt (GB for Great Britain) and the codec.

```
Router# show call prompt-mem-usage detail
Prompt memory usage:

 config'd   wait  active  free   mc    total    ms      total
file(s) 0200 0001 -001 00200 00001 00002
memory 02097152 00003000 00000000 02094152 00003000
Prompt load counts: (counters reset 0)
success 0(1st try) 0(2nd try), failure 0
Other mem block usage:
mcDynamic mcReader
gauge 00001 00001
Number of prompts playing: 1
Number of start delays : 0
MCs in the ivr MC sharing table
==============================
Media Content: NoPrompt (0x83C64554)
URL:
cid=0, status=MC_READY size=24184 coding=g711ulaw refCount=0
Media Content: tone://GB_g729_tone_ringback (0x83266EC8)
URL: tone://GB_g729_tone_ringback
```

Configure SIP-to-SIP Phone Call Forwarding

To configure SIP-to-SIP call forwarding using a back-to-back user agent (B2BUA) which allows call forwarding on any dial peer, perform the following steps.

Restriction

- SIP-to-SIP call forwarding is invoked only if that phone is dialed directly. Call forwarding is not invoked when the phone number is called through a sequential, longest-idle, or peer hunt group.
- If call forwarding is configured for a hunt group member, call forward is ignored by the hunt group.
- In Cisco Unified CME 4.1 and later versions, Call Forward All requires SIP phones to be configured with a directory number (using **dn** keyword in **number** command); direct line numbers are not supported.

Before you begin

- Cisco CME 3.4 or a later version.
- Connections between specific types of endpoints in a Cisco IP-to-IP gateway must be configured by using the **allow-connections** command. For configuration information, see Enable Calls in Your VoIP Network.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register dn** *dn-tag*
4. **call-forward b2bua all** *directory- number*
5. **call-forward b2bua busy** *directory- number*
6. **call-forward b2bua mailbox** *directory- number*
7. **call-forward b2bua night-service** *directory- number*
8. **call-forward b2bua noan** *directory- number* **timeout** *seconds*
9. **call-forward b2bua unreachable** *directory- 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.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice register dn** *dn-tag* **Example:** `Router(config)# voice register dn 1`	Enters voice register dn mode to define a directory number for a SIP phone, intercom line, voice port, or an MWI.
Step 4	**call-forward b2bua all** *directory- number* **Example:** `Router(config-register-dn)# call-forward b2bua all 5005`	Enables call forwarding for a SIP back-to-back user agent so that all incoming calls will be forwarded to the designated directory-number. • In Cisco CME 3.4 and Cisco Unified CME 4.0, this command is also available in voice register pool configuration mode. The configuration under voice register dn takes precedence over the configuration under voice register pool. • If the **call-forward b2bua all** command is configured in voice register pool configuration mode, it applies to all directory numbers on the phone.
Step 5	**call-forward b2bua busy** *directory- number* **Example:** `Router(config-register-dn)# call-forward b2bua busy 5006`	Enables call forwarding for a SIP back-to-back user agent so that incoming calls to an extension that is busy will be forwarded to the designated directory number.

	Command or Action	Purpose
		• In Cisco CME 3.4 and Cisco Unified CME 4.0, this command is also available in voice register pool configuration mode. The configuration under voice register dn takes precedence over the configuration under voice register pool.
Step 6	**call-forward b2bua mailbox** *directory- number* **Example:** `Router(config-register-dn)# call-forward b2bua mailbox 5007`	Enables call forwarding for a SIP back-to-back user agent so that incoming calls that have been forwarded to a busy or no-answer extension will be forwarded to the recipient's voice mail. • In Cisco CME 3.4 and Cisco Unified CME 4.0, this command is also available in voice register pool configuration mode. The configuration under voice register dn takes precedence over the configuration under voice register pool.
Step 7	**call-forward b2bua night-service** *directory- number* **Example:** `Router(config-register-dn)# call-forward b2bua night-service 5007`	Enables call forwarding for a SIP back-to-back user agent so that incoming calls that have been forwarded to a busy or no-answer extension will be forwarded to the recipient's voice mail. • In Cisco CME 3.4 and Cisco Unified CME 4.0, this command is also available in voice register pool configuration mode. The configuration under voice register dn takes precedence over the configuration under voice register pool.

Step 8	**call-forward b2bua noan** *directory- number* **timeout** *seconds*
	Example:
	`Router(config-register-dn)# call-forward b2bua noan 5010 timeout 10 or`
	`Router(config-register-pool)# call-forward b2bua noan 5010 timeout 10`
		Enables call forwarding for a SIP back-to-back user agent so that incoming calls to an extension that does not answer will be forwarded to the designated directory number.
		• In Cisco CME 3.4 and Cisco Unified CME 4.0, this command is also available in voice register pool configuration mode. The configuration under voice register dn takes precedence over the configuration under voice register pool.
		• **timeout** *seconds*—Duration that a call can ring before it is forwarded to the destination directory number. Range: 3 to 60000. Default: 20.

Step 9	**call-forward b2bua unreachable** *directory- number* **Example:** `Router(config-register-dn)# call-forward b2bua unreachable 5009 or Router(config-register-pool)# call-forward b2bua unreachable 5009`	(Optional) Enables call forwarding for a SIP back-to-back user agent so that calls can be forwarded to a phone that has not registered in Cisco Unified CME. • Target directory-number must be configured in Cisco Unified CME. • In Cisco CME 3.4 and Cisco Unified CME 4.0, this command is also available in voice register pool configuration mode. The configuration under voice

	Command or Action	Purpose
		register dn takes precedence over the configuration under voice register pool. • This command was removed in Cisco Unified CME 4.1.
Step 10	**end** **Example:** `Router(config-register-dn)# end`	Exits to privileged EXEC mode.

Configure Call Forward Unregistered for SIP IP Phones

Before you begin

- Cisco Unified CME 8.6 or a later version.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register dn** *tag*
4. **call-forward b2bua unregistered** *directory-number*
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 dn** *tag* **Example:** `Router(config)#voice register dn 20`	Enters voice register dn mode to define a directory number for a SIP phone, intercom line, voice port, or an MWI.

	Command or Action	Purpose
Step 4	**call-forward b2bua unregistered** *directory-number* **Example:** `Router(config-register-dn)#call-forward b2bua unregistered 2345`	Enables call forwarding for a SIP back-to-back user agent so that all incoming calls are forwarded to the unregistered directory-number.
Step 5	**end** **Example:** `Router(config-ephone)# end`	Returns to privileged EXEC mode.

Troubleshooting Tips for Call Forward Unregistered

- Use the **show dial-peer voice summary** command to check whether a CFU dial peer is created or removed.
- Enable **deb voice reg event**, **deb voice reg state**, and **deb voice reg error** commands to trace the creation and deletion of the CFU dial peer.
- Enable **deb voice reg event**, **deb voip ccapi inout**, **deb voip app callsetup**, **deb voip app core**, **deb voip app state**, and **deb voip app error** commands to trace the call flow for CFU.

Configure Keepalive Timer Expiration in SIP Phones

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **sip**
5. **registrar server** [**expires** [**max** *seconds*] [**min** *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.

	Command or Action	Purpose
Step 3	**voice service voip** **Example:** `Router(conf)# voice service voip`	Enters voice-service configuration mode and specifies voice-over-IP encapsulation.
Step 4	**sip** **Example:** `Router(conf-serv)# sip`	Enters SIP configuration mode.
Step 5	**registrar server** [**expires** [**max** *seconds*] [**min** *seconds*]] **Example:** `Router(conf-serv-sip)# registrar server expires max 250 min 75`	Enables SIP registrar functionality in Cisco Unified CME. • expires—(Optional) Sets the active time for an incoming registration. • max sec—(Optional) Maximum time for a registration to expire, in seconds. Range: 120 to 86400. • min sec—(Optional) Minimum time for a registration to expire, in seconds.
Step 6	**end** **Example:** `Router (conf-serv-sip)# end`	Returns to privileged EXEC mode.

Configure Call-Forwarding-All Softkey URI on SIP Phones

To specify the uniform resource identifier (URI) for the call forward all (CfwdAll) softkey on supported SIP phones, perform the following steps. This URI and the call forward number is sent to Cisco Unified CME when a user enables Call Forward All on a SIP phone.

Restriction

- This feature is supported only on Cisco Unified IP Phone 7911G, 7941G, 7941GE, 7961G, 7961GE, 7970G, and 7971GE.
- If a user enables Call Forward All using the CfwdAll softkey, it is enabled on the primary line.

Before you begin

- Cisco Unified CME 4.1 or a later version.
- The **mode cme** command must be enabled in Cisco Unified CME.
- Call Forward All must be enabled on the directory number. For information, see Configure SIP-to-SIP Phone Call Forwarding, on page 63.

SUMMARY STEPS

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

4. **call-feature-uri cfwdall** *service-uri*
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 global parameters for all supported SIP phones in a Cisco Unified CME environment.
Step 4	**call-feature-uri cfwdall** *service-uri* **Example:** `Router(config-register-global)# call-feature-uri cfwdall http://1.4.212.11/cfwdall`	Specifies the URI for soft keys on SIP phones connected to a Cisco Unified CME router.
Step 5	**end** **Example:** `Router(config-register-global)# end`	Exits to privileged EXEC mode.

Specify Number of 3XX Responses To be Handled on SIP Phones

To specify how many subsequent 3XX responses an originating SIP phone can handle for a single call when the terminating side is a forwarding party which does not use B2BUA, perform the following steps.

Before you begin

- Cisco CME 3.4 or a later version.
- The **mode cme** command must be enabled

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register global**
4. **phone-redirect-limit** *number*

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	**phone-redirect-limit** *number* **Example:** `Router(config-register-global)# phone-redirect-limit 8`	Changes the default number of 3XX responses a SIP phone that originates a call can handle for a single call. • Default: 5
Step 5	**end** **Example:** `Router(config-register-global)# end`	Exits to privileged EXEC mode.

Configure Call Transfer on SIP Phones

To create and apply a template to enable call transfer softkeys on an individual SIP phone in Cisco Unified CME, perform the following steps.

Restriction

- Blind transfer is not supported on certain phones such as Cisco Unified IP Phone 7911G, 7941G, 7941GE, 7961G, 7961GE, 7970G, or 7971GE.
- In Cisco Unified CME 4.1, the soft key display can be customized only for certain IP phones, such as Cisco Unified IP Phone 7911G, 7941G, 7941GE, 7961G, 7961GE, 7970G, and 7971GE. For configuration information, see Modify Softkey Display on SIP Phone.

Before you begin

Cisco CME 3.4 or a later version.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice register template** *template-tag*
4. **transfer-attended**
5. **transfer-blind**
6. **exit**
7. **voice register pool** *pool-tag*
8. **template** *template-tag*
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 template** *template-tag* **Example:** `Router(config)# voice register template 1`	Enters voice register template configuration mode to define a template of common parameters for SIP phones in Cisco Unified CME. • Range: 1 to 5
Step 4	**transfer-attended** **Example:** `Router(config-register-template)# transfer-attended`	Enable a soft key for attended transfer on any supported SIP phone that uses a template in which this command is configure.
Step 5	**transfer-blind** **Example:** `Router(config-register-template)# transfer-blind`	Enable a soft key for blind transfer on any supported SIP phone that uses a template in which this command is configure.
Step 6	**exit** **Example:** `Router(config-register-template)# exit`	Exits configuration mode to the next highest mode in the configuration mode hierarchy.
Step 7	**voice register pool** *pool-tag* **Example:** `Router(config)# voice register pool 3`	Enters voice register pool configuration mode to set phone-specific parameters for SIP phones.

	Command or Action	Purpose
Step 8	**template** *template-tag* **Example:** `Router(config-register-pool)# voice register pool 1`	Applies a template created with the **voice register template** command. • *template-tag*—Range: 1 to 5
Step 9	**end** **Example:** `Router(config-register-pool)# end`	Exits to privileged EXEC mode.

Configuration Examples for Call Transfer and Forwarding

Example for Configuring H.450.2 and H.450.3 Support

The following example sets all transfers and forwards that are initiated by a Cisco CME 3.0 or later system to use the H.450 standards, globally enables H.450.2 and H.450.3 capabilities, and disables those capabilities for dial peer 37. The **supplementary-service** commands under voice-service configuration mode are not necessary because these values are the default, but they are shown here for illustration.

```
telephony-service
transfer-system full-consult
transfer-pattern .T
call-forward pattern .T
!
voice service voip
supplementary-service h450.2
supplementary-service h450.3
!
dial-peer voice 37 voip
destination-pattern 555....
session target ipv4:10.5.6.7
no supplementary-service h450.2
no supplementary-service h450.3
```

Example for Configuring Basic Call Forwarding

The following example sets up forwarding for extension 2777 to extension 2513 on all calls, busy, and no answer. During night service hours, calls are forwarded to a different number, extension 2879.

```
ephone-dn 20
 number 2777
 call-forward all 2513
```

```
 call-forward busy 2513
 call-forward noan 2513 timeout 45
 call-forward night-service 2879
```

Example for Configuring Call Forwarding Blocked for Local Calls

In the following example, extension 2555 is configured to not forward local calls that are internal to the Cisco Unified CME system. Extension 2222 dials extension 2555. If 2555 is busy, the caller hears a busy tone. If 2555 does not answer, the caller hears ringback. The internal call is not forwarded.

```
ephone-dn 25
 number 2555
 no forward local-calls
 call-forward busy 2244
 call-forward noan 2244 timeout 45
```

Example for Configuring Transfer Patterns

The following example shows how to configure transfer patterns beginning with 1234:

```
Router# configure terminal
Router(config)# telephony-service
Router(config-telephony)# transfer-pattern 1234
```

Example for Configuring Maximum Length of Transfer Number

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

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

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

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

Example for Configuring Conference Transfer Patterns

The following example configures transfer patterns that allow conference calls:

```
Router# configure terminal
Router(config)# telephony-service
Router(config-telephony)# transfer-pattern 1357
```

```
Router(config-telephony)# transfer-pattern 222....
Router(config-telephony)# conference transfer-pattern
```

Example for Blocking All Call Transfers

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

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

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

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

Example for Configuring Selective Call Forwarding

The following example sets call forwarding on busy and no answer for ephone-dn 38 only for its primary number, 2777. Callers who dial 2778 will hear a busy signal if the ephone-dn is busy or ringback if there is no answer.

```
ephone-dn 38
number 2777 secondary 2778
call-forward busy 3000 primary
call-forward noan 3000 primary timeout 45
```

Example for Configuring Call Transfer

The following example limits transfers from ephone 6, extension 2977, to numbers containing a maximum of 8 digits.

```
telephony-service
load 7910 P00403020214
load 7960-7940 P00305000600
load 7914 S00103020002
load 7905 CP7905040000SCCP040701A
load 7912 CP7912040000SCCP040701A
max-ephones 100
max-dn 500
ip source-address 10.104.8.205 port 2000
max-redirect 20
system message XYZ Inc.
create cnf-files version-stamp 7960 Jul 13 2004 03:39:28
```

```
voicemail 7189
max-conferences 8 gain -6
moh music-on-hold.au
web admin system name admin1 password admin1
dn-webedit
time-webedit
transfer-system full-consult
transfer-pattern 91.........
transfer-pattern 92......
transfer-pattern 93......
transfer-pattern 94......
transfer-pattern 95......
transfer-pattern 96......
transfer-pattern 97......
transfer-pattern 98......
transfer-pattern 99......
secondary-dialtone 9
fac standard
ephone-template 2
transfer max-length 8
ephone-dn 4
number 2977
ephone 6
button 1:4
ephone-template 2
```

Example for Configuring Call Transfer Recall for SCCP Phones

The following example shows that transfer recall is enabled globally. After 60 seconds an unanswered call is forwarded back to the phone that initiated the transfer (transferor).

```
telephony-service
 max-ephones 100
 max-dn 240
 timeouts transfer-recall 60
 max-conferences 8 gain -6
 transfer-system full-consult
```

The following example shows that transfer recall is enabled for extension 1030 (ephone-dn 103), which is assigned to ephone 3. If extension 1030 forwards a call and the transfer-to party does not answer, after 60 seconds the unanswered call is sent back to extension 1030 (transferor). The **timeouts transfer-recall** command can also be set in an ephone-dn template and applied to one or more directory numbers.

```
ephone-dn  103
```

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

Example for Configuring Call-Transfer Recall for SIP Phones

The following example shows that transfer recall is enabled globally. After 20 seconds, an unanswered call is forwarded back to the phone that initiated the transfer (transferor).

```
voice register global
 mode cme
 source-address 8.39.17.29 port 5060
 timeouts transfer-recall 20
 max-dn 100
 max-pool 100
 tftp-path flash:
 create profile sync 0342574150542703
 keepalive 140
 auto-register
```

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

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

Example for Enabling H.450.12 Capabilities

The following example globally disables H.450.12 capabilities and then enables them only on dial peer 24.

```
voice service voip
no supplementary-service h450.12
!
dial-peer voice 24 voip
destination-pattern 555....
session target ipv4:10.5.6.7
supplementary-service h450.12
```

Example for Enabling H.450.7 and QSIG Supplementary Services

The following example implements QSIG supplementary services on extension 74367 and globally enables H.450.7 supplementary services and QSIG call-forwarding supplementary services.

```
telephony-service
voicemail 74398
transfer-system full-consult
ephone-dn 25
number 74367
mwi qsig
call-forward all 74000
voice service voip
supplementary-service h450.7
voice service pots
supplementary-service qsig call-forward
```

Example for Configuring Cisco Unified CME and Cisco Unified Communications Manager in Same Network

The following example shows a running configuration for a Cisco CME 3.1 or later router that has a Cisco Unified Communications Manager in its network.

```
Router# show running-config

version 12.3
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname Router
!
enable password pswd
!
aaa new-model
!
!
aaa session-id common
no ip subnet-zero
!
ip dhcp pool phone1
 host 172.24.82.3 255.255.255.0
```

```
 client-identifier 0100.07eb.4629.9e
 default-router 172.24.82.2
 option 150 ip 172.24.82.2
!
ip dhcp pool phone2
 host 172.24.82.4 255.255.255.0
 client-identifier 0100.0b5f.f932.58
 default-router 172.24.82.2
 option 150 ip 172.24.82.2
!
ip cef
no ip domain lookup
no mpls ldp logging neighbor-changes
no ftp-server write-enable
!
voice service voip
 allow-connections h323 to h323
!
voice class codec 1
 codec preference 1 g711ulaw
!
no voice hpi capture buffer
no voice hpi capture destination
!
interface FastEthernet0/0
 ip address 172.24.82.2 255.255.255.0
 duplex auto
 speed auto
 h323-gateway voip interface
 h323-gateway voip bind srcaddr 172.24.82.2
!
ip classless
ip route 0.0.0.0 0.0.0.0 172.24.82.1
ip route 192.168.254.254 255.255.255.255 172.24.82.1
!
ip http server
!
tftp-server flash:P00303020700.bin
!
voice-port 1/0/0
```

```
!
voice-port 1/0/1
!
dial-peer cor custom
!
dial-peer voice 1001 voip
 description points-to-CCM
 destination-pattern 1.T
 voice-class codec 1
 session target ipv4:172.26.82.10
!
dial-peer voice 1002 voip
 description points to router
 destination-pattern 4...
 voice-class codec 1
 session target ipv4:172.25.82.2
!
dial-peer voice 1 pots
 destination-pattern 3000
 port 1/0/0
!
dial-peer voice 1003 voip
 destination-pattern 26..
 session target ipv4:10.22.22.38
!
!
telephony-service
 load 7960-7940 P00303020700
 max-ephones 48
 max-dn 15
 ip source-address 172.24.82.2 port 2000
 create cnf-files version-stamp Jan 01 2002 00:00:00
 keepalive 10
 max-conferences 4
 moh minuet.au
 transfer-system full-consult
 transfer-pattern ....
!
ephone-dn  1
 number 3001
```

```
 name abcde-1
 call-forward busy 4001
!
ephone-dn  2
 number 3002
 name abcde-2
!
ephone-dn  3
 number 3003
 name abcde-3
!
ephone-dn  4
 number 3004
 name abcde-4
!
ephone  1
 mac-address 0003.EB27.289E
 button  1:1 2:2
!
ephone  2
 mac-address 000D.39F9.3A58
 button  1:3 2:4
!
line con 0
 exec-timeout 0 0
 logging synchronous
line aux 0
line vty 0 4
 password pswd
!
end
```

Example for Configuring H.450 Tandem Gateway Working with Cisco Unified CME and Cisco Unified Communications Manager

The following example shows a sample configuration for a Cisco CME 3.1 or later system that is linked to an H.450 tandem gateway that serves as a proxy for Cisco Unified Communications Manager.

```
Router# show running-config

Building configuration...
```

```
Current configuration : 1938 bytes
!
version 12.3
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname Router
!
boot-start-marker
boot-end-marker
!
enable password pswd
!
aaa new-model
!
aaa session-id common
no ip subnet-zero
!
ip cef
no ip domain lookup
no ftp-server write-enable
no scripting tcl init
no scripting tcl encdir
!
voice call send-alert
!
voice service voip
 allow-connections h323 to h323
 supplementary-service h450.12
 h323
!
voice class codec 1
codec preference 1 g711ulaw
codec preference 2 g729r8
codec preference 3 g729br8
!
interface FastEthernet0/0
ip address 172.27.82.2 255.255.255.0
```

```
duplex auto
speed auto
h323-gateway voip interface
h323-gateway voip h323-id host24
!
ip classless
ip route 0.0.0.0 0.0.0.0 172.26.82.1
ip route 0.0.0.0 0.0.0.0 172.27.82.1
ip http server
!
dial-peer cor custom
!
dial-peer voice 1001 voip
description points-to-CCM
 destination-pattern 4...
session target ipv4:172.24.89.150
!
dial-peer voice 1002 voip
description points to CCME1
destination-pattern 28..
session target ipv4:172.24.22.38
!
dial-peer voice 1003 voip
description points to CCME3
destination-pattern 9...
 session target ipv4:192.168.1.29
!
dial-peer voice 1004 voip
description points to CCME2
destination-pattern 29..
session target ipv4:172.24.22.42
!
line con 0
exec-timeout 0 0
logging synchronous
line aux 0
line vty 0 4
password pswd
!
end
```

Example for Configuring Call Forward to Cisco Unity Express

The following example enables the ability to forward calls that originate from Cisco Unified Communications Manager phones and are routed through a Cisco Unified CME system to a Cisco Unity Express extension. Call forwarding is enabled for all calling parties, H.450.3 is disabled, and connections are allowed to SIP endpoints.

```
telephony-service
 call-forward pattern .T


voice service voip
 no supplementary-service h450.3
 allow connections from h323 to sip
```

Example for Configuring Call Forward Unregistered for SIP IP Phones

The following example shows CFU configured for voice register dn 20:

```
!
!
!
voice service voip
 allow-connections sip to sip
 sip
  registrar server expires max 250 min 75
!
!
voice register global
 mode cme
 source-address 10.100.109.10 port 5060
 bandwidth video tias-modifier 256 negotiate end-to-end
 max-dn 200
 max-pool 42
 url directory http://1.4.212.11/localdirectory
 create profile sync 0004625832149157
!
voice register dn  20
 number 10
 call-forward b2bua unregistered 2345
!
voice register pool  1
 number 1 dn 20
 id mac 1111.1111.1111
```

```
 camera
 video
!
voice register pool  2
id mac 0009.A3D4.1234
```

Example for Configuring Keepalive Timer Expiration in SIP Phones

The following example shows the minimum and maximum registrar server expiration time for SIP phones:

```
Router#show run
!
!
!
!
!
!
voice service voip
allow-connections sip to sip
sip
registrar server expires max 250 min 75
!
!
voice register global
mode cme
source-address 10.100.109.10 port 5060
bandwidth video tias-modifier 256 negotiate end-to-end
max-dn 200
```

Where to Go Next

If you are finished modifying the configuration, generate a new configuration file and restart the phones. See Generate Configuration Files for Phones.

Softkeys

To block the function of the call-forward-all or transfer softkey without removing the key display or to remove the softkey from one or more phones, see Customize Softkeys.

Feature Access Codes (FACs)

Phone users can activate and deactivate a phone's call-forward-all setting by using a feature access code (FAC) instead of a soft key on the phone if standard or custom FACs have been enabled for your system. The following are the standard FACs for call forward all:

- **callfwd all**—Call forward all calls. Standard FAC is **1 plus an optional target extension.

- **callfwd cancel**—Cancel call forward all calls. Standard FAC is **2.

For more information about FACs, see Feature Access Codes.

Night Service

Calls can be automatically forwarded during night service hours, but you must define the night-service periods, which are the dates or days and hours during which night service will be active. For instance, you may want to designate night service periods that include every weeknight between 5 p.m. and 8 a.m. and all day every Saturday and Sunday. For more information, see Configure Call Coverage Features.

Feature Information for Call Transfer and Forwarding

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 www.cisco.com/go/cfn. An account on Cisco.com is not required.

Table 4: Feature Information for Call Transfer and Forwarding

Feature Name	Cisco Unified CME Version	Feature Information
Calling Number Local	12.0	Introduced support to configure Calling Number Local feature for Voice Register DNs.
Call Transfer Recall on SIP Phones	11.6	Call Transfer Recall feature returns a transferred call to the phone that initiated the transfer if the destination is busy or does not answer.
Trunk-to-Trunk Transfer Blocking for Toll Fraud Prevention on Cisco Unified SIP IP Phones	9.5	Introduced support Trunk-to-Trunk Transfer Blocking for Toll Fraud Prevention on Cisco Unified SIP IP Phones.

Feature Name	Cisco Unified CME Version	Feature Information
Call Forwarding	4.1	• Call Forward All synchronization between Cisco Unified CME and SIP phones was added. • Disabling SIP supplementary services for call forward and call transfer was added.
	4.0	• Automatic call forwarding during night service was introduced. • Selective call forwarding was introduced. • Forwarding of local (internal) calls can be blocked. • H.450.7 standards support and QSIG supplementary services capability was introduced.
	3.4	Calls into a SIP device can be forwarded to other SIP or SCCP devices including Cisco Unity, third- party voice mail systems, or an auto-attendant (AA) or other interactive voice response (IVR) devices. SCCP devices may also be forwarded to SIP devices.
	3.1	• Number of digits that can be entered using the CfwdALL (call-forward all) soft key can be limited. • H.450.12 standards support, which provide dynamic detection of H.450.2 and H.450.3 capabilities on a call-by-call basis, was introduced.
	3.0	

Feature Name	Cisco Unified CME Version	Feature Information
		• CFwdALL soft key was introduced. • Local hairpin call routing was supported as an option for networks that cannot support H.450 call transfer and forwarding. This feature requires installation of the Tcl script app_h450_transfer.2.0.0.8.tcl or a later version.
	2.1	Call forwarding using the H.450.3 standard was introduced.
	1.0	Call forwarding for all calls, busy conditions, and no-answer conditions was introduced, using a Cisco-proprietary method.
Call Forward Unregistered	8.6	The Call Forward Unregistered (CFU) feature was introduced for SIP phones.

Feature Name	Cisco Unified CME Version	Feature Information
Call Transfer	4.3	• Call-Transfer Recall was added. • Consultative Call Transfer digit-collection process was modified.
	4.1	• Disabling SIP supplementary services for call transfer and call forward was added.
	4.0	• Default for the **transfer-system** command was changed from the **blind** keyword to the **full-consult** keyword. • Transfers to phones outside the Cisco Unified CME system can be blocked for individual ephones. • Number of digits in transfer destination numbers can be limited.
	3.4	Support for attended and blind transfer s using SIP IP phone directly connected to Cisco CME.
	3.2	• Consultative transfer to monitored lines using direct station select was introduced. • Transcoding between G.711 and G.729 is supported when one leg of a Voice over IP (VoIP)-to-VoIP hairpin call uses G.711 and the other leg uses G.729.
	3.1	

Feature Name	Cisco Unified CME Version	Feature Information
		Support was introduced for the following: • Enhancements for VoIP networks which contain a mix of platforms that support H.450.2 and H.450.3 standards, such as Cisco CME 3.1, Cisco CME 3.0, Cisco ITS V2.1, and platforms that do not support H.450.2 and H.450.3 standards, such as Cisco Unified Communications Manager, Cisco BTS Softswitch (BTS), and Cisco PSTN Gateway (PGW). • H.450.12 standards, which provide dynamic detection of H.450.2 and H.450.3 capabilities on a call-by-call basis. • Automatic detection of Cisco Unified Communications Manager endpoints. • Hairpin VoIP-to-VoIP call routing and routing to an H.450 tandem gateway. • Hairpin call routing does not require a Tcl script.
	3.0	Local hairpin call routing was supported as an option for networks that cannot support H.450 call transfer and forwarding. This feature requires installation of the Tcl script app_h450_transfer.2.0.0.8.tcl or a later version.
	2.1	Consultative transfer using the ITU-T H.450.2 standard was introduced.
	1.0	Call transfer was introduced, using a Cisco proprietary method.