Configuring Call Transfer and Forwarding

Last Updated: March 15, 2013

This chapter describes call transfer and forwarding features in Cisco Unified Communications Manager Express (Cisco Unified CME) to enable interworking with various network requirements.

Finding Feature Information in This Module
Your Cisco Unified CME version may not support all of the features documented in this module. For a list of the versions in which each feature is supported, see the “Feature Information for Call Transfer and Forwarding” section on page 834.

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Information About Call Transfer and Forwarding

To configure transfer and forwarding features, you should understand the following concepts:

- Call Forwarding, page 756
- Call Forward Unregistered, page 757
- B2BUA Call Forwarding for SIP Devices, page 758
- Call Forward All Synchronization for SIP Phones, page 758
- Call Transfer, page 759
- Trunk-to-Trunk Transfer Blocking for Toll Fraud Prevention on Cisco Unified SIP IP Phones, page 760
- H.450.2 and H.450.3 Support, page 764
Call Forwarding

Call forwarding 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 the “SCCP: Enabling Call Forwarding for a Directory Number” section on page 784.

Selective Call Forwarding

You can apply call forwarding 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:
Configuring Call Transfer and Forwarding

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- 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 the “SCCP: Enabling Call Forwarding for a Directory Number” section on page 784.

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 the, “Configuring Keepalive Timer Expiration in SIP Phones: Example” section on page 831.

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 the “Configuring Call Forward Unregistered for SIP IP Phones: Example” section on page 831.

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 Forwarding for SIP Devices

Cisco Unified CME 3.4 and 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 the “SIP: Configuring SIP-to-SIP Phone Call Forwarding” section on page 812.

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 the “SIP: Configuring Call-Forwarding-All Soft Key URI” section on page 818.

### 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 interoperate 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 systemwide 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 the “SCCP: Configuring Call Transfer Options for Phones” section on page 788.

**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.

Table 66 lists the transfer-blocking commands and the appropriate configuration modes for Cisco Unified CME and Cisco Unified SRST.

<table>
<thead>
<tr>
<th>Table 66</th>
<th>Configuration Modes for Transfer-Blocking Commands</th>
</tr>
</thead>
<tbody>
<tr>
<td>Commands</td>
<td>Cisco Unified CME</td>
</tr>
<tr>
<td>transfer-pattern</td>
<td>telephony-service</td>
</tr>
<tr>
<td>transfer max-length</td>
<td>voice register pool or voice register template</td>
</tr>
<tr>
<td>transfer-pattern blocked</td>
<td>voice register pool or voice register template</td>
</tr>
<tr>
<td>conference transfer-pattern</td>
<td>telephony-service</td>
</tr>
</tbody>
</table>

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**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.

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**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].`

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**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.

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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.
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.

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.

### `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 67 compares the behaviors of Cisco Unified SCCP and SIP IP phones for specific configurations.

Table 67  Behaviors of Cisco Unified IP Phones for Specific Configurations

<table>
<thead>
<tr>
<th>Configuration</th>
<th>Cisco Unified SCCP IP Phones</th>
<th>Cisco Unified SIP IP Phones</th>
</tr>
</thead>
<tbody>
<tr>
<td>No transfer patterns are configured.</td>
<td>All non-local call transfers are blocked.</td>
<td>All non-local call transfers are allowed for backward compatibility.</td>
</tr>
<tr>
<td>Specific transfer patterns are configured.</td>
<td>Call transfers to specific external entities are allowed.</td>
<td>Call transfers to specific external entities are allowed.</td>
</tr>
<tr>
<td>The transfer-pattern blocked command is configured.</td>
<td>All non-local call transfers are blocked.</td>
<td>All non-local call transfers are blocked.</td>
</tr>
<tr>
<td>Note</td>
<td>The configuration reverts to the default, where no transfer patterns are configured.</td>
<td>Note</td>
</tr>
</tbody>
</table>

Conference Transfer-Pattern

When both the transfer-pattern and conference transfer-pattern commands are configured and 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 the “SIP: Specifying Transfer Patterns for Trunk-to-Trunk Calls and Conferences” section on page 792 and “SIP: Blocking Trunk-to-Trunk Call Transfers” section on page 794.

For configuration examples, see the “Configuring Transfer Patterns: Example” section on page 823, “Configuring Maximum Length of Transfer Number: Example” section on page 823, “Configuring Conference Transfer Patterns: Example” section on page 824, and “Blocking All Call Transfers: Example” section on page 824.

Call-Transfer Recall

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 xxx” displays on the transferor phone.

The transfer-to directory number cannot have Call Forward Busy enabled and cannot be a member of any hunt group. 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 a 15-second retry-timer 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 the “Enabling Call Transfer and Forwarding at System-Level” section on page 779.

**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 the “Enabling Call Transfer and Forwarding at System-Level” section on page 779.

**Consultative Transfer With Direct Station Select**

Direct Station Select (DSS) is a feature that allows a multibutton 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 “Configuring Conferencing” on page 941.

### 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 31 on page 765 shows A calling B. Figure 32 on page 765 shows B consulting with C and putting A on hold. Figure 33 on page 765 shows that B has connected A and C, and Figure 34 on page 766 shows A and C directly connected, with B no longer involved in the call.
Figure 31  Call Transfer Using H.450.2: A Calls B

Figure 32  Call Transfer Using H.450.2: B Consults with C

Figure 33  Call Transfer Using H.450.2: B Transfers A to C
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 the “Enabling Call Transfer and Forwarding at System-Level” section on page 779.
**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 68 summarizes transfer method recommendations for all Cisco Unified CME versions.

<table>
<thead>
<tr>
<th>Cisco Unified CME Version</th>
<th>transfer-system Command Default</th>
<th>transfer-system Keyword to Use</th>
<th>Transfer Method Recommendation</th>
</tr>
</thead>
<tbody>
<tr>
<td>4.0 and later</td>
<td><strong>full-consult</strong></td>
<td><strong>full-consult</strong> or <strong>full-blind</strong></td>
<td>Use H.450.2 for call transfer, which is the default for this version. You do not need to use the <strong>transfer-system</strong> command unless you want to use the <strong>full-blind</strong> or <strong>dss</strong> keyword. Optionally, you can use the proprietary Cisco method by using the <strong>transfer-system</strong> command with the <strong>blind</strong> or <strong>local-consult</strong> keyword. Use H.450.7 for call transfer using QSIG supplementary services.</td>
</tr>
<tr>
<td>3.0 to 3.3</td>
<td><strong>blind</strong></td>
<td><strong>full-consult</strong> or <strong>full-blind</strong></td>
<td>Use H.450.2 for call transfer. You must explicitly configure the <strong>transfer-system</strong> command with the <strong>full-consult</strong> or <strong>full-blind</strong> keyword because H.450.2 is not the default for this version. Optionally, you can use the proprietary Cisco method by using the <strong>transfer-system</strong> command with the <strong>blind</strong> or <strong>local-consult</strong> keyword.</td>
</tr>
<tr>
<td>2.1</td>
<td><strong>blind</strong></td>
<td><strong>blind</strong> or <strong>local-consult</strong></td>
<td>Use the Cisco proprietary method, which is the default for this version. You do not need to use the <strong>transfer-system</strong> command unless you want to use the <strong>local-consult</strong> keyword. Optionally, you can use the <strong>transfer-system</strong> command with the <strong>full-consult</strong> or <strong>full-blind</strong> 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 <a href="http://www.cisco.com/cgi-bin/tablebuild.pl/ip-iostsp">http://www.cisco.com/cgi-bin/tablebuild.pl/ip-iostsp</a>.</td>
</tr>
<tr>
<td>Earlier than 2.1</td>
<td><strong>blind</strong></td>
<td><strong>blind</strong></td>
<td>Use the Cisco proprietary method, which is the default for this version. You do not need to use the <strong>transfer-system</strong> command unless you want to use the <strong>local-consult</strong> keyword.</td>
</tr>
</tbody>
</table>
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 the “Enabling H.450.12 Capabilities” section on page 795.
  - 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 “Enabling H.323-to-H.323 Connection Capabilities” section on page 797. The 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-VoIIP, 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. For information about transcoding, see “Configuring Transcoding Resources” on page 443.

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 35 on page 769 shows a call that is made from A to B. Figure 36 on page 770 shows that B has forwarded all calls to C. Figure 37 on page 770 shows that A and C are connected by an H.323 hairpin.

*Figure 35 Hairpin with H.323: A Calls B*
Figure 36 Hairpin with H.323: Call is Forwarded to C

Figure 37 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 the “Enabling H.450.12 Capabilities” section on page 795.
- 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 the “Forwarding Calls Using Local Hairpin Routing” section on page 799.
Support for VoIP-to-VoIP connections is disabled by default and can be enabled globally. For configuration information, see the “Enabling H.323-to-H.323 Connection Capabilities” section on page 797.

**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.

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**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 38 on page 772** 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 38 on page 772, 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.
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 the “Enabling H.323-to-H.323 Connection Capabilities” section on page 797.

Use dial peers to set up an H.450 tandem gateway. See the “Dial Peers” section on page 773.

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. For information about configuring dial peers, see the Dial Peer Configuration on Voice Gateway Routers guide.

QSIG Supplementary Services

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 39 illustrates a topology for a Cisco Unified CME system with some phones under the control of a PBX.

Figure 39 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 the “Enabling H.450.7 and QSIG Supplementary Services at a System-Level” section on page 801 and “Enabling H.450.7 and QSIG Supplementary Services on a Dial Peer” section on page 802.

For more information about configuring Cisco Unified CME to integrate with voice-mail systems, see “Integrating Voice Mail” on page 531.

### Disabling 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 the “Disabling SIP Supplementary Services for Call Forward and Call Transfer” section on page 804.
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.

- Cisco CME 3.1 or Later and Cisco IOS Gateways, page 775
- Cisco CME 3.0 or an Earlier Version and Cisco IOS Gateways, page 776
- Cisco CME 3.1 or Later, Non-H.450 Gateways, and Cisco IOS Gateways, page 776
- Cisco Unified CME, Non-H.450 Gateways, and Cisco IOS Gateways, page 777
- Cisco CME 3.1 or Later, Cisco Unified Communications Manager, and Cisco IOS Gateways, page 777
- Cisco CME 3.0 or an Earlier Version, Cisco Unified Communications Manager, and Cisco IOS Gateways, page 778

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 the “Enabling Call Transfer and Forwarding at System-Level” section on page 779.
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 the “Enabling H.450.12 Capabilities” section on page 795.
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 the “Enabling H.323-to-H.323 Connection Capabilities” section on page 797.
4. Setting up dial peers to manage call legs within the network. See Dial Peer Configuration on Voice Gateway Routers.
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:

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 the “Enabling Call Transfer and Forwarding at System-Level” section on page 779.

2. 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 the “Enabling H.450.12 Capabilities” section on page 795.

3. 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 the “Enabling H.323-to-H.323 Connection Capabilities” section on page 797.

4. Setting up dial peers to manage call legs within the network. See Dial Peer Configuration on Voice Gateway Routers.

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 the “Enabling Call Transfer and Forwarding at System-Level” section on page 779.

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 the “Enabling H.450.12 Capabilities” section on page 795.

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 the “Enabling H.323-to-H.323 Connection Capabilities” section on page 797.

4. Setting up dial peers to manage call legs within the network. See Dial Peer Configuration on Voice Gateway Routers.

Note: If your network contains a Cisco Unified Communications Manager, also see the instructions in the “Enabling Interworking with Cisco Unified Communications Manager” section on page 806.
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 the “Enabling Call Transfer and Forwarding at System-Level” section on page 779.
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 the “Enabling H.450.12 Capabilities” section on page 795.
3. Setting up VoIP-to-VoIP connections (hairpin call routing or H.450 tandem gateway) to route all transferred and forwarded calls. See the “Enabling H.323-to-H.323 Connection Capabilities” section on page 797.
4. Setting up dial peers to manage call legs within the network. See Dial Peer Configuration on Voice Gateway Routers.

Note
If your network contains a Cisco Unified Communications Manager, also see the instructions in the “Enabling Interworking with Cisco Unified Communications Manager” section on page 806.

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 the “Enabling Call Transfer and Forwarding at System-Level” section on page 779.
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 the “Enabling H.450.12 Capabilities” section on page 795.

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. See the “Enabling H.323-to-H.323 Connection Capabilities” section on page 797.

4. Setting up specific parameters for Cisco Unified Communications Manager. See the instructions in the “Enabling Interworking with Cisco Unified Communications Manager” section on page 806.

5. Setting up dial peers to manage call legs within the network. See Dial Peer Configuration on Voice Gateway Routers.

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. See “Configuring Loopback Call Routing” on page 1185.
- 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).

How to Configure Call Transfer and Forwarding

This section contains the following procedures:

SCCP
- Enabling Call Transfer and Forwarding at System-Level, page 779 (required)
- SCCP: Enabling Call Forwarding for a Directory Number, page 784 (required)
- SCCP: Enabling Call Transfer for a Directory Number, page 787 (required)
- SCCP: Configuring Call Transfer Options for Phones, page 788 (optional))
- SCCP: Verifying Call Transfer, page 791 (optional)
- SIP: Specifying Transfer Patterns for Trunk-to-Trunk Calls and Conferences, page 792
- SIP: Blocking Trunk-to-Trunk Call Transfers, page 794
- Enabling H.450.12 Capabilities, page 795 (optional)
Enabling Call Transfer and Forwarding 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.

**Prerequisites**

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

**Restrictions**

- Call transfers are handled differently depending on the Cisco Unified CME version. See Table 68 on page 767 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; transeree 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 is a member of any hunt group.

• 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.

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

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1 enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td></td>
<td>• Enter your password if prompted.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router&gt; enable</td>
<td></td>
</tr>
<tr>
<td>Step 2 configure terminal</td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router# configure terminal</td>
<td></td>
</tr>
</tbody>
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### Configuring Call Transfer and Forwarding

#### How to Configure Call Transfer and Forwarding

<table>
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<th>Command or Action</th>
<th>Purpose</th>
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<tbody>
<tr>
<td><strong>Step 3</strong></td>
<td>telephony-service</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Router(config)# telephony-service</td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td>transfer-system {blind</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Router(config-telephony)# transfer-system full-consult</td>
</tr>
</tbody>
</table>

- **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 soft-key "Transfer" and the call will be transferred to the number selected and then you can hang up.

- **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.)
## Configuring Call Transfer and Forwarding

- **Step 5**
  
  **Command or Action**
  
  `transfer-pattern transfer-pattern [blind]`
  
  **Example:**
  
  Router(config-telephony)# transfer-pattern .T
  
  **Purpose**
  
  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. For more information, see “Configuring Dialing Plans” on page 375.

- **Step 6**
  
  **Command or Action**
  
  `call-forward pattern pattern`
  
  **Example:**
  
  Router(config-telephony)# call-forward pattern .T
  
  **Purpose**
  
  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**
  
  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. For more information, see “Configuring Dialing Plans” on page 375.

- **Step 7**
  
  **Command or Action**
  
  `timeouts transfer-recall seconds`
  
  **Example:**
  
  Router(config-telephony)# timeouts transfer-recall 30
  
  **Purpose**
  
  (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.
### How to Configure Call Transfer and Forwarding

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
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</thead>
<tbody>
<tr>
<td><strong>Step 8</strong> transfer-digit-collect *(new-call</td>
<td>orig-call)*</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(config-telephony)# transfer-digit-collect orig-call</td>
<td></td>
</tr>
<tr>
<td><strong>Step 9</strong> exit</td>
<td>Exits telephony-service configuration mode.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(config-telephony)# exit</td>
<td></td>
</tr>
<tr>
<td><strong>Step 10</strong> voice service voip</td>
<td>(Optional) Enters voice-service configuration mode to establish global call transfer and forwarding parameters.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(config)# voice service voip</td>
<td></td>
</tr>
<tr>
<td><strong>Step 11</strong> supplementary-service h450.2</td>
<td>(Optional) Enables H.450.2 supplementary services capabilities globally.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(conf-voi-serv)# supplementary-service h450.2</td>
<td></td>
</tr>
<tr>
<td><strong>Step 12</strong> supplementary-service h450.3</td>
<td>(Optional) Enables H.450.3 supplementary services capabilities globally.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(conf-voi-serv)# supplementary-service h450.3</td>
<td></td>
</tr>
<tr>
<td><strong>Step 13</strong> exit</td>
<td>(Optional) Exits voice-service configuration mode.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(conf-voi-serv)# exit</td>
<td></td>
</tr>
<tr>
<td><strong>Step 14</strong> dial-peer voice tag voip</td>
<td>(Optional) Enters dial-peer configuration mode.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(config)# dial-peer voice 1 voip</td>
<td></td>
</tr>
</tbody>
</table>
SCCP: Enabling 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. For more information, see the “Voice Translation Rules and Profiles” section in “Configuring Dialing Plans” on page 375.
Restrictions

- 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

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td>enable</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Router&gt; enable</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td>configure terminal</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Router# configure terminal</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td>telephony-service</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Router(config)#</td>
</tr>
</tbody>
</table>
### Command or Action

**Step 4**

`call-forward pattern pattern`

**Example:**

Router(config-telephony)# call-forward pattern .T

**Purpose**

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 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

**Purpose**

Exits telephony-service configuration mode.

**Step 6**

`ephone-dn dn-tag [dual-line]`

**Example:**

Router(config)# ephone-dn 20

**Purpose**

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

**Purpose**

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

**Purpose**

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

**Purpose**

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

**Purpose**

Forwards calls for an extension that does not answer.
SCCP: Enabling 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.

Prerequisites

Call transfer must be enabled globally. See the “Configuring Call Coverage Features” section on page 837.

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

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong></td>
<td><strong>enable</strong>&lt;br&gt;Enables privileged EXEC mode.&lt;br&gt;• Enter your password if prompted.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Router&gt; enable</td>
</tr>
<tr>
<td><strong>Step 2</strong></td>
<td><strong>configure terminal</strong>&lt;br&gt;Enters global configuration mode.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Router# configure terminal</td>
</tr>
<tr>
<td><strong>Step 3</strong></td>
<td><strong>ephone-dn dn-tag [dual-line]</strong>&lt;br&gt;Enters ephone-dn configuration mode, creates an ephone-dn, and optionally assigns it dual-line status.&lt;br&gt;• 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.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Router(config)# ephone-dn 20</td>
</tr>
<tr>
<td><strong>Step 4</strong></td>
<td>**transfer-mode {blind</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Router(config-ephone-dn)# transfer-mode blind</td>
</tr>
<tr>
<td><strong>Step 5</strong></td>
<td><strong>timeouts transfer-recall seconds</strong>&lt;br&gt;(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.&lt;br&gt;• seconds—Duration, in seconds, to wait before recalling a transferred call. Range: 1 to 1800. Default: 0 (disabled).&lt;br&gt;• This command is supported in Cisco Unified CME 4.3 and later versions.&lt;br&gt;• This command can also be configured in ephone-dn-template and telephony-service configuration mode.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Router(config-ephone-dn)# timeouts transfer-recall 30</td>
</tr>
<tr>
<td><strong>Step 6</strong></td>
<td><strong>end</strong>&lt;br&gt;Returns to privileged EXEC mode.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Router(config-ephone-dn)# end</td>
</tr>
</tbody>
</table>

### SCCP: Configuring Call Transfer Options for Phones

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

- 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

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> <code>enable</code></td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td>Example: <code>enable</code></td>
<td>Enter your password if prompted.</td>
</tr>
<tr>
<td><strong>Step 2</strong> <code>configure terminal</code></td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td>Example: <code>configure terminal</code></td>
<td></td>
</tr>
<tr>
<td><strong>Step 3</strong> <code>ephone-template template-tag</code></td>
<td>Enters ephone-template configuration mode.</td>
</tr>
<tr>
<td>Example: <code>configure terminal</code></td>
<td></td>
</tr>
<tr>
<td><strong>Step 4</strong> <code>transfer-pattern blocked</code></td>
<td>(Optional) Prevents directory numbers on the phone to which this template is applied from transferring calls to patterns specified in the <code>transfer-pattern</code> (telephony-service) command.</td>
</tr>
<tr>
<td>Example: <code>configure terminal</code></td>
<td></td>
</tr>
</tbody>
</table>

**Note** This command is also available in ephone configuration mode to block external transfers from individual phones without using a template.
<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 5</strong></td>
<td><strong>transfer max-length digit-length</strong></td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Router(config-ephone-template)# transfer max-length 8</td>
</tr>
<tr>
<td><strong>Step 6</strong></td>
<td><strong>exit</strong></td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Router(config-ephone-template)# exit</td>
</tr>
<tr>
<td><strong>Step 7</strong></td>
<td><strong>ephone phone-tag</strong></td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Router(config)# ephone 25</td>
</tr>
<tr>
<td><strong>Step 8</strong></td>
<td><strong>ephone-template template-tag</strong></td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Router(config-ephone)# ephone-template 1</td>
</tr>
<tr>
<td><strong>Step 9</strong></td>
<td><strong>restart</strong></td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Router(config-ephone)# restart</td>
</tr>
<tr>
<td><strong>Step 10</strong></td>
<td><strong>end</strong></td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Router(config-ephone)# end</td>
</tr>
</tbody>
</table>
SCCP: Verifying Call Transfer

**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.

```
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.

```
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.
SIP: Specifying Transfer Patterns for Trunk-to-Trunk Calls and Conferences

To specify transfer patterns that will enable trunk-to-trunk calls and conferences, perform the following steps.

Prerequisites

Cisco Unified CME 9.5 or a later version.

Restrictions

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.

SUMMARY STEPS

1. enable
2. configure terminal
3. telephony-service
4. transfer-pattern transfer-pattern
5. exit
6. voice register pool pool-tag
   or
   voice register template template-tag
7. transfer max-length max-length
8. exit
9. telephony-service
10. conference transfer-pattern
11. restart
12. end
## DETAILED STEPS

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
</table>
| 1     | enable            | Enables privileged EXEC mode.  
  - Enter your password if prompted. |
|       |                   |         |
| 2     | configure terminal| Enters global configuration mode. |
|       |                   |         |
| 3     | telephony-service | Enters telephony-service configuration mode for configuring Cisco Unified CME. |
|       |                   |         |
| 4     | transfer-pattern  | 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. |
|       | transfer-pattern  |         |
|       |                   |         |
| 5     | exit              | Exits telephony-service configuration mode and enters global configuration mode. |
|       |                   |         |
| 6     | voice register pool pool-tag | 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  
  - voice register template template-tag | 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. |
|       | voice register pool pool-tag |         |
|       | or voice register template template-tag |         |
|       |                   |         |
| 7     | transfer max-length max-length | (Optional) Specifies the maximum length of the transfer number.  
  - max-length—Maximum length of the transfer number. Range is 3 to 16. |
**SIP: Blocking Trunk-to-Trunk Call Transfers**

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

**Prerequisites**

Cisco Unified CME 9.5 or a later version.

**Restrictions**

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.

**SUMMARY STEPS**

1. `enable`
2. `configure terminal`
3. `voice register pool pool-tag`
   or
   `voice register template template-tag`
4. `transfer-pattern blocked`
5. `end`

---

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 8 <code>exit</code></td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td><code>Router(config-register-pool)# exit</code></td>
<td></td>
</tr>
<tr>
<td>Step 9 <code>telephony-service</code></td>
<td>Enters telephony-service configuration mode for configuring Cisco Unified CME.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td><code>Router(config)# telephony-service</code></td>
<td></td>
</tr>
<tr>
<td>Step 10 <code>conference transfer-pattern</code></td>
<td>Enables a Cisco Unified CME system to apply transfer patterns to a conference call using conference softkeys or feature buttons.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td><code>Router(config-telephony)# conference transfer-pattern</code></td>
<td></td>
</tr>
<tr>
<td>Step 11 <code>end</code></td>
<td>Exits telephony-service configuration mode and enters privileged EXEC mode.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td><code>Router(config-telephony)# end</code></td>
<td></td>
</tr>
</tbody>
</table>
## DETAILED STEPS

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td><strong>Example:</strong> Router&gt; enable</td>
<td>• Enter your password if prompted.</td>
</tr>
<tr>
<td><strong>Step 2</strong> configure terminal</td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td><strong>Example:</strong> Router# configure terminal</td>
<td></td>
</tr>
<tr>
<td><strong>Step 3</strong> voice register pool pool-tag or voice register template template-tag</td>
<td>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.</td>
</tr>
<tr>
<td><strong>Example:</strong> Router(config)# voice register template 5</td>
<td>• 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.</td>
</tr>
<tr>
<td></td>
<td>• template-tag—Declarations a template tag. Range is 1 to 10.</td>
</tr>
<tr>
<td><strong>Step 4</strong> transfer-pattern blocked</td>
<td>Blocks all call transfers for a specific Cisco Unified SIP IP phone or a set of Cisco Unified SIP IP phone.</td>
</tr>
<tr>
<td><strong>Example:</strong> Router(config-register-temp)# transfer-pattern blocked</td>
<td></td>
</tr>
<tr>
<td><strong>Step 5</strong> end</td>
<td>Exits voice register template configuration mode and enters privileged EXEC mode.</td>
</tr>
<tr>
<td><strong>Example:</strong> Router(config-register-temp)# end</td>
<td></td>
</tr>
</tbody>
</table>

## Enabling 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.

## Restrictions

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

## SUMMARY STEPS

1. enable
### DETAILED STEPS

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1 enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router&gt; enable</td>
<td></td>
</tr>
<tr>
<td>Step 2 configure terminal</td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router# configure terminal</td>
<td></td>
</tr>
<tr>
<td>Step 3 voice service voip</td>
<td>(Optional) Enters voice service configuration mode to establish global call transfer and forwarding parameters.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(config)# voice service voip</td>
<td></td>
</tr>
<tr>
<td>Step 4 supplementary-service h450.12 [advertise-only]</td>
<td>(Optional) Enables H.450.12 supplementary services capabilities globally for VoIP endpoints.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(conf-voi-serv)# supplementary-service h450.12</td>
<td></td>
</tr>
<tr>
<td>Step 5 exit</td>
<td>(Optional) Exits voice-service configuration mode.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(conf-voi-serv)# exit</td>
<td></td>
</tr>
<tr>
<td>Step 6 dial-peer voice tag voip</td>
<td>(Optional) Enters dial-peer configuration mode.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(config)# dial-peer 1 voip</td>
<td></td>
</tr>
</tbody>
</table>
Enabling H.323-to-H.323 Connection Capabilities

VoIP-to-VoIP connections 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 the “Enabling H.450.12 Capabilities” section on page 795.
- Explicitly disable H.450.2 and H.450.3 capabilities globally or on the routes that your transfers and forwards take. See the “Enabling Call Transfer and Forwarding at System-Level” section on page 779.

Restrictions

- 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**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1 enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td><strong>Example:</strong> Router&gt; enable</td>
<td>• Enter your password if prompted.</td>
</tr>
<tr>
<td>Step 2 configure terminal</td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td><strong>Example:</strong> Router# configure terminal</td>
<td></td>
</tr>
<tr>
<td>Step 3 voice service voip</td>
<td>Enters voice service configuration mode to establish global call transfer and forwarding parameters.</td>
</tr>
<tr>
<td><strong>Example:</strong> Router(config)# voice service voip</td>
<td></td>
</tr>
<tr>
<td>Step 4 allow-connections h323 to h323</td>
<td>Enables VoIP-to-VoIP call connections. Use the no form of the command to disable VoIP-to-VoIP connections; this is the default.</td>
</tr>
<tr>
<td><strong>Example:</strong> Router(conf-voi-serv)# allow-connections h323 to h323</td>
<td></td>
</tr>
<tr>
<td>Step 5 end</td>
<td>Returns to privileged EXEC mode.</td>
</tr>
<tr>
<td><strong>Example:</strong> Router(config-voi-serv)# end</td>
<td></td>
</tr>
</tbody>
</table>
Forwarding 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

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td><strong>Example:</strong> Router&gt; enable</td>
<td>• Enter your password if prompted.</td>
</tr>
<tr>
<td><strong>Step 2</strong> configure terminal</td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td><strong>Example:</strong> Router# configure terminal</td>
<td></td>
</tr>
<tr>
<td><strong>Step 3</strong> telephony-service</td>
<td>Enters telephony-service configuration mode.</td>
</tr>
<tr>
<td><strong>Example:</strong> Router(config)# telephony-service</td>
<td></td>
</tr>
</tbody>
</table>
### Command or Action

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>4</td>
<td><code>call-forward pattern pattern</code></td>
<td>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.</td>
</tr>
<tr>
<td></td>
<td>Example: <code>Router(config-telephony)# call-forward pattern 6000</code></td>
<td></td>
</tr>
<tr>
<td>5</td>
<td><code>calling-number local</code></td>
<td>(Optional) Replaces a calling-party number and name with the forwarding-party (local) number and name for hairpin-forwarded calls only.</td>
</tr>
<tr>
<td></td>
<td>Example: <code>Router(config-telephony)# calling-number local</code></td>
<td>- Before Cisco CME 3.3, this command must be used with Tool Command Language (Tcl) script <code>app-h450-transfer.2.0.0.7</code> or a later version. The local-hairpin attribute-value (AV) pair must be set to 1.</td>
</tr>
<tr>
<td>6</td>
<td><code>exit</code></td>
<td>Exits telephony-service configuration mode.</td>
</tr>
<tr>
<td></td>
<td>Example: <code>Router(config-telephony)# exit</code></td>
<td></td>
</tr>
<tr>
<td>7</td>
<td><code>voice service voip</code></td>
<td>Enters voice-service configuration mode.</td>
</tr>
<tr>
<td></td>
<td>Example: <code>Router(config)# voice service voip</code></td>
<td></td>
</tr>
<tr>
<td>8</td>
<td><code>allow connections from-type to to-type</code></td>
<td>Allows connections between specific types of endpoints in a network.</td>
</tr>
<tr>
<td></td>
<td>Example: <code>Router(conf-voi-serv)# allow connections h323 to sip</code></td>
<td>- <code>from-type</code>—Originating endpoint type. Valid choices are <code>h323</code> and <code>sip</code>.</td>
</tr>
<tr>
<td></td>
<td></td>
<td>- <code>to-type</code>—Terminating endpoint type. Valid choices are <code>h323</code> and <code>sip</code>.</td>
</tr>
<tr>
<td>9</td>
<td><code>supplementary-service h450.3</code></td>
<td>(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.</td>
</tr>
<tr>
<td></td>
<td>Example: <code>Router(conf-voi-serv)# no supplementary-service h450.3</code></td>
<td><strong>Note</strong> If this command is disabled globally and either enabled or disabled on a dial peer, the functionality is disabled for the dial peer.</td>
</tr>
<tr>
<td>10</td>
<td><code>end</code></td>
<td>Exits to privileged EXEC mode.</td>
</tr>
<tr>
<td></td>
<td>Example: <code>Router(config-voi-serv)# end</code></td>
<td></td>
</tr>
</tbody>
</table>
Enabling H.450.7 and QSIG Supplementary Services at a System-Level

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

Prerequisites

Cisco Unified CME 4.0 or a later version.

Restrictions

- 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.

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

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td></td>
</tr>
<tr>
<td>enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router&gt; enable</td>
<td></td>
</tr>
<tr>
<td>Step 2</td>
<td></td>
</tr>
<tr>
<td>configure terminal</td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router# configure terminal</td>
<td></td>
</tr>
</tbody>
</table>

Example:

Router> enable

Router# configure terminal
### Enabling 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.

#### Prerequisites

Cisco Unified CME 4.0 or a later version.

#### Restrictions

- QSIG integration supports SCCP phones only.

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 3</strong> voice service voip</td>
<td>Enters VoIP voice-service configuration mode to define global call transfer and forwarding parameters.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(config)# voice service voip</td>
<td></td>
</tr>
<tr>
<td><strong>Step 4</strong> supplementary-service h450.7</td>
<td>Enables H.450.7 supplementary services capabilities exchange at a system-level.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(config-voi-serv)# supplementary-service h450.7</td>
<td></td>
</tr>
<tr>
<td><strong>Step 5</strong> qsig decode</td>
<td>Enables decoding for QSIG supplementary services.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(config-voi-serv)# qsig decode</td>
<td></td>
</tr>
<tr>
<td><strong>Step 6</strong> exit</td>
<td>Exits VoIP voice-service configuration mode.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(config-voi-serv)# exit</td>
<td></td>
</tr>
<tr>
<td><strong>Step 7</strong> voice service pots</td>
<td>Enters POTS voice-service configuration mode to define global call transfer and forwarding parameters.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(config)# voice service pots</td>
<td></td>
</tr>
<tr>
<td><strong>Step 8</strong> supplementary-service qsig call-forward</td>
<td>Enables QSIG call-forwarding supplementary services (ISO 13873) to forward calls to another number.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(config-voi-serv)# supplementary-service qsig call-forward</td>
<td></td>
</tr>
<tr>
<td><strong>Step 9</strong> end</td>
<td>Exits to privileged EXEC mode.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(config-voi-serv)# end</td>
<td></td>
</tr>
</tbody>
</table>
- 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.

**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**

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
</table>
| Step 1 | enable | Enables privileged EXEC mode.  
- Enter your password if prompted. |
| Example: | Router> enable | |
| Step 2 | configure terminal | Enters global configuration mode. |
| Example: | Router# configure terminal | |
| Step 3 | voice service voip | Enters VoIP voice-service configuration mode to define global call transfer and forwarding parameters. |
| Example: | Router(config)# voice service voip | |
| Step 4 | qsig decode | Enables decoding for QSIG supplementary services. |
| Example: | Router(config-voi-serv)# qsig decode | |
| Step 5 | exit | Exits VoIP voice-service configuration mode. |
| Example: | Router(config-voi-serv)# exit | |
Disabling 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.

Prerequisites

Cisco Unified CME 4.1 or a later version.

Restrictions

- 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.
### SUMMARY STEPS

1. enable
2. configure terminal
3. voice service voip
   or
dial-peer voice tag voip
4. no supplementary-service sip moved-temporarily
5. no supplementary-service sip refer
6. end

### DETAILED STEPS

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td></td>
<td>• Enter your password if prompted.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Router&gt; enable</td>
</tr>
<tr>
<td><strong>Step 2</strong> configure terminal</td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Router# configure terminal</td>
</tr>
<tr>
<td><strong>Step 3</strong> voice service voip</td>
<td>Enters voice-service configuration mode to set global parameters for VoIP features.</td>
</tr>
<tr>
<td>or dial-peer voice tag voip</td>
<td>or</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Router(config)# voice service voip</td>
</tr>
<tr>
<td></td>
<td>Router(config)# dial-peer voice 99 voip</td>
</tr>
<tr>
<td><strong>Step 4</strong> no supplementary-service sip moved-temporarily</td>
<td>Disables SIP redirect response for call forwarding either globally or for a dial peer.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Router(conf-voi-serv)# no supplementary-service sip moved-temporarily</td>
</tr>
<tr>
<td></td>
<td>Router(config-dial-peer)# no supplementary-service sip moved-temporarily</td>
</tr>
<tr>
<td></td>
<td>• Sending redirect message to the destination is the default behavior.</td>
</tr>
</tbody>
</table>
Enabling 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:

- Configuring Cisco CME 3.1 or Later to Interwork with Cisco Unified Communications Manager, page 807
- Enabling Cisco Unified Communications Manager to Interwork with Cisco Unified CME, page 810
- Troubleshooting Transfer and Forwarding Configuration, page 811

Figure 40 shows a network containing Cisco Unified CME and Cisco Unified Communications Manager systems.
Prerequisites

- Cisco Unified CME must be configured to forward calls using local hairpin routing. For configuration information, see the “Forwarding Calls Using Local Hairpin Routing” section on page 799.

Configuring 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 nondefault 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

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>• Enter your password if prompted.</td>
</tr>
<tr>
<td>Router&gt; enable</td>
<td></td>
</tr>
<tr>
<td><strong>Step 2</strong> configure terminal</td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router# configure terminal</td>
<td></td>
</tr>
<tr>
<td><strong>Step 3</strong> voice service voip</td>
<td>Enters voice-service configuration mode to establish global parameters.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(config)# voice service voip</td>
<td></td>
</tr>
<tr>
<td><strong>Step 4</strong> h323</td>
<td>Enters H.323 voice-service configuration mode.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(conf-voi-serv)# h323</td>
<td></td>
</tr>
<tr>
<td><strong>Step 5</strong> telephony-service ccm-compatible</td>
<td>(Optional) Globally enables a Cisco CME 3.1 or later system to detect</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Cisco Unified Communications Manager and exchange calls with it. This</td>
</tr>
<tr>
<td>Router(conf-serv-h323)# telephony-service ccm-compatible</td>
<td>is the default.</td>
</tr>
<tr>
<td></td>
<td>• Use the no form of this command to disable Cisco Unified Communications</td>
</tr>
<tr>
<td></td>
<td>Manager detection and exchange. We do not recommend using the no form</td>
</tr>
<tr>
<td></td>
<td>of the command.</td>
</tr>
<tr>
<td></td>
<td>• Using this command in an H.323 voice class definition allows you to</td>
</tr>
<tr>
<td></td>
<td>specify this behavior for an individual dial peer.</td>
</tr>
</tbody>
</table>
### Command or Action

<table>
<thead>
<tr>
<th>Step 6</th>
<th>h225 h245-address on-connect</th>
</tr>
</thead>
</table>

**Example:**

```bash
Router(conf-serv-h323)# h225 h245-address on-connect
```

(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 Cisco Unified Communications Manager to generate local ringback for calls to Cisco Unified CME phones. This is the default.

- 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.

<table>
<thead>
<tr>
<th>Step 7</th>
<th>exit</th>
</tr>
</thead>
</table>

**Example:**

```bash
Router(conf-serv-h323)# exit
```

Exits H.323 voice-service configuration mode.

<table>
<thead>
<tr>
<th>Step 8</th>
<th>supplementary-service h225-notify cid-update</th>
</tr>
</thead>
</table>

**Example:**

```bash
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.

- 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.

<table>
<thead>
<tr>
<th>Step 9</th>
<th>exit</th>
</tr>
</thead>
</table>

**Example:**

```bash
Router(config-voice-service)# exit
```

Exits voice-service configuration mode.

<table>
<thead>
<tr>
<th>Step 10</th>
<th>voice class h323 tag</th>
</tr>
</thead>
</table>

**Example:**

```bash
Router(config)# voice class h323 48
```

(Optional) Creates a voice class that contains commands to be applied to one or more dial peers.

<table>
<thead>
<tr>
<th>Step 11</th>
<th>telephony-service ccm-compatible</th>
</tr>
</thead>
</table>

**Example:**

```bash
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.

- 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.
### Command or Action

<table>
<thead>
<tr>
<th>Step 12</th>
<th>h225 h245-address on-connect</th>
<th><strong>Purpose</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Example:</strong></td>
<td>Router(config-voice-class)# h225 h245-address on-connect</td>
<td>(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.</td>
</tr>
<tr>
<td>-</td>
<td>-</td>
<td>• The <strong>no</strong> form of this command disables the delay. We do not recommend using the <strong>no</strong> form of the command.</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Step 13</th>
<th>exit</th>
<th><strong>Purpose</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Example:</strong></td>
<td>Router(config-voice-class)# exit</td>
<td>Exits voice-class configuration mode.</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Step 14</th>
<th>dial-peer voice tag voip</th>
<th><strong>Purpose</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Example:</strong></td>
<td>Router(config)# dial-peer voice 28 voip</td>
<td>(Optional) Enters dial-peer configuration mode to set parameters for an individual dial peer.</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Step 15</th>
<th>supplementary-service h225-notify cid-update</th>
<th><strong>Purpose</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Example:</strong></td>
<td>Router(config-dial-peer)# no supplementary-service h225-notify cid-update</td>
<td>(Optional) Enables H.225 messages with caller-ID updates to Cisco Unified Communications Manager for a specific dial peer. This is the default.</td>
</tr>
<tr>
<td>-</td>
<td>-</td>
<td>• The <strong>no</strong> form of the command disables caller-ID updates. We do not recommend using the <strong>no</strong> form of the command.</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Step 16</th>
<th>voice-class h323 tag</th>
<th><strong>Purpose</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Example:</strong></td>
<td>Router(config-dial-peer)# voice-class h323 48</td>
<td>(Optional) Applies the previously defined voice class with the specified tag number to this dial peer.</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Step 17</th>
<th>end</th>
<th><strong>Purpose</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Example:</strong></td>
<td>Router(config-dial-peer)# end</td>
<td>Exits to privileged EXEC mode.</td>
</tr>
</tbody>
</table>

### What to Do Next

Set up Cisco Unified Communications Manager using the configuration procedure in the “Enabling Cisco Unified Communications Manager to Interwork with Cisco Unified CME” section on page 810.

### Enabling 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.

### SUMMARY STEPS

1. Set Cisco Unified Communications Manager service parameters.
2. Configure Cisco Unified CME as an ICT in the Cisco Unified Communications Manager network.

3. Ensure that the Cisco Unified Communications Manager network uses an MTP.

4. Set up dial peers to establish routing.

**DETAILED STEPS**

**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 the 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 the 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 Transfer and Forwarding Configuration**

**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

SIP: Configuring 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.

Prerequisites

- 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 the “Enabling Calls in Your VoIP Network” on page 88.

Restrictions

- 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.
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 noan *directory-number timeout* *seconds*
8. call-forward b2bua unreachable *directory-number*
9. end

DETAILED STEPS

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1 enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td>Example:</td>
<td>• Enter your password if prompted.</td>
</tr>
<tr>
<td>Step 2 configure terminal</td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Step 3 voice register dn <em>dn-tag</em></td>
<td>Enters voice register dn mode to define a directory number for a SIP phone, intercom line, voice port, or an MWI.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Step 4 call-forward b2bua all <em>directory-number</em></td>
<td>Enables call forwarding for a SIP back-to-back user agent so that all incoming calls will be forwarded to the designated directory-number.</td>
</tr>
<tr>
<td>Example:</td>
<td>• 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.</td>
</tr>
<tr>
<td></td>
<td>• If the <strong>call-forward b2bua all</strong> command is configured in voice register pool configuration mode, it applies to all directory numbers on the phone.</td>
</tr>
</tbody>
</table>
### Command or Action

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 5</td>
<td><code>call-forward b2bua busy directory-number</code></td>
<td>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.</td>
</tr>
<tr>
<td></td>
<td>Example: Router(config-register-dn)# call-forward b2bua busy 5006</td>
<td>• 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.</td>
</tr>
<tr>
<td>Step 6</td>
<td><code>call-forward b2bua mailbox directory-number</code></td>
<td>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.</td>
</tr>
<tr>
<td></td>
<td>Example: Router(config-register-dn)# call-forward b2bua mailbox 5007</td>
<td>• 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.</td>
</tr>
<tr>
<td>Step 7</td>
<td><code>call-forward b2bua noan directory-number timeout seconds</code></td>
<td>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.</td>
</tr>
<tr>
<td></td>
<td>Example: Router(config-register-dn)# call-forward b2bua noan 5010 timeout 10 or Router(config-register-pool)# call-forward b2bua noan 5010 timeout 10</td>
<td>• 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.</td>
</tr>
<tr>
<td></td>
<td></td>
<td>• <code>timeout seconds</code>—Duration that a call can ring before it is forwarded to the destination directory number. Range: 3 to 60000. Default: 20.</td>
</tr>
<tr>
<td>Step 8</td>
<td><code>call-forward b2bua unreachable directory-number</code></td>
<td>(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.</td>
</tr>
<tr>
<td></td>
<td>Example: Router(config-register-dn)# call-forward b2bua unreachable 5009 or Router(config-register-pool)# call-forward b2bua unreachable 5009</td>
<td>• Target directory-number must be configured in Cisco Unified CME.</td>
</tr>
<tr>
<td></td>
<td></td>
<td>• 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.</td>
</tr>
<tr>
<td></td>
<td></td>
<td>• This command was removed in Cisco Unified CME 4.1.</td>
</tr>
<tr>
<td>Step 9</td>
<td><code>end</code></td>
<td>Exits to privileged EXEC mode.</td>
</tr>
</tbody>
</table>
SIP: Configuring Call Forward Unregistered for SIP IP Phones

To configure Call Forward Unregistered (CFU) on SIP IP phones, follow these steps:

Prerequisites

- 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**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1 <code>enable</code></td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Router&gt; enable</td>
</tr>
<tr>
<td></td>
<td>• Enter your password if prompted.</td>
</tr>
<tr>
<td>Step 2 <code>configure terminal</code></td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Router# configure terminal</td>
</tr>
<tr>
<td>Step 3 <code>voice register dn tag</code></td>
<td>Enters voice register dn mode to define a directory number for a SIP phone, intercom line, voice port, or an MWI.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Router(config)#voice register dn 20</td>
</tr>
<tr>
<td>Step 4 <code>call-forward b2bua unregistered directory-number</code></td>
<td>Enables call forwarding for a SIP back-to-back user agent so that all incoming calls are forwarded to the unregistered directory-number.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Router(config-register-dn)#call-forward b2bua unregistered 2345</td>
</tr>
<tr>
<td>Step 5 <code>end</code></td>
<td>Returns to privileged EXEC mode.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Router(config-ephone)# end</td>
</tr>
</tbody>
</table>

**Troubleshooting Tips**

- 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.
Configuring Keepalive Timer Expiration in SIP Phones

To configure keepalive timer expiration in SIP phones, follow these steps:

**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**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td>Example: Router&gt; enable</td>
<td>• Enter your password if prompted.</td>
</tr>
<tr>
<td><strong>Step 2</strong> configure terminal</td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td>Example: Router# configure terminal</td>
<td></td>
</tr>
<tr>
<td><strong>Step 3</strong> voice service voip</td>
<td>Enters voice-service configuration mode and specifies voice-over-IP encapsulation.</td>
</tr>
<tr>
<td>Example: Router#(conf)voice service voip</td>
<td></td>
</tr>
<tr>
<td><strong>Step 4</strong> sip</td>
<td>Enters SIP configuration mode.</td>
</tr>
<tr>
<td>Example: Router#(conf-serv)sip</td>
<td></td>
</tr>
<tr>
<td><strong>Step 5</strong> registrar server {expires {max seconds} {min seconds} }</td>
<td>Enables SIP registrar functionality in Cisco Unified CME.</td>
</tr>
<tr>
<td>Example: Router(conf-serv-sip)#registrar server expires max 250 min 75</td>
<td>• expires—(Optional) Sets the active time for an incoming registration.</td>
</tr>
<tr>
<td></td>
<td>• max sec—(Optional) Maximum time for a registration to expire, in seconds. Range: 120 to 86400.</td>
</tr>
<tr>
<td></td>
<td>• min sec—(Optional) Minimum time for a registration to expire, in seconds.</td>
</tr>
<tr>
<td><strong>Step 6</strong> end</td>
<td>Returns to privileged EXEC mode.</td>
</tr>
<tr>
<td>Example: Routerconf-serv-sip)# end</td>
<td></td>
</tr>
</tbody>
</table>
SIP: Configuring Call-Forwarding-All Soft Key URI

To specify the uniform resource identifier (URI) for the call forward all (CfwdAll) soft key 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.

**Prerequisites**

- 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 “SIP: Configuring SIP-to-SIP Phone Call Forwarding” on page 812.

**Restrictions**

- 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 soft key, it is enabled on the primary line.

**SUMMARY STEPS**

1. `enable`
2. `configure terminal`
3. `voice register global`
4. `call-feature-uri cfwdall service-uri`
5. `end`

**DETAILED STEPS**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> <code>enable</code></td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td><strong>Example:</strong> <code>Router&gt; enable</code></td>
<td></td>
</tr>
<tr>
<td><strong>Step 2</strong> <code>configure terminal</code></td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td><strong>Example:</strong> <code>Router# configure terminal</code></td>
<td></td>
</tr>
<tr>
<td><strong>Step 3</strong> <code>voice register global</code></td>
<td>Enters voice register global configuration mode to set global parameters for all supported SIP phones in a Cisco Unified CME environment.</td>
</tr>
<tr>
<td><strong>Example:</strong> <code>Router(config)# voice register global</code></td>
<td></td>
</tr>
</tbody>
</table>
SIP: Specifying Number of 3XX Responses To be Handled

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.

**Prerequisites**

- 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**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td>Example:</td>
<td>Router&gt; enable</td>
</tr>
<tr>
<td><strong>Step 2</strong> configure terminal</td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td>Example:</td>
<td>Router# configure terminal</td>
</tr>
</tbody>
</table>
How to Configure Call Transfer and Forwarding

SIP: Configuring Call Transfer

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

Prerequisites

Cisco CME 3.4 or a later version.

Restrictions

- 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 “SIP: Modifying Soft-Key Display” on page 1343.

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

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 3 voice register global</td>
<td>Enters voice register global configuration mode to set parameters for all supported SIP phones in Cisco Unified CME.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router(config)# voice register global</td>
<td></td>
</tr>
<tr>
<td>Step 4 phone-redirect-limit number</td>
<td>Changes the default number of 3XX responses a SIP phone that originates a call can handle for a single call.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router(config-register-global)#</td>
<td></td>
</tr>
<tr>
<td>phone-redirect-limit 8</td>
<td></td>
</tr>
<tr>
<td>Step 5 end</td>
<td>Exits to privileged EXEC mode.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router(config-register-global)#</td>
<td></td>
</tr>
<tr>
<td>end</td>
<td></td>
</tr>
</tbody>
</table>
### Detailed Steps

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router&gt; enable</td>
<td></td>
</tr>
<tr>
<td></td>
<td>• Enter your password if prompted.</td>
</tr>
<tr>
<td><strong>Step 2</strong> configure terminal</td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router# configure terminal</td>
<td></td>
</tr>
<tr>
<td><strong>Step 3</strong> voice register template</td>
<td>Enters voice register template configuration mode to define a template</td>
</tr>
<tr>
<td>template-tag range: 1 to 5</td>
<td>of common parameters for SIP phones in Cisco Unified CME.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router(config)# voice register</td>
<td></td>
</tr>
<tr>
<td>template 1</td>
<td></td>
</tr>
<tr>
<td><strong>Step 4</strong> transfer-attended</td>
<td>Enable a soft key for attended transfer on any supported SIP phone that</td>
</tr>
<tr>
<td></td>
<td>uses a template in which this command is configure.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router(config-register-template)#</td>
<td></td>
</tr>
<tr>
<td>transfer-attended</td>
<td></td>
</tr>
<tr>
<td><strong>Step 5</strong> transfer-blind</td>
<td>Enable a soft key for blind transfer on any supported SIP phone that</td>
</tr>
<tr>
<td></td>
<td>uses a template in which this command is configure.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router(config-register-template)#</td>
<td></td>
</tr>
<tr>
<td>transfer-blind</td>
<td></td>
</tr>
<tr>
<td><strong>Step 6</strong> exit</td>
<td>Exits configuration mode to the next highest mode in the configuration</td>
</tr>
<tr>
<td></td>
<td>mode hierarchy.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router(config-register-template)#</td>
<td></td>
</tr>
<tr>
<td>exit</td>
<td></td>
</tr>
<tr>
<td><strong>Step 7</strong> voice register pool</td>
<td>Enters voice register pool configuration mode to set phone-specific</td>
</tr>
<tr>
<td>pool-tag</td>
<td>parameters for SIP phones.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router(config)# voice register</td>
<td></td>
</tr>
<tr>
<td>pool 3</td>
<td></td>
</tr>
<tr>
<td><strong>Step 8</strong> template</td>
<td>Applies a template created with the <strong>voice register template</strong></td>
</tr>
<tr>
<td>template-tag range: 1 to 5</td>
<td>command.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router(config-register-pool)#</td>
<td></td>
</tr>
<tr>
<td>voice register pool 1</td>
<td></td>
</tr>
<tr>
<td><strong>Step 9</strong> end</td>
<td>Exits to privileged EXEC mode.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router(config-register-pool)#</td>
<td></td>
</tr>
<tr>
<td>end</td>
<td></td>
</tr>
</tbody>
</table>
Configuration Examples for Call Transfer and Forwarding

The following configuration examples are included in this section:

- H.450.2 and H.450.3: Example, page 822
- Basic Call Forwarding: Example, page 823
- Call Forwarding Blocked for Local Calls: Example, page 823
- Configuring Transfer Patterns: Example, page 823
- Configuring Maximum Length of Transfer Number: Example, page 823
- Configuring Conference Transfer Patterns: Example, page 824
- Blocking All Call Transfers: Example, page 824
- Selective Call Forwarding: Example, page 824
- Call Transfer: Example, page 824
- Call-Transfer Recall: Example, page 825
- H.450.12: Example, page 826
- H.450.7 and QSIG Supplementary Services: Example, page 826
- Cisco Unified CME and Cisco Unified Communications Manager in Same Network: Example, page 826
- H.450 Tandem Gateway Working with Cisco Unified CME and Cisco Unified Communications Manager: Example, page 828
- Forwarding Calls to Cisco Unity Express: Example, page 830
- Configuring Call Forward Unregistered for SIP IP Phones: Example, page 831
- Configuring Keepalive Timer Expiration in SIP Phones: Example, page 831

H.450.2 and H.450.3: Example

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
Basic Call Forwarding: Example

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.

```plaintext
ephone-dn 20
  number 2777
  call-forward all 2513
  call-forward busy 2513
  call-forward noan 2513 timeout 45
  call-forward night-service 2879
```

Call Forwarding Blocked for Local Calls: Example

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.

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

Configuring Transfer Patterns: Example

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

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

Configuring Maximum Length of Transfer Number: Example

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.

```plaintext
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:

```plaintext
Router# configure terminal
Router(config)# voice register template 2
Router(config-register-temp)# transfer max-length 10
```
**Configuring Conference Transfer Patterns: Example**

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
```

**Blocking All Call Transfers: Example**

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
```

**Selective Call Forwarding: Example**

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
```

**Call Transfer: Example**

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
```
Call-Transfer Recall: Example

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
**H.450.12: Example**

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
```

**H.450.7 and QSIG Supplementary Services: Example**

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
```

**Cisco Unified CME and Cisco Unified Communications Manager in Same Network: Example**

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
Configuring Call Transfer and Forwarding

Configuration Examples for Call Transfer and Forwarding

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Cisco Unified Communications Manager Express System Administrator Guide

OL-10663-03

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

H.450 Tandem Gateway Working with Cisco Unified CME and Cisco Unified Communications Manager: Example
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 CME2
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

Forwarding Calls to Cisco Unity Express: Example

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
Configuring Call Forward Unregistered for SIP IP Phones: Example

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
```

Configuring Keepalive Timer Expiration in SIP Phones: Example

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 “Generating Configuration Files for Phones” on page 353.

Soft Keys
To block the function of the call-forward-all or transfer soft key without removing the key display or to remove the soft key from one or more phones, see the “How to Customize Soft Keys” section on page 1339.

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 “Configuring Feature Access Codes” on page 1131.

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 “Configuring Call Coverage Features” on page 837.

Additional References

The following sections provide references related to Cisco Unified CME features.

Related Documents

<table>
<thead>
<tr>
<th>Related Topic</th>
<th>Document Title</th>
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</thead>
<tbody>
<tr>
<td>Cisco Unified CME configuration</td>
<td>• <a href="#">Cisco Unified CME Command Reference</a></td>
</tr>
<tr>
<td></td>
<td>• <a href="#">Cisco Unified CME Documentation Roadmap</a></td>
</tr>
<tr>
<td>Cisco IOS commands</td>
<td>• <a href="#">Cisco IOS Voice Command Reference</a></td>
</tr>
<tr>
<td></td>
<td>• <a href="#">Cisco IOS Software Releases 12.4T Command References</a></td>
</tr>
<tr>
<td>Cisco IOS configuration</td>
<td>• <a href="#">Cisco IOS Voice Configuration Library</a></td>
</tr>
<tr>
<td></td>
<td>• <a href="#">Cisco IOS Software Releases 12.4T Configuration Guides</a></td>
</tr>
<tr>
<td>Phone documentation for Cisco Unified CME</td>
<td>• <a href="#">User Documentation for Cisco Unified IP Phones</a></td>
</tr>
</tbody>
</table>
## Technical Assistance

<table>
<thead>
<tr>
<th>Description</th>
<th>Link</th>
</tr>
</thead>
<tbody>
<tr>
<td>The Cisco Support website provides extensive online resources, including documentation and tools for troubleshooting and resolving technical issues with Cisco products and technologies. To receive security and technical information about your products, you can subscribe to various services, such as the Product Alert Tool (accessed from Field Notices), the Cisco Technical Services Newsletter, and Really Simple Syndication (RSS) Feeds. Access to most tools on the Cisco Support website requires a Cisco.com user ID and password.</td>
<td><a href="http://www.cisco.com/techsupport">http://www.cisco.com/techsupport</a></td>
</tr>
</tbody>
</table>
Feature Information for Call Transfer and Forwarding

Table 69 lists the features in this module and enhancements to the features by version.


Use Cisco Feature Navigator to find information about platform support and software image support. Cisco Feature Navigator enables you to determine which Cisco IOS software images support a specific software release, feature set, or platform. To access Cisco Feature Navigator, go to http://www.cisco.com/go/cfn. An account on Cisco.com is not required.

Note: Table 69 lists the Cisco Unified CME version that introduced support for a given feature. Unless noted otherwise, subsequent versions of Cisco Unified CME software also support that feature.

<table>
<thead>
<tr>
<th>Feature Name</th>
<th>Cisco Unified CME Version</th>
<th>Feature Information</th>
</tr>
</thead>
<tbody>
<tr>
<td>Trunk-to-Trunk Transfer Blocking for Toll Fraud Prevention on Cisco Unified SIP IP Phones</td>
<td>9.5</td>
<td>Introduced support Trunk-to-Trunk Transfer Blocking for Toll Fraud Prevention on Cisco Unified SIP IP Phones.</td>
</tr>
</tbody>
</table>
| 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. |
### Feature Information for Call Transfer and Forwarding

<table>
<thead>
<tr>
<th>Feature Name</th>
<th>Cisco Unified CME Version</th>
<th>Feature Information</th>
</tr>
</thead>
</table>
| Call Forwarding           | 3.0                       | - 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.                                                                                                                                                                                                                                                                                               |
| Call Forward Unregistered | 1.0                       | Call forwarding for all calls, busy conditions, and no-answer conditions was introduced, using a Cisco-proprietary method.                                                                                                                                                                                                                                                                               |
|                           | 8.6                       | The Call Forward Unregistered (CFU) feature was introduced for SIP phones.                                                                                                                                                                                                                                                                                                                                      |
| 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 transfers 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.                                                                                                                                                           |
Table 69  Feature Information for Call Transfer and Forwarding

<table>
<thead>
<tr>
<th>Feature Name</th>
<th>Cisco Unified CME Version</th>
<th>Feature Information</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>3.1</td>
<td>Support was introduced for the following:</td>
</tr>
<tr>
<td></td>
<td></td>
<td>• 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).</td>
</tr>
<tr>
<td></td>
<td></td>
<td>• H.450.12 standards, which provide dynamic detection of H.450.2 and H.450.3 capabilities on a call-by-call basis.</td>
</tr>
<tr>
<td></td>
<td></td>
<td>• Automatic detection of Cisco Unified Communications Manager endpoints.</td>
</tr>
<tr>
<td></td>
<td></td>
<td>• Hairpin VoIP-to-VoIP call routing and routing to an H.450 tandem gateway.</td>
</tr>
<tr>
<td></td>
<td></td>
<td>• Hairpin call routing does not require a Tcl script.</td>
</tr>
<tr>
<td></td>
<td>3.0</td>
<td>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.</td>
</tr>
<tr>
<td></td>
<td>2.1</td>
<td>Consultative transfer using the ITU-T H.450.2 standard was introduced.</td>
</tr>
<tr>
<td></td>
<td>1.0</td>
<td>Call transfer was introduced, using a Cisco proprietary method.</td>
</tr>
</tbody>
</table>