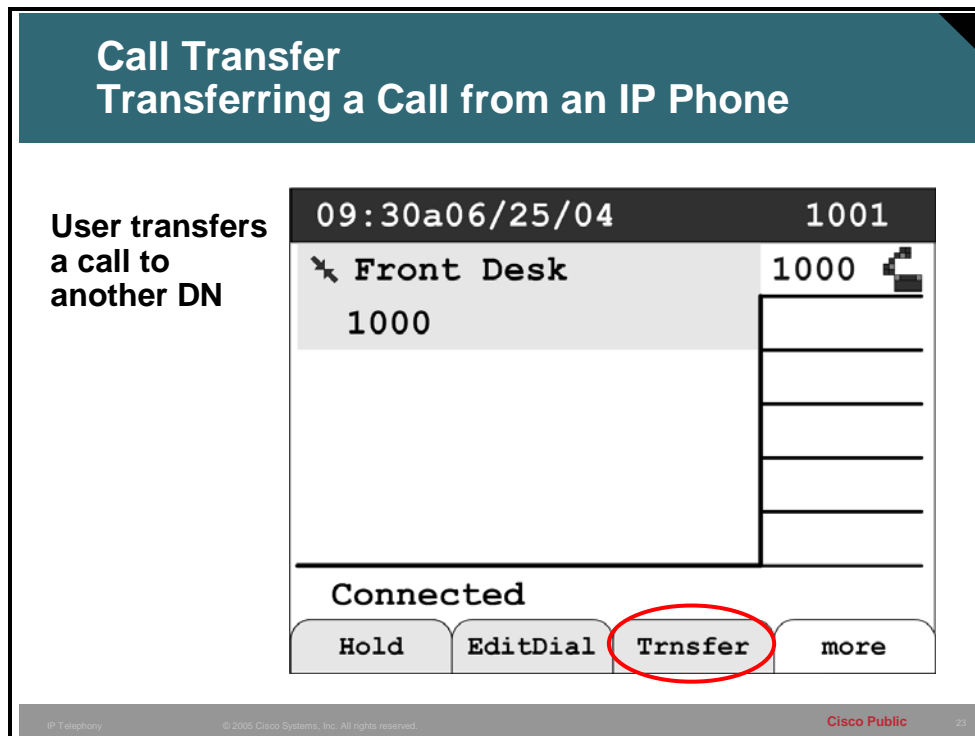


Configuring Phone Features

Call Transfer

This topic describes the Cisco CME transfer commands.



Transferring a caller to another directory number is a very common occurrence in the enterprise. The person on the IP phone can initiate a transfer by using the functions displayed on the IP phone display. To transfer a caller the user can initiate the transfer by pressing the "Trnsfer" softkey button and dialing the number that the call will be transferred to.

Depending on the configuration deployed on the Cisco CME system the call will either be blindly transferred or transferred with a consultation first. A blind transfer occurs when the transferor does not know if the extension that the call was transferred to will answer the call. The extension may be busy, ring no answer, or the extension may be answered. A consultative transfer differs in that the transferor is connected to the transfer target extension and then if satisfied finalizes the transfer which will connect the caller and the transfer target together.

Call Transfer (Cont.) Configuring

Call Transfer Commands

- **Configuring system transfer settings**
- **Configuring individual IP phone transfer settings**
- **Configuring a transfer pattern**

Call transfer is a function that can be configured in various ways, depending on the supported protocols. These call transfer commands include system-wide settings that can be overridden with phone specific settings, which can be overridden by settings on the transfer pattern.

Call Transfer (Cont.) Configuring

```
router(config-telephony-service)#
```

```
transfer-system {blind | full-blind | full-consult |  
local-consult}
```

- Specifies the call transfer method for all Cisco CME extensions

```
CMERouter(config-telephony-service)#transfer-system  
full-blind
```

Transfer System

To specify the call transfer method for IP phone extensions system-wide that use the ITU-T H.450.2 standard, use the **transfer-system** command in telephony-service configuration mode. To disable the call transfer method, use the **no** form of this command.

When H.450.2 call transfer is selected using the **full-blind** keyword the call is transferred without consultation using the H.450.2 standard. When a call is transferred using the **blind** keyword (the default) the call is blindly transferred using a single line and a Cisco proprietary method. When the **full-consult** keyword is used the call is transferred with consultation using the H.450.2 standard. The **local-consult** uses consultation with local calls and blindly transfers for non-local calls. The **local-consult** uses a proprietary transfer mechanism and is not commonly used.

Note Cisco CME 3.1 provides full call-transfer and call-forwarding interoperability with call processing systems on the network 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 3.1 provides VoIP-to-VoIP hairpin call routing without requiring the use of the special Tool Command Language (Tcl) script that was needed in earlier releases of Cisco CME.

Transfer System Command

Command	Description
<pre>transfer-system {blind full-blind full-consult local-consult}</pre> <p>Example:</p> <pre>Router(config-telephony-service)# transfer-system full-consult</pre>	<p>Defines the call transfer method to allow call transfer with consultation for all lines served by the router.</p> <ul style="list-style-type: none">• blind—Calls are transferred without consultation with a single phone line using the Cisco-proprietary method.• 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. The calls fall back to full-blind if the second line is unavailable.• local-consult—Local calls are transferred with local consultation using a second phone line if available. The calls fall back to blind for non-local consultation or non-local transfer target.

Call Transfer (Cont.) Configuring

```
router(config-ephone-dn)#
```

```
transfer-mode {blind | consult}
```

- **Specifies the type of call transfer for an individual IP phone extension number**

```
CMERouter(config-ephone-dn)#transfer-mode consult
```

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Transfer Mode

To specify the type of call transfer for an individual IP phone extension that uses the ITU-T H.450.2 standard, use the **transfer-mode** command in ephone-dn configuration mode. To remove this specification, use the **no** form of this command.

The **transfer mode** command specifies the type of call transfer for an individual Cisco IP phone extension that is using the ITU-T H.450.2 protocol. It allows you to override the system default **transfer-system** setting (full-consult or full-blind) for that ephone-dn extension. For example, in a Cisco CME network 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.

Transfer Mode Command

Command	Description
<pre>transfer-mode {blind consult}</pre> <p>Example: Router(config-ephone-dn)#transfer-mode blind</p>	<p>This command specifies the type of call transfer for an individual Cisco IP phone extension that is using the ITU-T H.450.2 protocol. It allows you to override the system default transfer-system setting (full-consult or full-blind) for that extension.</p> <ul style="list-style-type: none">• blind — Transfers calls without consultation using a single phone line.• consult — Transfers calls with consultation using a second phone line, if available. <p>.</p>

Call Transfer (Cont.) Configuring

```
router(config-telephony-service)#
```

```
transfer-pattern transfer-pattern [blind]
```

- To allow transfer of telephone calls from Cisco IP phones to phones other than Cisco IP phones

```
CMERouter(config-telephony-service)#transfer-pattern  
2095559... blind
```

Transfer Pattern

To allow the transfer of telephone calls from CallManager IP phones to non-local destinations, use the **transfer-pattern** command in telephony-service configuration mode. To disable these transfers, use the **no** form of this command.

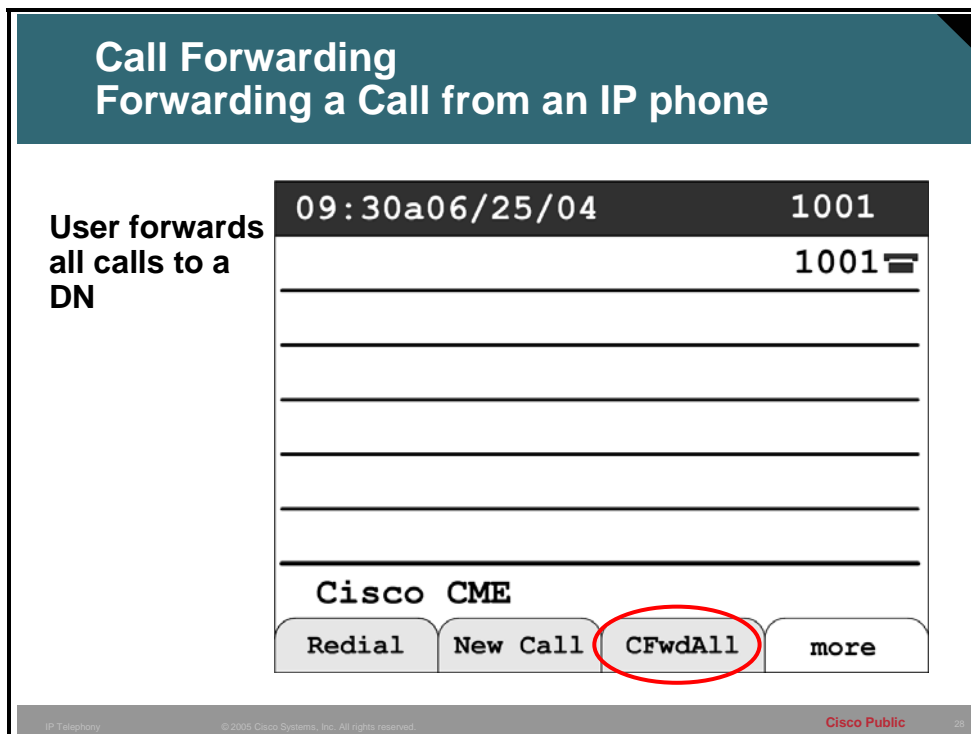
The **transfer-pattern** command allows you to transfer calls to destinations other than local IP Phones. This includes non-IP phones and external destinations. A call is then established between the transferred party and the new recipient. By default, all Cisco IP phone extension numbers are allowed as transfer targets. The default is that all transfers will be consultative in nature. The optional **blind** keyword forces calls that are transferred to numbers that match the transfer pattern to be executed as blind or full-blind transfers, overriding any settings made using the **transfer-system** and **transfer-mode** commands. When defining transfers to non-local numbers, it is important to note that 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.

Transfer Pattern Command

Command	Description
<p><code>transfer-pattern <i>transfer-pattern</i> [<i>blind</i>]</code></p> <p>Example: Router(config-telephony-service) # transfer-pattern .T</p>	<p>Allows transfer of telephone calls by Cisco 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.</p> <p><i>transfer-pattern</i>—String of digits for permitted call transfers. Wildcards are allowed. A pattern of .T transfers all calling parties using the H.450.2 standard.</p> <ul style="list-style-type: none">• 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. <p>Note When defining transfers to non-local numbers, it is important to note that 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.</p>

Call Forwarding

This topic describes the Cisco CME call forwarding commands.



Forwarding call settings govern the behaviors and destinations that the calls will be forwarded to in various situations that may occur such as the destination is busy, or the phone rings but no one answers, or all calls may be desired to be forwarded. On a user's phone the ring no answer and busy settings usually will be set by the administrator to go to the voice mailbox of that user. Other type of extensions may not have the same behavior and might have no settings or may forward to another extension constructing something like a hunt group.

The forward all setting can be set on the IP phone by the user. For example a user may be going on vacation and wishes all calls to be handled by another employee. This is a very common situation that occurs in many deployments.

Using the phone to set the forward all setting is done by pressing the "CFwdAll" softkey button and entering the number to which all calls will be forwarded, followed by the # key to tell the system you are done entering digits. The forward destination will be displayed on the bottom of the IP phone screen. To remove the forward all, press the "CFwdAll" softkey button again. This turns off the forward all. The user or administrator can also use the GUI Web interface of Cisco CME to configure the call forward options.

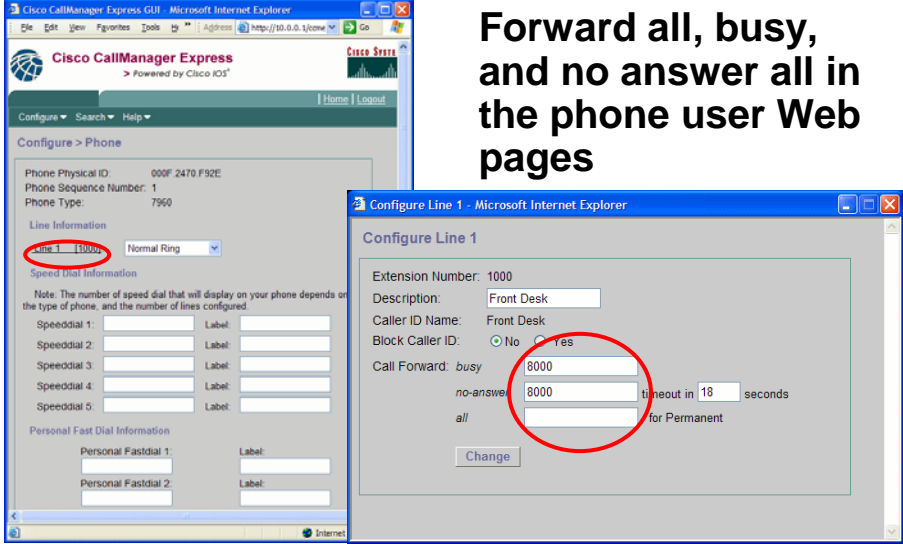
Call Forwarding (Cont.) Forwarding a Call from an IP phone

The image shows two overlapping browser windows from the Cisco CallManager Express GUI. The background window is titled 'Cisco CallManager Express' and shows the 'Configure > Phone' page. It displays phone details like 'Phone Physical ID: 000F.2470.F92E' and 'Phone Type: 7960'. Under 'Line Information', 'Line 1' is selected and circled in red. The foreground window is titled 'Configure Line 1' and shows configuration for 'Line 1'. It includes fields for 'Extension Number: 1000', 'Description: Front Desk', and 'Block Caller ID: No'. The 'Call Forward' section is circled in red and shows three options: 'busy' with destination '8000', 'no-answer' with destination '8000' and a '18' second timeout, and 'all' for permanent forwarding. A 'Change' button is at the bottom.

Forward all, busy, and no answer all in the phone user Web pages

From the phone user Web interface, the user is able to configure a line on the phone to forward all, forward busy and forward no answer. A user is only able to configure the phone on which he/she has credentials defined.

Call Forwarding (Cont.) Forwarding a Call from an IP phone



The image shows two overlapping screenshots from the Cisco CallManager Express GUI. The background screenshot is the 'Configure > Phone' page, showing fields for 'Phone Physical ID', 'Phone Sequence Number', and 'Phone Type'. A red circle highlights the 'Line 1' dropdown menu, which is set to '8000'. The foreground screenshot is the 'Configure Line 1' dialog box, showing fields for 'Extension Number' (1000), 'Description' (Front Desk), 'Caller ID Name' (Front Desk), and 'Block Caller ID' (No). The 'Call Forward' section is highlighted with a red circle and shows three options: 'busy' (8000), 'no-answer' (8000, timeout in 18 seconds), and 'all' (for Permanent).

Forward all, busy, and no answer all in the phone user Web pages

There are five call forwarding commands that can be configured from the command line of the Cisco CME router. These commands are the following:

- Call forward all (CLI, GUI, Phone)
- Call forward busy (CLI, GUI)
- Call forward no answer (CLI, GUI)
- Call forward max-length (CLI)
- Call forward pattern (CLI)

Call Forwarding (Cont.) Configuring

```
CMERouter(config-ephone-dn)#
```

```
call-forward all directory-number
```

- Forwards all calls to the Directory Number specified

```
CMERouter(config-ephone-dn)#
```

```
call-forward busy directory-number
```

- Forwards incoming calls when the DN is busy to another DN

```
CMERouter(config-ephone-dn)#
```

```
call-forward noan directory-number timeout seconds
```

- Forwards calls that are not answered in the time specified to another DN

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Call Forward All, Call Forward Busy and Call Forward No Answer

The **call-forward all** command can be configured from the CLI, this setting can also be made from GUI interface, or the IP phone. The **call-forward busy** command can be configured from the CLI and this setting can also be configured in the GUI interface. The **call-forward noan** command can be configured from the CLI and this setting can also be defined from the GUI interface.

Command	Purpose
<p>call-forward all <i>directory-number</i></p> <p>Example: Router(config-ephone-dn)#call-forward all 1001</p>	To configure call forwarding so that all incoming calls to an extension (ephone-dn) are forwarded to another extension.
<p>call-forward busy <i>directory-number</i></p> <p>Example: Router(config-ephone-dn)# call-forward busy 8000</p>	To configure call forwarding so that incoming calls to a busy extension (ephone-dn) are forwarded to another extension.
<p>call-forward noan <i>directory-number</i> timeout <i>seconds</i></p> <p>Example: Router(config-ephone-dn)# call-forward noan 8000 timeout 15</p>	To configure call forwarding so that incoming calls to an extension (ephone-dn) that does not answer are forwarded to another extension.

Call Forwarding (Cont.) Configuring

```
CMERouter(config-ephone-dn)#
```

```
call-forward max-length length
```

- Restricts the number of digits that can be used with call forward

```
CMERouter(config-telephony-service)#
```

```
call-forward pattern pattern
```

- To specify a pattern for calling-party numbers that support H.450.3

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Call Forward Maximum Length and Call Forward Pattern

The **call-forward max-length** command is used to restrict the number digits that a user can enter for the forwarding calls. This is important to prevent a user from forwarding calls to destinations that might incur toll charges. This command can only be configured from the command line. If the **call-forward max-length 0** is configured the call forward softkey will be grayed out and not available through the IP phone. However it is important to realize that the phone user can still set the forward setting in the GUI Web interface even if the max length is set to 0.

The **call-forward pattern** command attaches a pattern against the phone number of the calling party. When an extension number has forwarded its calls, and an incoming call is received for that number, the router sends an H.450.3 response back to the original calling party to request that the call be placed again using the forward-to destination. Calling numbers that do not match the patterns defined using this command are forwarded using Cisco-proprietary call forwarding for backward compatibility. Configuring this for numbers that do not support H.450.3 could result in dropped calls. This command should be configured to match only calling numbers that support H.450.3 protocol.

Command	Purpose
call-forward max-length <i>length</i> Example: Router(config-ephone-dn)#call-forward max-length 5	Restricts the number of digits that can be entered using the CfwdAll soft key on an IP phone..
call-forward pattern <i>pattern</i> Example: Router(config-telephony-service)#call-forward pattern 9.T	Specifies a pattern for calling-party numbers that are able to support the ITU-T H.450.3 standard for call forwarding,.

IP Phone Display

This topic describes the IP phone display options.

IP Phone Display Configuring

The Phone display can be customized in the following ways

- **IP Phone Header Bar**
- **System Text Message**
- **System Display Message (Idle URL)**

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The display of the IP phone can be customized to reflect the needs and identity of the enterprise that the CallManager system and phones are being deployed for.

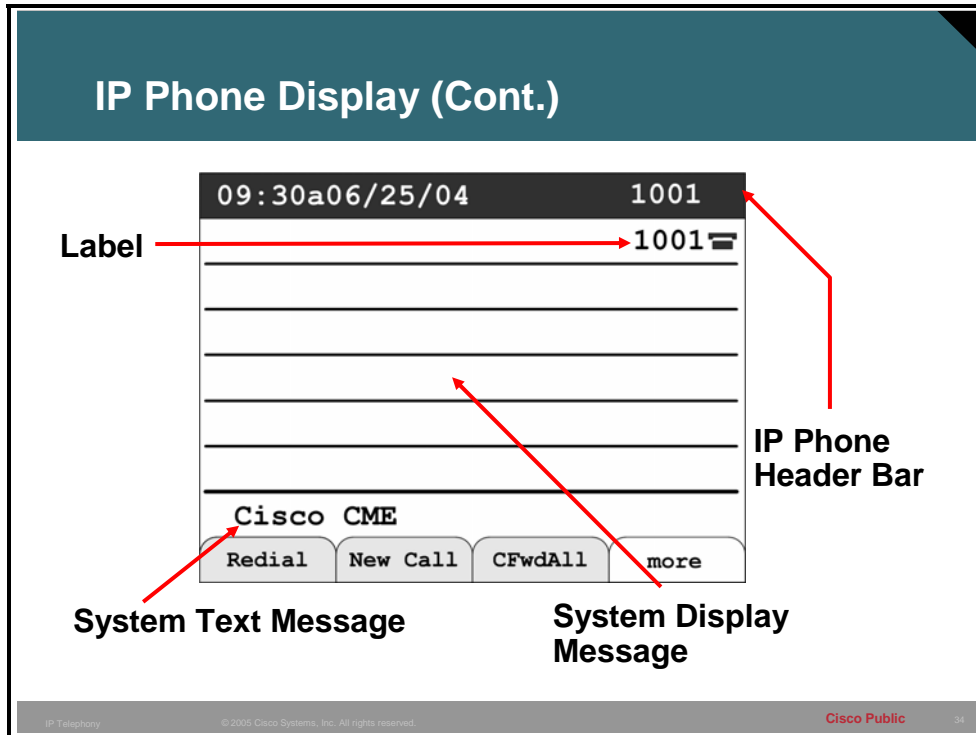
Normally, the IP phone header bar, or top line, of a 7940 or 7960 phone replicates the text that appears next to the first line button. The header bar can, however, contain a user-definable message instead of the extension number. For example, the header bar can be used to display a name or the full E.164 number of the phone. If no description is specified, the header bar replicates the extension number that appears next to the first button on the phone.

The system text message replaces the default “Cisco CallManager Express” message towards the bottom of the phone. There is room for about 30 characters to be displayed. The message will appear when the phone is idle. This occurs when one of the following three conditions is met:

- A busy phone goes on-hook
- A phone receives a keepalive
- The phone restarts

The system display message feature allows you to specify a file to display on 7940 and 7960 Cisco IP phones when they are not in use. You can use this feature to provide the phone display with a system message that is refreshed at configurable intervals, similar to the way that the system text message feature provides a message. The difference between the two is that the system text message feature displays a single line of text at the bottom of the phone display, whereas the system display message feature can use the entire display area and contain graphic images.

The system display message feature requires a back-end Web server to serve the browser page to the phone display, as the Cisco CME system only provisions the URL. The system display message display can also provide soft keys for the phone and thereby take input from the phone user for interactive services.



This graphic shows the different areas that will be present on the display of a Cisco CME controlled phone. These features can be customized for the current implementation.

IP Phone Display (Cont.)

IP Phone Header Bar

```
CMERouter(config)#
```

```
ephone-dn dn-tag
```

- Enters ephone-dn configuration mode

```
CMERouter(config-ephone-dn)#
```

```
description display-text
```

- Enters the header bar for the IP Phone

```
CMERouter(config-ephone-dn)#
```

```
label string
```

- Configures a label on the line instead of the line number

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The **description** command is used to change the IP header bar of a phone. A common use of this command is to enter the DID (if there is one) of the first line. This allows users to easily see the number that someone on the PSTN could use to call them on the current phone. However, any text or numbers could be displayed here.

To create a text identifier instead of a phone-number display for an extension on an IP phone console uses the **label** command.

Note The phone must be reset to have any changes to the header bar appear

Command	Purpose
<p>ephone-dn <i>dn-tag</i></p> <p>Example: Router(config)#ephone-dn 5</p>	<p>Enters ephone-dn configuration mode.</p> <ul style="list-style-type: none"> • <i>dn-tag</i>—The unique sequence number that identifies the ephone-dn for which the description should be in effect.
<p>Description <i>display-text</i></p> <p>Example: Router(config-ephone-dn)#description 206-295-7561</p>	<p>Defines a description for the header bar of a Cisco IP Phone 7940 or Cisco IP Phone 7960 that has the specified ephone-dn associated with its first line button.</p> <ul style="list-style-type: none"> • <i>display-text</i>—Alphanumeric character string, up to 40 characters. String is truncated to 14 characters in the display.
<p>Label <i>string</i></p> <p>Example: Router(config-ephone-dn)#label user10</p>	<p>To create a text identifier instead of a phone-number display for an extension on an IP phone console. The phone must be rebooted to receive the changes.</p>

IP Phone Display (Cont.)

System Display Message (Idle URL)

```
CMERouter(config-telephony-service)#
```

```
system message text-message
```

- Sets the text message to play when the phone is idle

```
CMERouter(config-telephony-service)#
```

```
url idle url idle-timeout seconds
```

- Sets a url to be displayed on the IP phone when idle for the set number of seconds

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The **system message** command allows a message to be displayed on all phones in the Cisco CME system. This message is alphanumeric text only and the size of the message will vary based upon the model of phone. A common use of this command is to display the name of the company on all the phones.

The **url idle** command allows the functionality of the **system message** command to be extended to more than just a text message. The **url idle** command allows the Cisco CME router to point all the phones to a URL that resides on a back-end Web server. This Web server can then provide content in the form of text, graphics or interactive applications that will appear on the IP phones after a definable period of inactivity. These applications are written using XML. For more information, go to <http://www.cisco.com/go/developersupport>.

Note The Cisco CME router should not be the Web server that serves the idle url as it will ask for credentials.

Command	Purpose
<p>System message <i>text-message</i></p> <p>Example: Router(config-telephony-service)#system message IPTXClass</p>	<p>Sets the message to display when a phone is idle. Uses proportional-width font, so the number of characters that are displayed varies based on the width of the characters that are used. The maximum number of displayed characters is approximately 30.</p>
<p>url idle <i>url</i> idle-timeout <i>seconds</i></p> <p>Example: Router(config-telephony-service)#url idle http://10.0.0.1/phoneapps/logo timeout 35</p>	<p>Defines a URL that contains a file to display on the phone when the phone is not in use, and specifies the interval between refreshes of the display, in seconds.</p> <ul style="list-style-type: none"> • url—Any URL that conforms to RFC 2396. • seconds—Range is from 0 to 300.

IP Phone Display (Cont.)

Provisioning URL for customized function buttons

```
CMERouter(config-telephony-service)#
```

```
url {directory | services} url
```

- Sets the url that will be used when the corresponding function button is pressed

```
telephony-service
```

```
url directories http://10.4.212.11/localedirectory
```

```
url services http://10.4.212.4/CCMUser/123456/urltest.html
```

The Cisco IP Phone 7940 and the Cisco IP Phone 7960 have customized function buttons that show the phone call status and activities on the display panels. These customized function buttons also invoke programmable non-call related services. There are two buttons that would be commonly modified to be linked to programmable URLs. This allows the administrator to override any default settings that may be assigned to the function buttons. The Messages and information button should not be customized and the Settings button cannot be customized.

Specific URLs are provisioned on the Cisco IP phone to populate these buttons. The URLs point to XML-based Web pages formatted with XML tags that the Cisco IP phone understands and uses. When you press a function button, the Cisco IP phone uses the configured URL to access the appropriate XML Web page for instructions. The Web page sends instructions to the Cisco IP phone to display information on the screen for you to navigate. You can select options and enter information by using soft keys and the scroll button.

The Cisco IP Phone 7940 and Cisco IP Phone 7960 can support four URLs in association with the four programmable feature buttons on IP phones. The four feature buttons on an IP phone are configured using the **url** command keywords. The Settings button is managed entirely by the phone and this fifth button may not be modified. Operation of these services is determined by the IP phone capabilities and the content of the referenced URL.

Note The Cisco CME router should not be the Web server that serves the idle url as it will ask for credentials.

URL Commands

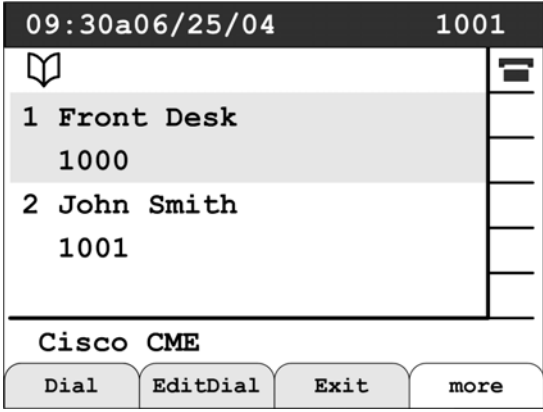

Command	Description
<p>url { directory information messages services } <i>url</i></p> <p>Example:</p> <pre>Router(config-telephony-service)# url information http://10.4.212.4/CCMUser/GetTelecaster HelpText.asp</pre>	<p>Provisions URLs for use by the Cisco IP phones. The four keywords (directory, information, messages, and services) correspond with the four feature buttons on an IP phone: Directories, Information, Messages, and Services. The purpose of the url command is simply to provision the URLs through the SEPDEFAULT.cnf configuration file supplied by the Cisco CME router to the Cisco IP phones during phone registration. The maximum character length for the URL is 128.</p> <p>You can disable the local directory by entering the url directories none command. You must reset the Cisco IP phones before the url command can take effect.</p> <p>Note By default, the router automatically uses the local directory service. Provisioning the directory URL to select an external directory resource disables the Cisco CME local directory service.</p>

Calling and Directory Features

This topic describes calling and directory features.

Calling and Directory Features

Directory can be accessed by pressing the directory button



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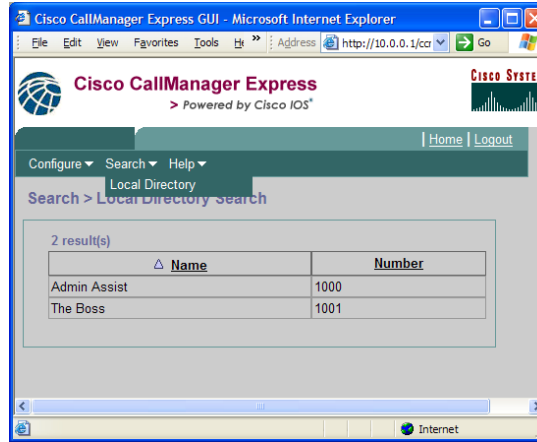
When a user does not know the number of another subscriber of commonly used external number the corporate directory that resides in the Cisco CME can be accessed and the number looked up and connected.

The directory of the Cisco CME is built and stored on the router from the configuration. It can be accessed by the phone users by pressing the directory button (assuming the url command has not been used to change this behavior) on the phone and then selecting the local directory or by using the phone user Web page. The user can be connected by pressing the dial softkey with an entry highlighted.

Calling and Directory Features (Cont.)

[HTTP://ip_address/ccme.html](http://ip_address/ccme.html)

The directory can be accessed through the phone user Web page



The directory can also be accessed through the GUI Web based interface by the phone user.

Calling and Directory Features (Cont.)

Configure > Extensions

Telephone Number	Sequence Number
<input type="checkbox"/> 1000	1
<input checked="" type="checkbox"/> 1001	2
<input type="checkbox"/> 1002	3
<input type="checkbox"/> 1100	8
<input type="checkbox"/> 1300	7
<input type="checkbox"/> 9000.....	9
<input type="checkbox"/> 9001.....	10

Change Extension

Extension Number : 1001
Name : Jane Doe
Block Caller ID : No Yes
Description :
Sequence Number : 2
Extension Type : Normal
Call Forward busy : 1999
Call Forward no-answer : 1999 timeout in seconds 10

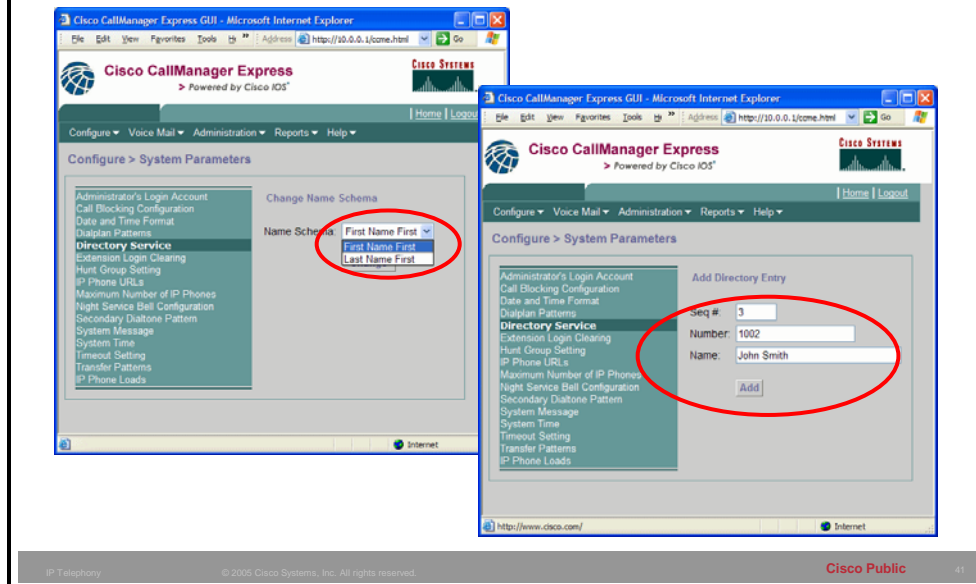
Advanced Features

Secondary Number :
E.164 Registration : Both Reg
Line Mode : Dual
Hunt Stop : Off On
Preference : 0
Preference (Sec) : 9
Hold-alert :
Pickup-Group :
Receive Night Service Bell : No Yes
Call Forward Permanent :

Adding a user to the directory of CallManager Express

To add a user to the directory of Cisco CME the name field under the properties of the ephone-dn will need to be defined. The name entered should be in the same order that the Cisco CME is set to use. This topic is discussed in the next slide.

Calling and Directory Features (Cont.)



The system administrator and, if allowed, the customer administrator can configure not only the order by which the names are listed in the directory but can also, through the Web page, add entries to the directory that are not Cisco CME controlled IP Phones.

Note This could be used to enter numbers for another site in the company that is not part of this Cisco CME installation.

Calling and Directory Features (Cont.)

Directory order and entry

```
router(config-telephony-service)#
```

```
directory {first-name-first | last-name-first}
```

- Sets the order that the directory will be listed in

```
router(config-telephony-service)#
```

```
directory entry {[entry-tag number name name] | clear}
```

- Sets the order that the directory will be listed in

```
router(config-ephone-dn)#
```

```
name name
```

- Associates a name with an extension (ephone-dn)

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The **directory** command is used to set the system-wide setting for how names will be displayed in the CME directory. The default is first name first. Entries that represent non-IP phones controlled by Cisco CME are entered into the directory from the CLI using the **directory entry** command. This can also be done by using the GUI, as seen in the previous page.

The **name** command is how an identity is associated to the ephone-dn in Cisco CME. Enter the name in the same order that was defined using the **directory** command. These names will appear in the Cisco CME directory.

Command	Purpose
<p>directory {first-name-first first-name-last}</p> <p>Example: Router(config-telephony-service)#directory first-name-first</p>	<p>Defines the local directory naming order. Note: the actual directory of names and phone numbers is built using the name command and the number command in ephone-dn configuration mode. When the command is set with the first-name-first keyword, you see the directory information displayed on the phone, for example, Jane J. Jones; and when the command is set with the last-name-first keyword, you see the directory information displayed on the phone as, for example, Jones, Jane J.</p>
<p>directory entry {[entry-tag number name name] clear}</p> <p>Example: Router(config-telephony-service)#directory entry entry-tag 1 2065551212 name Seattle</p>	<p>To add an entry to a local phone directory that can be displayed on IP phones.</p>
<p>name name</p> <p>Example: Router(config-ephone-dn)# name John Smith</p> <p>Example: Router(config-ephone-dn)# name Smith John</p>	<p>Associates a name to an ephone-dn. The name should follow the entry that order defined in the directory {first-name-first first-name-last} command.</p>

Productivity Tools

This topic describes the Cisco CME productivity tools.

Productivity Tools

Productivity tools for Cisco CME include:

- Flash softkey for hookflash functionality
- Intercom
- Paging

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Productivity Tools (Cont.)

Flash Softkey for Hookflash Functionality

```
router(config-telephony-service)#
```

```
fxo hook-flash
```

- Enables the flash softkey button on the IP phones

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Certain PSTN services, such as three-way calling and call waiting, require hookflash intervention from a phone user. A new soft key labeled “Flash” has been introduced to provide this functionality on FXO lines attached to the Cisco CME system. The “Flash” soft key is enabled using the **fxo hook-flash** command. Once Flash has been enabled and a reboot of the IP phone performed, the softkey is available to provide hookflash functionality during all calls except for local IP-phone-to-IP-phone calls.

Note The hookflash-controlled services can be activated only if they are supported by the PSTN connection that is involved in the call. The availability of the Flash soft key does not guarantee that hookflash-based services are actually accessible to the phone user.

Command	Purpose
fxo hook-flash Example: Router(config-telephony-service)#fxo hook-flash	Enables the flash button on the 7940 and 7960 phones for hookflash functionality

Productivity Tools (Cont.)

Intercom

```
ephone-dn 10
 number 1100
 name "Admin Assistant"
ephone-dn 20
 number 1199
 name "The Boss"
ephone-dn 51
 number B5000
 name "Admin Assistant"
 intercom B4999 label "Admin Assist"
ephone-dn 52
 number B4999
 name "The Boss"
 intercom B5000 label "The Boss"
ephone 6
 button 1:10 2 51
ephone 7
 button 1:20 2 52
```



Phone A – The Boss

Line 1 – 1100

Line 2 – Admin Assist



Phone B – Admin Assistant

Line 1 – 1199

Line 2 – The Boss

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Intercoms are a common feature that many deployments will desire. The intercom is commonly used between executives and assigned administrative assistants. While this is not the only situation where an intercom is used it is the most common.

Cisco CME supports intercom functionality for one-way and press-to-answer voice connections using a dedicated pair of intercom ephone-dns on two phones that speed-dial each other. When an intercom speed-dial button is pressed, a call is speed-dialed to the ephone-dn that is the other half of the dedicated pair. The called ephone-dn automatically answers the call in speakerphone mode with mute activated, which provides a one-way voice path from the initiator to the recipient.

A beep is sounded when the call is auto-answered to alert the recipient to the incoming call. To respond to the intercom call and open a two-way voice path, the recipient deactivates the mute function by taking one of the following actions:

- On a multibutton phone, pressing the **Mute** button.
- On a Cisco IP Phone 7910, lifting the handset.

Intercom lines cannot be used in shared-line configurations. If an ephone-dn is configured for intercom operation, it must be associated with one IP phone only. The intercom attribute causes an IP phone line (ephone-dn) to operate as an autodial line for outbound calls and as an auto answer-with-mute line for inbound calls. The slide shows an intercom between a receptionist and a manager.

Any user can dial the intercom if the number of the intercom is made with the keys present on the phones. In order to configure a non-dialable intercom line you can assign the intercom ephone-dn a dialing string with an alphabetic character of A, B, C or D. No one can dial the alphabetic character from a normal phone, but the phone at the other end of the intercom can be configured to dial the number that contains the alphabetic character through the Cisco CME router. For example, the intercom ephone-dns have been assigned numbers with alphabetic

characters so that no one but the receptionist can call the manager on their intercom line, and no one but the manager can call the receptionist on their intercom line.

Note An intercom requires configuration of two ephone-dns, one for each phone that makes up the intercom pair.

Productivity Tools (Cont.)

Intercom

```
router(config-ephone-dn)#
```

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

- Programs an extension to call another intercom phone

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The **intercom** command is used under the ephone-dn configuration mode; it is used to configure one half of the intercom pair. There will need to be another ephone-dn with the **intercom** command on it. This ephone-dn will then need to be assigned to a line button using the **button** command covered in Module 3. The IP phone will then need to be restarted to receive the changes.

Command	Purpose
<p>Intercom <i>extension-number</i> [barge-in no-auto-answer] [label <i>label</i>]</p> <p>Example: Router(config-ephone-dn)#intercom J3333 label "intercom to Joe"</p>	<p>Creates an intercom by programming a pair of extensions (ephone-dns) to automatically call and answer each other.</p> <p>barge-in (Optional) Allows inbound intercom calls to force an existing call into the call-hold state and allows the intercom call to be answered immediately.</p> <p>no-auto-answer (Optional) Disables the intercom auto-answer feature.</p> <p>label <i>label</i> (Optional) Defines an alphanumeric label for the intercom, of up to 30 characters.</p>

Productivity Tools (Cont.)

Paging Function

- **Creates a one way voice path**
- **Unicast or Multicast**
- **Single Group**
- **Multiple Groups**

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Audio paging provides a one-way voice path to the phones that have been designated to receive paging. It does not have a press-to-answer option like the intercom feature. Intercoms are commonly used to locate a person away from their desk, for emergency situations like a fire drill, for overhead pages using and other situations.

The paging mechanism supports audio distribution using IP multicast, replicated unicast, and a mixture of both (so that multicast is used where possible, and unicast is used for specific phones that cannot be reached using multicast).

Paging groups can be configured for a single group or for a combined group. Several paging groups can be specified in a Cisco CME system and two or more paging groups can be joined into a combined group.

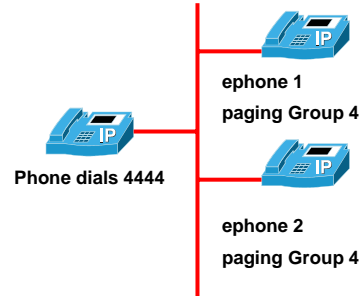
A paging group is created using a dummy ephone-dn, known as the paging ephone-dn, that can be associated with any number of local IP phones. The paging ephone-dn can be dialed from anywhere, including on-net.

When a caller dials the paging number (ephone-dn), each idle IP phone that has been configured with the paging number automatically answers using its speakerphone mode. Displays on the phones that answer the page show the caller ID that has been set using the **name** command under the paging ephone-dn. When the caller finishes speaking the message and hangs up, the phones are returned to their idle states.

Productivity Tools (Cont.)

Single Paging Group

```
ephone-dn 4
 number 4444
 name Paging Shipping
 paging ip 239.0.1.20 port 2000
ephone-dn 21
 number 2121
ephone-dn 22
 number 2222
ephone 1
 mac-address 3662.0234.6ae2
 button 1:21
 paging-dn 4
ephone 2
 mac-address 9387.6738.2873
 button 1:22
 paging-dn 4
```



The paging feature defines an ephone-dn that sends one-way voice through a unicast or multicast mechanism to a single group of idle Cisco IP phones that have been associated with the paging ephone-dn tag. In this example, when a caller dials 4444, both ephone 1 and ephone 2 receive a page on the speaker of the IP phone.

Productivity Tools (Cont.)

Paging DN set up by adding a new extension through the GUI

The image shows two overlapping web browser windows from the Cisco CME GUI. The left window, titled 'Add an Extension Number', has the 'Extension Number' field set to '9999' and the 'Extension Type' dropdown set to 'Paging'. The right window, titled 'Change Phone', has the 'Paging Extension' dropdown set to '9999'. Red circles highlight the '9999' values in both windows. Red arrows point from the text on the left to these highlighted values.

Assign the paging extension to the phone

Paging groups can be setup through the GUI by adding a new paging extension and then assigning it to one or more phones in the Cisco CME system.

Note An IP phone can only be assigned to one paging group at a time.

Productivity Tools (Cont.)

Paging Commands

```
router(config-ephone-dn)#
```

```
paging [ip multicast-address port udp-port]
```

- Configures the ephone-dn as paging extension using either unicast or multicast

```
router(config-ephone)#
```

```
paging-dn paging-dn-tag [unicast]
```

- Creates a paging extension to receive audio pages on the ephone

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To configure a paging DN first create the ephone-dn with the command **ephone-dn tag**. Then use the **paging** command to configure the ephone-dn as a paging extension, the paging ephone-dn can then be assigned to the target ephones. The **paging** command will by default use unicast for the pages and this limits the number of target phones to 10. If the **paging ip *ip_multicast address port udp-port*** is used then the page can go out to more than 10 phones. The configurable range of multicast addresses is 225.0.0.0 – 239.255.255.255.

The use of multicast for paging will provide for many ephones receiving the same page without generating a separate stream of traffic for each ephone. In fact all of the IP phones in the paging group can use the same stream thereby conserving bandwidth, increasing the scalability of the page, and reducing overhead on the Cisco CME router.

Note The network must be multicast enabled to support multicast of paging.

This assignment to one or more target ephones is done through the use of the **paging-dn** command in the ephone configuration mode for one or more ephones. The configuration of the paging ephone-dn determines whether a page will use a multicast or a unicast mechanism.

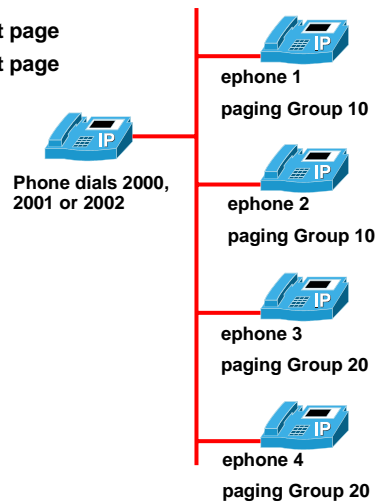
Command	Purpose
<p>paging [ip <i>multicast-address</i> port <i>udp-port</i>]</p> <p>Example Unicast: Router(config-ephone-dn)#paging</p> <p>Example Multicast: Router(config-ephone-dn)#paging ip 239.0.1.1 port 2000</p>	<p>Defines an extension (ephone-dn) as a paging extension that can be called to broadcast an audio page to a set of Cisco IP phones.</p> <p>ip <i>multicast-address</i> (Optional) Uses an IP multicast address to multicast voice packets for audio paging; for example, 239.0.1.1. Note that IP phones do not support multicast at 224.x.x.x addresses. Default is that multicast is not used and IP phones are paged individually using IP unicast transmission (up to ten phones).</p> <p>port <i>udp-port-number</i> (Optional) Uses this UDP port for the multicast. Range is from 2000 to 65535. Default is 2000.</p>
<p>paging-dn <i>paging-dn-tag</i> [unicast]</p> <p>Example: Router(config-ephone)#paging-dn 77 unicast</p>	<p>Create a paging extension (paging-dn) to receive audio pages on a Cisco IP phone in a Cisco CME system, the unicast keyword will override the multicast configuration in the paging command</p>

Productivity Tools (Cont.)

Combined Paging Group

- Phone dials 2000 and phone 1 and phone 2 get page
- Phone dials 2001 and phone 3 and phone 4 get page
- Phone dials 2002 and all 4 phones get page

```
ephone-dn 10
number 2000
name Tech Support
paging ip 239.0.1.20 port 2000
ephone-dn 20
number 2001
name Sales Dept
paging ip 239.0.1.21 port 2000
ephone-dn 22
number 2002
paging ip 239.0.2.22 port 2000
paging group 10,20
ephone 1
paging-group 10
ephone 2
paging-group 10
ephone 3
paging-group 20
ephone 4
paging-group 20
```



By configuring the ability to page combined groups in addition to single groups, you provide phone users with the flexibility to page a small local paging group, like paging two phones in a company's technical support department or to page a combined set of several paging groups by paging a group that consists of technical support and sales phones.

Productivity Tools (Cont.)

Paging Group

```
router(config-ephone-dn)#
```

```
paging-group paging-dn-tag, paging-dn-tag,...
```

- **Creates a combined paging group from two or more previously defined paging directory numbers**

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The **paging-group** command is configured under a paging ephone-dn. This allows multiple paging groups to be combined into larger groups. A common use of this is a system-wide emergency page.

Command	Purpose
Paging-group <i>paging-dn tag, paging-dn tag,...</i> Example: Router(config-ephone-dn)#paging-group 10, 20, 30, 40	Creates a combined paging group from two or more previously established paging sets.

Custom IP Phone Rings

This topic describes custom IP phone rings.

Custom IP Phone Rings

To create custom rings for IP phones follow these steps:

- **Step 1 - Create a ring in the form of a raw PCM file**
- **Step 2 - Create a RingList.xml file using a text editor to point to the various rings that are desired**
- **Step 3 - Load the rings and RingList.xml file to flash on the CallManager Express router**
- **Step 4 - Configure the tftp server to serve up the rings and RingList.xml**
- **Step 5 - Reboot the phones**

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One of the unique features that is enabled by using IP phones is the ability for the administrator to make custom rings that will be available for the users to select from. While this is not commonly a mission critical feature it is a commonly requested feature.

The IP Phone rings can be customized by creating your own PCM audio files and constructing a custom RingList.xml file.

Cisco IP Phones ship with two default ring types that are implemented in hardware: Chirp1 and Chirp2. Cisco IOS Telephony Service is also implemented to support custom ring sounds that are implemented in software as pulse code modulation (PCM) files. An XML file named RingList.xml that describes the ring list options available at your site is needed on the flash of Cisco CME router.

The procedure described below only applies to creating custom phone rings for the Cisco IP Phone 7940 and 7960 models.

Step 1

Create a PCM file for each custom ring (one ring per file). The PCM files for the rings must meet the following requirements for proper playback on Cisco IP Phones:

- Raw PCM (no header)
- 8000 samples per second
- 8 bits per sample
- uLaw compression

- Maximum ring size — 16080 samples
- Minimum ring size — 240 samples
- Number of samples in the ring must be evenly divisible by 240
- Ring should start and end at the zero crossing

Step 2

Use an ASCII editor to edit the RingList.xml file.

Step 3

TFTP the new PCM files and XML file to the flash of the Cisco CME router

Step 4

Set **tftp-server** command to allow access to the files

e.g. **tftp-server flash:RingList.xml**

tftp-server flash:Piano1.raw

Step 5

Reboot the IP Phones. When IP Phones are rebooted, the IP Phones will get the files and show the ring types in Ring Type Option list under Settings button.

The RingList.xml file defines an XML object that contains a list of phone ring types. Each ring type contains a pointer to the PCM file used for that ring type and the text that will be displayed on the Ring Type menu on a Cisco IP Phone for that ring. The CiscoIPPhoneRingList XML object uses the following simple tag set to describe the information:

```
<CiscoIPPhoneRingList>
<Ring>
<DisplayName/>
<FileName/>
</Ring>
</CiscoIPPhoneRingList>
```

In the above definition:

Ring contains two fields, DisplayName and FileName, that are required for each phone ring type. Up to 50 rings can be listed.

- DisplayName defines the name of the custom ring for the associated PCM file that will be displayed on the Ring Type menu of the Cisco IP Phone.
- FileName specifies the name of the PCM file for the custom ring to associate with DisplayName.

- The DisplayName and FileName fields must not exceed 25 characters.

Example: Sample RingList.xml

A sample RingList.xml file that defines two phone ring types is shown below:

```
<CiscoIPPhoneRingList>  
  
<Ring>  
  
<DisplayName>Piano1</DisplayName>  
  
<FileName>Piano1.raw</FileName>  
  
</Ring>  
  
<Ring>  
  
<DisplayName>Chime</DisplayName>  
  
<FileName>Chime.raw</FileName>  
  
</Ring>  
  
</CiscoIPPhoneRingList>
```

Music on Hold

This topic describes music on hold (MoH).

Music on Hold

Music on Hold (MoH) can be derived from two sources:

- **Audio file in .wav or .au format**
- **Live audio source via a feed**

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MoH is an audio stream that is played to PSTN and VoIP G.711 callers who are placed on hold by phones in a Cisco CME system. This audio stream is intended to reassure callers that they are still connected to their calls. MoH is not played to local Cisco CME phones that are on hold with other Cisco CME phones. These parties hear a periodic repeating tone instead.

The audio stream that is used for MoH can derive from one of two sources: an audio file or a live feed. If both are configured concurrently on the Cisco CME router, the router seeks the live feed first. If the live feed is found, it displaces the audio file source. If the live feed is not found or fails at any time, the router falls back to the audio file source that was specified for MoH during configuration.

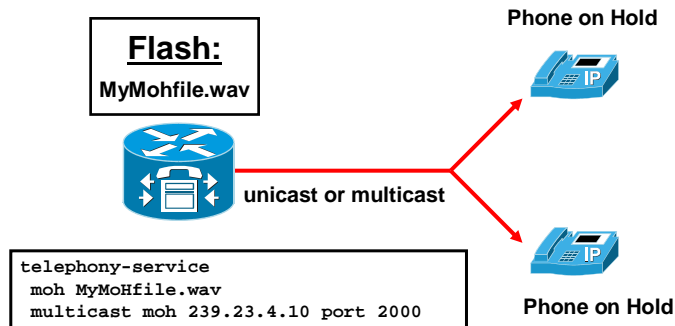
If the MoH audio stream is also identified as a multicast source, the Cisco CME router additionally transmits the stream on the physical IP interfaces of the Cisco CME router that you specify during configuration, which permits external devices to have access to it.

A MoH audio stream from an audio file is supplied from an .au or .wav file held in router Flash memory. A MoH audio stream from a live feed is supplied from a standard line-level audio connection that is directly connected to the router through an FXO or E&M analog voice port. The live-feed feature is typically used to connect to a CD jukebox player. Only one live MoH feed is supported per system.

Caution Use royalty-free music to avoid potential legal issues.

Music on Hold (Cont.)

Configuring MoH from file



This graphic shows a file name “MyMohfile.wav” in flash of the Cisco CME router. The file is configured to be used for music on hold by entering the **moh MyMoHfile.wav** command in telephony service mode. It is currently configured to use the multicast address of 239.23.4.10 for transmission of the music on hold.

Music on Hold (Cont.)

Configuring MoH from file

```
router(config-telephony-service)#
```

```
moh filename
```

- Configures MoH using the file specified in flash

```
router(config-telephony-service)#
```

```
multicast moh ip-address port port-number [route ip-address-list]
```

- (Optional) Specifies that the MoH file should be multicast using the parameters specified

The **moh filename** command enables MoH from .au and .wav format music files. MoH is played for G.711 callers and on-net VoIP and PSTN callers who are on hold in a Cisco CME system. Local callers within a Cisco CME system hear a repeating tone while they are on hold. Audio files that are used for MoH must be copied to the Cisco CME router Flash memory. A MoH file can be in .au or .wav file format; however, the file format must contain 8-bit 8-kHz data in a-law or mu-law data format.

To replace or modify the audio file that is currently specified, you must first disable the MoH capability using the **no moh** command. The following example replaces file1 with file2:

- Router(config-telephony-service)# moh file1
- Router(config-telephony-service)# no moh
- Router(config-telephony-service)# moh file2

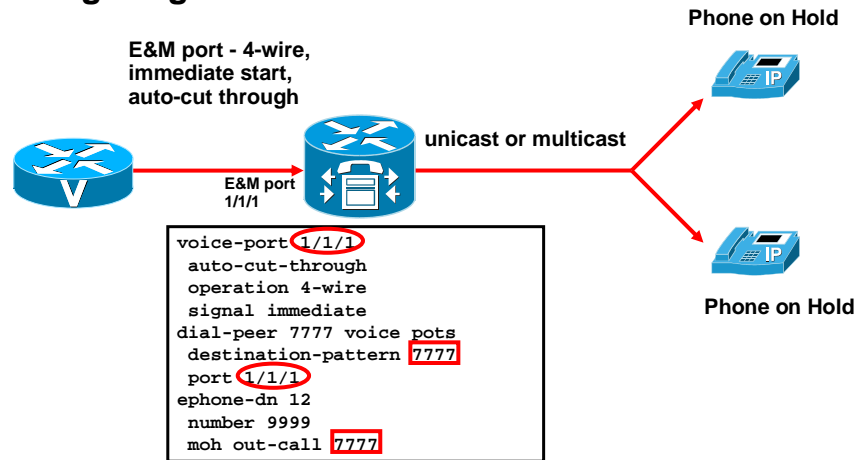
If a second file is specified without first removing the original file, the MoH mechanism stops working and may require a router reboot to clear the problem.

Note IP phones do not support multicast at 224.x.x.x addresses.

Command	Purpose
<p>moh filename</p> <p>Example: Router(config-telephony-service)#moh MyMoHfile.wav</p>	<p>Generates an audio stream from a file for music on hold (MoH) in a Cisco CME system.</p>
<p>multicast moh ip address port port-number [route ip-address-list]</p> <p>Example: Router(config-telephony-service)#moh 239.2.2.2 port 2000 route 10.0.0.1 11.0.0.1</p>	<p>Configures MoH audio stream as a multicast source in a Cisco CME system.</p> <p>ip-address Specifies the destination IP address for multicast.</p> <p>port port-number Specifies the media port for multicast. Range is from 2000 to 65535. Port 2000 is recommended because this port is already used for normal Real-Time Transport Protocol (RTP) media transmissions between IP phones and the Cisco CME router.</p> <p>route ip-address-list (Optional) Indicates specific router interfaces over which to transmit the IP multicast packets. Up to four IP addresses can be listed, each separated from the other by a space. The default is that the MoH multicast stream is automatically output on the interfaces that correspond to the address that was configured with the ip source-address command.</p>

Music on Hold (Cont.)

Configuring MoH from a live source



To configure MoH from a live feed, you establish a voice port and dial peer for the call and also create a “dummy” ephone-dn. The ephone-dn must have a phone or extension number assigned to it so that it can make and receive calls, but the number is never assigned to a physical phone. The recommended interface for live-feed MoH is an analog E&M port because it requires the minimum number of external components. You connect a line-level audio feed (standard audio jack) directly to pins 3 and 6 of an E&M RJ-45 connector.

The E&M Voice interface card (VIC) has a built-in audio transformer that provides appropriate electrical isolation for the external audio source. (An audio connection on an E&M port does not require loop-current.) The **signal immediate** and **auto-cut-through** commands disable E&M signaling on this voice port. A digital signal processor (DSP) on the E&M port generates a G.711 audio packet stream.

If you are using an FXO voice port for live-feed MoH instead of an E&M port, connect the MoH source to the FXO voice port. This connection requires an external adapter to supply normal telephone company (telco) battery voltage with the correct polarity to the tip and ring leads of the FXO port. The adapter must also provide transformer-based isolation between the external audio source and the tip and ring leads of the FXO port.

Music from a live feed is continuously fed into the MoH playout buffer instead of being read from a Flash file, so there is typically a 2-second delay. An outbound call to a MoH live-feed source is attempted (or reattempted) every 30 seconds until the connection is made by the directory number that has been configured for MoH. If the live-feed source is shut down for any reason, the Flash memory source will automatically activate. A live-feed MoH connection is established as an automatically connected voice call that is made by the Cisco CME MoH system itself or by an external source directly calling in to the live-feed MoH port.

Music on Hold (Cont.)

Configuring MoH from a live source

```
CMERouter(config)#
```

```
voice-port port
```

- Enters voice port config mode for the E&M port

```
CMERouter(config-voice-port)#
```

```
input gain decibels
```

- (Optional) sets gain on the MoH signal

```
CMERouter(config-voice port)#
```

```
auto-cut-through
```

- Enables call completion when no M-lead response is sent

If the MoH arrives on an analog port that FXO or E&M port will need to be configured. The command **input gain** allows the volume of the feed to be tuned up or down on either an FXO or E&M port. E&M ports require additional configuration, one of these E&M command is the **auto-cut-through** command which allows the connection to the feed to be set up even though the source of the feed will not provide a M-lead response.

Command	Purpose
voice-port <i>port</i> Example: Router(config)#voice-port 1/0/1	Enters voice port configuration mode
Input gain <i>decimals</i> Example: Router(config-voice-port)#input gain <i>decimals</i>	Specifies, in decibels, the amount of gain to be inserted at the receiver side of the interface. Acceptable values are integers from -6 to 14.
auto-cut-through Example: Router(config-voice-port)#auto-cut-through	(E&M ports only) Enables call completion when a device does not provide an M-lead response. MoH requires that you use this command with E&M ports.

Music on Hold (Cont.)

Configuring MoH from a live source

```
CMERouter(config-voice-port)#
```

```
operation 4-wire
```

- Sets the E&M port to use a 4-wire scheme

```
CMERouter(config-voice-port)#
```

```
signal immediate
```

- Directs calling side to seize the E-lead and send DTMF digits

The E&M port that music on hold arrives on will also need to be configured in 4-wire mode. This is done by entering the **operation 4-wire** command. E&M ports also needs to be configured to proceed with connecting the call by seizing the line and sending DTMF digits without waiting for any signal from the other side of the connection. This is done by the use of the command **signal immediate**.

Command	Purpose
operation 4-wire Example: Router(config-voice-port)#operations 4-wire	(E&M ports only) Selects the 4-wire cabling scheme. MoH requires that you specify 4-wire operation with this command for E&M ports.
signal immediate Example: Router(config-voice-port)#signal immediate	(E&M ports only) For E&M tie trunk interfaces, directs the calling side to seize a line by going off-hook on its E-lead and to send address information as DTMF digits.

Music on Hold (Cont.)

Configuring MoH from a live source

```
CMERouter(config)#
```

```
dial-peer voice tag pots
```

- Creates the dial peer for the MOH connection

```
CMERouter(config-dial-peer)#
```

```
port port
```

- Associates the voice port to the dial peer

```
CMERouter(config-dial-peer)#
```

```
destination-pattern string
```

- Specifies the DN of the MoH source

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The dial peer that is used to connect the physical E&M or FXO port to the destination pattern that will be used to connect to the MoH feed is created with the **dial-peer voice tag pots** command. The physical voice port that will be used is associated with the **port** command and the telephone number used is defined by the **destination-pattern** command.

MoH Commands

Command	Description
dial-peer voice tag pots Example: Router(config)# dial peer voice 7777 pots	Enters dial peer configuration mode
port port Example: Router(config-dial-peer)# port 1/1/0	Associates the dial peer with a voice port
destination-pattern string Example: Router(config-dial-peer)# destination-pattern 7777	Specifies either the prefix or the full E.164 telephone number to be used for a dial peer.

Music on Hold (Cont.)

Configuring MoH from a live source

```
CMERouter(config)#
```

```
ephone-dn tag
```

- **Creates an ephone-dn**

```
CMERouter(config-ephone-dn)#
```

```
number number
```

- **Configures a dummy DN for this ephone-dn instance**

```
CMERouter(config-ephone-dn)#
```

```
moh [out-call outcall-number]
```

- **Specifies this ephone-dn is used for an incoming or outgoing call that is to be the source for a MOH stream**

An ephone-dn will need to be configured with the **ephone-dn** command. Then the number of the source of the MoH feed will need to be configured with the **number** command. The ephone-dn will need the **moh** command to take the specified live-feed audio stream and uses it as MoH for a Cisco CME system. The connection for the live-feed audio stream is established as an automatically connected voice call. If the **out-call** keyword is used, the type of connection can include VoIP calls if voice activity detection (VAD) is disabled. The typical operation is for the MoH ephone-dn to establish a call to a local router E&M port.

If the out-call keyword is used, an outbound call to the MoH live-feed source is attempted (or reattempted) every 30 seconds until the call is connected to the ephone-dn (extension) that has been configured for MoH. Note that this ephone-dn is not associated with any physical phone.

If the MoH (ephone-dn) command is used without any keywords or arguments, the ephone-dn will accept an incoming call and use the audio stream from the call as the source for the MoH stream, displacing any audio stream that is available from a Flash file. To accept an incoming call, the ephone-dn must have an extension or phone number configured for it. A typical usage would be for an external H.323-based server device to call the ephone-dn to deliver an audio stream to the Cisco CME system. Normally, only a single ephone-dn would be configured like this. If there is more than one ephone configured to accept incoming calls for MoH, the first ephone-dn that is successfully connected to a call (incoming or outgoing) is the MoH source for the system.

MoH Commands

Command	Description
ephone-dn <i>dn-tag</i> Example: Router(config)# ephone-dn 55	Enters ephone-dn configuration mode. <ul style="list-style-type: none">• <i>dn-tag</i>—Unique sequence number that identifies this ephone-dn during configuration tasks. Range is from 1 to 288.
number <i>number</i> Example: Router(config-ephone-dn)# number 5555	Configures a valid extension number for this ephone-dn instance. This number is not assigned to any phone; it is only used to make and receive calls that contain an audio stream to be used for MoH. <ul style="list-style-type: none">• <i>number</i>—String of up to 16 digits that represents a telephone or extension number to be associated with this ephone-dn.
moh [out-call <i>outcall-number</i>] Example: Router(config-ephone-dn)# moh out-call 7777	Specifies that this ephone-dn is to be used for an incoming or outgoing call that is to be the source for an MoH stream. If this command is used without the out-call keyword, the MoH stream is received from an incoming call. <ul style="list-style-type: none">• out-call <i>outcall-number</i>—(Optional) Indicates that the router is calling out for a live feed that is to be used for MoH and specifies the number to be called. Forces a connection to the local router voice port that was specified