CHAPTER 5

Configuring Call-Control Integration for Cisco Unified MeetingPlace Express

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An external call-control device is required to route calls to and from Cisco Unified MeetingPlace Express. The following topics describe how to integrate Cisco Unified MeetingPlace Express with specific call-routing devices:

About Integration With Cisco Unified CallManager

We recommend that you use Cisco Unified CallManager to provide call-routing and other services for Cisco Unified MeetingPlace Express and your IP telephony network.

There are three call-routing service options for Cisco Unified CallManager integration, the benefits and restrictions of which are described in the following topics:

- Integration in an H.323 environment with a gatekeeper. This is the preferred method. See the "About Integration With a Call-Control Device Through a Gatekeeper" section on page 5-28.
- Integration in an H.323 environment without a gatekeeper. See the "Configuring the Cisco Unified CallManager Integration Without a Gatekeeper" section on page 5-2.
- Integration in a SIP environment. See the "About Integration in a SIP Environment" section on page 5-39.

Cisco Unified CallManager also provides the services described in the following topics:

- About Cisco Unified IP Phone Services, page 5-7
- About User Authentication By an External Directory, page 5-14

Related Topics

- Configuring Call-Control Integration for Cisco Unified MeetingPlace Express, page 5-1

Configuring the Cisco Unified CallManager Integration Without a Gatekeeper

This topic describes how to integrate Cisco Unified CallManager with Cisco Unified MeetingPlace Express in an H.323 environment. In this method, Cisco Unified MeetingPlace Express is configured as a gateway to Cisco Unified CallManager. No gatekeeper is used in this setup.

Note If you have a Cisco Unified CallManager cluster that provides call-processing redundancy, and if the primary Cisco Unified CallManager server goes down, Cisco Unified MeetingPlace Express cannot complete dial-out calls without a gatekeeper. For information about Cisco Unified CallManager integration with a gatekeeper, see the "About Integration With a Call-Control Device Through a Gatekeeper" section on page 5-28.

Prerequisites

- Verify that your Cisco Unified CallManager release and Cisco Unified MeetingPlace Express release are compatible. See the *Release Notes for Cisco Unified MeetingPlace Express*.
- Verify that your IP telephony network is already set up and working properly. For example:
  - Verify that the Cisco Unified IP Phones are properly connected and added to the Cisco Unified CallManager database.
  - Verify that you can place and receive internal and external calls on the Cisco Unified IP Phones.

Required Tasks

To integrate Cisco Unified MeetingPlace Express with Cisco Unified CallManager in an H.323 environment without a gatekeeper, complete the following tasks:

- Configuring Cisco Unified CallManager: Adding the Gateway and Route Pattern, page 5-2
- Configuring Cisco Unified MeetingPlace Express: Connecting to Cisco Unified CallManager, page 5-6

Configuring Cisco Unified CallManager: Adding the Gateway and Route Pattern

This topic describes how to identify Cisco Unified MeetingPlace Express as a gateway in the Cisco Unified CallManager configuration database. This topic also describes how to enable Cisco Unified CallManager to route calls to Cisco Unified MeetingPlace Express by associating a phone number with the gateway. This association is called a route pattern.

Before You Begin

- See the Prerequisites and other information in the "Configuring the Cisco Unified CallManager Integration Without a Gatekeeper" section on page 5-2.
- This task is performed in the Cisco Unified CallManager Administration pages. Because the pages and menus vary by Cisco Unified CallManager release, you may need to see the Cisco Unified CallManager Administration online help for more accurate step-by-step instructions than those provided in this procedure. The following procedure refers to Cisco Unified CallManager release 4.1.

Procedure

Step 1 Go to http://*ccm-server*/ccmadmin/main.asp, where *ccm-server* is the fully qualified domain name or IP address of the Cisco Unified CallManager server.

Step 2 Log in with your Cisco Unified CallManager administrator username and password.

Step 3 Add the gateway to the Cisco Unified CallManager database by completing the following actions:

a. In the menu at the top of the Cisco Unified CallManager Administration page, click **Device** > **Gateway**.

b. In the top right corner of the page, click **Add a New Gateway**.

c. Select the **H.323 Gateway** type.

d. Select the **H.225** device protocol.

e. Click **Next**.

f. In the Gateway Configuration page, configure the fields described in Table 5-1.

Table 5-1 Fields for Adding A New Gateway to Cisco Unified CallManager

Gateway Configuration Field	Action
Device Name	Enter the IP address or hostname for Port 1 (eth0) of the Cisco Unified MeetingPlace Express server. If you are using DNS, then enter the full server name with the qualifying domain, for example: mpx-server.example.com.
Description	Enter a short description to distinguish this gateway from other gateways to Cisco Unified CallManager.
Device Pool	If no device pools are defined, then select **Default**. If the Cisco Unified CallManager deployment utilizes customer-defined device pools, then either create a new device pool or choose an existing device pool for a region with a codec that is compatible with the conferencing gateway. Currently, Cisco Unified MeetingPlace Express supports only the G.711 audio codec. The device pool specifies a collection of properties for this device including Cisco Unified CallManager Group, Date/Time Group, and Region.

Table 5-1 Fields for Adding A New Gateway to Cisco Unified CallManager (continued)

Gateway Configuration Field	Action
Location	If no locations are defined, then keep the default value of **None**. If the Cisco Unified CallManager deployment utilizes customer-defined locations, then configure this field to avoid conflicts with QoS settings on the WAN. Either create a new location or choose an existing location that represents the conferencing gateway within the corporate WAN. The location specifies the total bandwidth that is available for calls to and from this location. A location setting of None means that the locations feature does not keep track of the bandwidth that this device consumes.
Signaling Port	Keep the default value of **1720**.
Media Termination Point Required	Uncheck this checkbox. Media Termination Point (MTP) is required only if you have non-G.711 endpoints joining your meetings. In such circumstances, make sure that the Media Resource Group List (MRGL) associated with the Cisco Unified MeetingPlace Express server has a transcoder and that the inter-region setting between the transcoder and the Cisco Unified MeetingPlace Express server is G.711.
Calling Search Space	If no calling search spaces (CSSs) are defined, then keep the default value of **None**. If the Cisco Unified CallManager deployment utilizes customer-defined CSSs, then either create a new CSS or choose an existing CSS that allows the conferencing gateway to dial any numbers that are required to join attendees in to conferences and to reach the Help Desk attendant. A CSS specifies the collection of Route Partitions that are searched to determine how a collected (originating) number should be routed. Use the CSS to prevent toll fraud by controlling which dial patterns may be dialed out from Cisco Unified MeetingPlace Express. For example, you can use CSS to block international calls.
AAR Calling Search Space	If no AAR CSSs are defined, then keep the default value of **None**. If the Cisco Unified CallManager deployment utilizes customer-defined CSSs, then choose the appropriate CSS for the device to use when it performs automated alternate routing (AAR). The AAR CSS specifies the collection of route partitions that are searched to determine how to route a collected (originating) number that is otherwise blocked due to insufficient bandwidth.

g. For all other required fields on the Gateway Configuration page, configure them appropriately for the current Cisco Unified CallManager deployment. For information about each field, see the Cisco Unified CallManager online help or see the administration guide for your specific Cisco Unified CallManager release.

h. Click **Insert**.

Step 4 Add the route pattern to the Cisco Unified CallManager database by completing the following actions:

a. In the menu at the top of the Cisco Unified CallManager Administration page, click **Route Plan** > **Route/Hunt** > **Route Pattern**.

b. In the top right corner, click **Add a New Route Pattern**.

c. In the Route Pattern Configuration page, configure the fields described in Table 5-2.

Table 5-2 Fields for Adding A New Route Pattern to Cisco Unified CallManager

Route Pattern Configuration Field	Action
Route Pattern	Enter the phone number that users will use to call in to Cisco Unified MeetingPlace Express. Do not enter any spaces in this field.
Gateway or Route List	Select the value that matches the Device Name you entered for the gateway in Step 3f.

d. Click **Insert** to add the route pattern to Cisco Unified MeetingPlace Express.

Step 5 If you have multiple Cisco Unified MeetingPlace Express access numbers, repeat Step 4 for each access number, including the following:

- Phone numbers entered on the Usage Configuration page—See the "Configuring Meeting Phone Numbers and Notification Labels" section on page 3-7.
- Direct meeting dial-in numbers—See the "Configuring Direct Meeting Dial-In" section on page 4-16.

Step 6 Proceed to the "Configuring Cisco Unified MeetingPlace Express: Connecting to Cisco Unified CallManager" section on page 5-6.

Related Topics

- About Integration With Cisco Unified CallManager, page 5-1
- Configuring the Cisco Unified CallManager Integration Without a Gatekeeper, page 5-2

Configuring Cisco Unified MeetingPlace Express: Connecting to Cisco Unified CallManager

This topic describes how to configure Cisco Unified MeetingPlace Express to connect directly to Cisco Unified CallManager in an H.323 environment.

Before You Begin

- See the Prerequisites and other information in the "Configuring the Cisco Unified CallManager Integration Without a Gatekeeper" section on page 5-2.
- Complete the task described in the "Configuring Cisco Unified CallManager: Adding the Gateway and Route Pattern" section on page 5-2.
- This task is completed in the Cisco Unified MeetingPlace Express Administration Center.

Procedure

Step 1 Log in to Cisco Unified MeetingPlace Express.

Step 2 Click **Administration** at the top of the page.

Step 3 On the left side of the page:

a. Click **System Configuration**.

b. Click **Call Configuration**.

c. Click **H.323 Configuration**.

Step 4 In the H.323 Configuration page, configure the fields in Table 5-3.

Table 5-3 Required Configuration for H.323 Configuration Page on Cisco Unified MeetingPlace Express for Cisco Unified CallManager Integration Without a Gatekeeper

H.323 Configuration Page Field	Required Value
H.323 enabled	Yes
Local H.323 port	1720 (default)
Use gatekeeper	No
H.323 gateway 1	IP address of the Cisco Unified CallManager server. If you have a cluster of Cisco Unified CallManager servers, then enter the IP address of the primary call-processing server in the cluster.
H.323 gateway 2 H.323 gateway 3 H.323 gateway 4 H.323 gateway 5	IP addresses of other Cisco Unified CallManager servers in the cluster that provide call-processing redundancy, if any. **Note** If the primary Cisco Unified CallManager server goes down, Cisco Unified MeetingPlace Express cannot complete dialed-out calls without a gatekeeper. These fields enable only incoming calls to be routed by the failover Cisco Unified CallManager servers.

Step 5 Click **Save**.

Step 6 On the left side of the page:

a. Click **System Configuration**.

b. Click **Call Configuration**.

c. Click **Dial Configuration**.

Step 7 In the Dial Configuration Page, configure the Outdials field to **H.323**.

Step 8 Click **Save**.

Step 9 Test this integration by placing a call from any phone to the phone number that is used to access the Cisco Unified MeetingPlace Express system. You should hear the "Welcome to Cisco Unified MeetingPlace Express" greeting.

Related Topics

- About Integration With Cisco Unified CallManager, page 5-1
- Configuring the Cisco Unified CallManager Integration Without a Gatekeeper, page 5-2
- About This Page: H.323 Configuration, page B-100
- About This Page: Dial Configuration, page B-53

About Cisco Unified IP Phone Services

A Cisco Unified CallManager feature, Cisco Unified IP Phone services comprise XML applications that enable the display of interactive content with text and graphics on certain models of Cisco Unified IP Phones. For a list of supported phone models, see the Cisco Unified IP Phone services section of the *Cisco Unified CallManager System Guide* for your release of Cisco Unified CallManager.

The Cisco Unified MeetingPlace Express service for the Cisco Unified IP Phone offers users a convenient way to join meetings, start reservationless meetings, view a list of upcoming meetings, and view meeting details. After joining a meeting, a user can perform in-meeting operations such as locking the meeting, recording the meeting, viewing a list of participants, and muting or ejecting participants.

Note The Cisco Unified MeetingPlace Express service for the Cisco Unified IP Phone is available only to Cisco Unified IP Phones that are registered to Cisco Unified CallManager.

Before you use the Cisco Unified MeetingPlace Express service for the Cisco Unified IP Phone, read the following sections:

- About the Security of the Cisco Unified MeetingPlace Express Service for the Cisco Unified IP Phone, page 5-8
- About Username and Password Requirements for the Cisco Unified MeetingPlace Express Service for the Cisco Unified IP Phone, page 5-8
- About Language Requirements for the Cisco Unified MeetingPlace Express Service for the Cisco Unified IP Phone, page 5-9

To configure this Cisco Unified IP Phone service, see the "Configuring the Cisco Unified MeetingPlace Express Service for the Cisco Unified IP Phone" section on page 5-10.

Related Topics

- About Integration With Cisco Unified CallManager, page 5-1

About the Security of the Cisco Unified MeetingPlace Express Service for the Cisco Unified IP Phone

Using the Cisco Unified MeetingPlace Express service for the Cisco Unified IP Phone may affect the way you secure your Cisco Unified MeetingPlace Express system or network:

- Once a Cisco Unified IP Phone is subscribed to the Cisco Unified MeetingPlace Express service for the Cisco Unified IP Phone, anyone can use that Cisco Unified IP Phone screen to *view* the meeting details and invitees of published meetings. To *join* a meeting through this Cisco Unified IP Phone service, however, the user is always prompted for his phone profile password (numeric PIN).
- Each time a Cisco Unified IP Phone accesses the Cisco Unified MeetingPlace Express service for the Cisco Unified IP Phone, the following items are sent as clear text over the network:
  - Username
  - PIN or password (See Table 5-4.)
  - Phone number of the Cisco Unified IP Phone

Related Topics

- About Cisco Unified IP Phone Services, page 5-7
- About Username and Password Requirements for the Cisco Unified MeetingPlace Express Service for the Cisco Unified IP Phone, page 5-8
- Configuring the Cisco Unified MeetingPlace Express Service for the Cisco Unified IP Phone, page 5-10

About Username and Password Requirements for the Cisco Unified MeetingPlace Express Service for the Cisco Unified IP Phone

The username and password required to subscribe to the Cisco Unified MeetingPlace Express service for the Cisco Unified IP Phone depends on how the user is authenticated when logging in to Cisco Unified MeetingPlace Express from a workstation. See Table 5-4.

Table 5-4 Required Username and Password for Subscribing to the Cisco Unified IP Phone Service

Cisco Unified MeetingPlace Express Authentication Method	Required Username for name Parameter	Required Password for wfpassword Parameter	Cisco Unified MeetingPlace Express Release
Locally by the Cisco Unified MeetingPlace Express database	User ID in Cisco Unified MeetingPlace Express user profile	User password in Cisco Unified MeetingPlace Express user profile	Release 1.1.1 and earlier releases
		Profile Password in Cisco Unified MeetingPlace Express user profile	Release 1.1.2 and later releases
Externally by Cisco Unified CallManager	Username in Cisco Unified CallManager	Password in Cisco Unified CallManager	Release 1.1.1 and earlier releases
		Numeric PIN in Cisco Unified CallManager	Release 1.1.2 and later releases

Table 5-4 Required Username and Password for Subscribing to the Cisco Unified IP Phone Service (continued)

Cisco Unified MeetingPlace Express Authentication Method	Required Username for name Parameter	Required Password for wfpassword Parameter	Cisco Unified MeetingPlace Express Release
Externally by Active Directory, Netscape Directory, or iPlanet Directory	Username in external directory	Password in external directory	Release 1.1.1 and earlier releases
		Numeric PIN in external directory	Release 1.1.2 and later releases

Related Topics

- Configuring the Cisco Unified MeetingPlace Express Service for the Cisco Unified IP Phone, page 5-10
- About Cisco Unified IP Phone Services, page 5-7
- About User Authentication By an External Directory, page 5-14
- About Integration With Cisco Unified CallManager, page 5-1

About Language Requirements for the Cisco Unified MeetingPlace Express Service for the Cisco Unified IP Phone

The requirements in this section apply when Cisco Unified MeetingPlace Express, Cisco Unified CallManager, or the Cisco Unified IP Phones are configured for multiple *locales*, which are language versions for specific regions. For example, U.S. English is English for the United States, and U.K. English is English for the United Kingdom. Although both versions use the English language, they are different locales.

- For each language enabled on Cisco Unified MeetingPlace Express, the matching locale must be installed on Cisco Unified CallManager. See the Cisco IP Telephony Locale Installer documentation.
- For each Cisco Unified IP Phone subscribed to the Cisco Unified MeetingPlace Express service, the User Locale specified in Cisco Unified CallManager must match the Language specified in the Cisco Unified MeetingPlace Express user profile.

Related Topics

- Configuring the Cisco Unified MeetingPlace Express Service for the Cisco Unified IP Phone, page 5-10
- About Cisco Unified IP Phone Services, page 5-7
- About Integration With Cisco Unified CallManager, page 5-1

Configuring the Cisco Unified MeetingPlace Express Service for the Cisco Unified IP Phone

This topic describes how to configure Cisco Unified CallManager to enable users to subscribe to and access the Cisco Unified MeetingPlace Express service for the Cisco Unified IP Phone.

Before You Begin

- Read the information in About Cisco Unified IP Phone Services, page 5-7.
- Configure Cisco Unified CallManager as the call-control device for Cisco Unified MeetingPlace Express. See the "About Integration With Cisco Unified CallManager" section on page 5-1.
- The Cisco Unified MeetingPlace Express service for the Cisco Unified IP Phone is available only to Cisco Unified IP Phones that are registered to Cisco Unified CallManager.
- This task is performed in the Cisco Unified CallManager Administration pages. Because the pages and menus vary by Cisco Unified CallManager release, you may need to see the Cisco Unified CallManager Administration online help for more accurate step-by-step instructions than those provided in this procedure. The following procedure refers to Cisco Unified CallManager release 4.1.

Procedure

Step 1 Go to http://*ccm-server*/ccmadmin/main.asp, where *ccm-server* is the fully qualified domain name or IP address of the Cisco Unified CallManager server.

Step 2 Log in with your Cisco Unified CallManager administrator username and password.

Step 3 Add a new Cisco Unified IP Phone service by completing the following actions:

a. In the menu at the top of the Cisco Unified CallManager Administration page, click **Feature > Cisco IP Phone Services**.

b. In the top right corner, click **Add a New IP Phone Service**.

c. In the Service Information area, configure the fields described in Table 5-5.

Table 5-5 Fields for the Cisco Unified MeetingPlace Express Service for the Cisco Unified IP Phone

Service Information Field	Action
Service Name Name of the service that appears on the Cisco Unified IP Phone and on the menu of available services on user subscription pages.	Enter a name, for example: Cisco Unified MeetingPlace Express. If you have more than one Cisco Unified MeetingPlace Express server, name the services appropriately so that users can distinguish among them.

Table 5-5 Fields for the Cisco Unified MeetingPlace Express Service for the Cisco Unified IP Phone (continued)

Service Information Field	Action
Service Description Description of what the service provides	Enter a brief description, for example: Integrated voice and web conferencing.
Service URL URL where the Cisco Unified MeetingPlace Express service application for the Cisco Unified IP Phone is located. For this service to be available, the phones in the Cisco Unified CallManager cluster must have network connectivity to this server.	Enter the URL in one of the following formats, where *server* is the hostname or IP address of the Cisco Unified MeetingPlace Express system: • If SSL is enabled on the system: **http://***server*/**MPAPI/ipphone/login?serverhost=***server* • If SSL is not enabled on the system: **http://***server***:8080**/**MPAPI/ipphone/login?serverhost=***server***:8080** **Note** This URL is case sensitive. Make sure that this server remains independent of the servers in your Cisco Unified CallManager cluster. Do not specify a Cisco Unified CallManager server or any server that is associated with Cisco Unified CallManager (such as a TFTP server or directory database publisher server).

d. Click **Insert** to add the Cisco Unified IP Phone service.

e. (Optional) To add another Cisco Unified IP Phone service, click **New** and repeat Steps 3c and 3d.

f. To finish, click **Insert and Close**.

g. Click **Update Subscriptions**.

Step 4 Find the Cisco Unified IP Phone service you just added by completing the following actions:

a. In the menu at the top of the Cisco Unified CallManager Administration page, click **Feature > Cisco IP Phone Services**.

b. In the Find and List IP Phone Services page, click **Find**.

c. In the Matching Records area, locate the Cisco Unified MeetingPlace Express service for the Cisco Unified IP Phone.

Step 5 Click the name of the Cisco Unified MeetingPlace Express service for the Cisco Unified IP Phone.

Step 6 Add the Cisco IP Phone Number parameter to this Cisco Unified IP Phone service by completing the following actions:

a. In the Service Parameter Information area, click **New**.

b. In the Configure Cisco IP Phone Service Parameter window, configure the fields described in Table 5-6.

Table 5-6 Fields for the Cisco IP Phone Number Parameter

Service Parameter Information Field	Action
Parameter Name Exact query string used to build the subscription URL.	Enter **ipphone**. This field is case-sensitive.
Parameter Display Name Descriptive parameter name displayed to the user on the Cisco IP Phone Users Options (ccmuser) website.	Enter **Cisco IP Phone Number**.
Parameter Description Description that the user can access while subscribing to the service. The parameter description should help users enter the correct value for the parameter.	Enter **Your Cisco IP Phone number for MeetingPlace Express to call you**.
Parameter is Required Specifies that this parameter is required for a Cisco Unified IP Phone to subscribe to the Cisco Unified MeetingPlace Express service.	Check **Parameter is Required**.

c. Click **Insert** to add the Cisco IP Phone Number parameter.

Step 7 Add the User Name parameter to this Cisco Unified IP Phone service by completing the following actions:

a. In the Service Parameter Information area, click **New**.

b. In the Configure Cisco IP Phone Service Parameter window, configure the fields described inTable 5-7.

Table 5-7 Fields for the User Name Parameter

Service Parameter Information Field	Action
Parameter Name Exact query string used to build the subscription URL.	Enter **name**. This field is case-sensitive.
Parameter Display Name Descriptive parameter name displayed to the user on the Cisco IP Phone Users Options (ccmuser) website.	Enter **User Name**.
Parameter Default Value	Enter **guest**.

Table 5-7 Fields for the User Name Parameter (continued)

Service Parameter Information Field	Action
Parameter Description Description to help users enter the correct username when they subscribe to the Cisco Unified MeetingPlace Express service for the Cisco Unified IP Phone.	Enter a helpful description. The username required depends on how the user is authenticated when logging into Cisco Unified MeetingPlace Express from a workstation. See the "About Username and Password Requirements for the Cisco Unified MeetingPlace Express Service for the Cisco Unified IP Phone" section on page 5-8.
Parameter is Required Specifies that this parameter is required for a Cisco Unified IP Phone to subscribe to the Cisco Unified MeetingPlace Express service.	Check **Parameter is Required**.

c. Click **Insert** to add the User Name parameter.

Step 8 Add the User Password parameter to this Cisco Unified IP Phone service by completing the following actions:

a. In the Service Parameter Information area, click **New**.

b. In the Configure Cisco IP Phone Service Parameter window, configure the fields described in Table 5-8:

Table 5-8 Fields for the User Password Parameter

Service Parameter Information Field	Action
Parameter Name Exact query string used to build the subscription URL.	Enter **wfpassword**. This field is case-sensitive.
Parameter Display Name Descriptive parameter name displayed to the user on the Cisco IP Phone Users Options (ccmuser) website.	Enter **User Password**. (Release 1.1.1 and earlier) or Enter **User PIN**. (Release 1.1.2 and later releases)
Parameter Description Description to help users enter the correct password when they subscribe to the Cisco Unified MeetingPlace Express service for the Cisco Unified IP Phone.	Enter a helpful description. The password required depends on how the user is authenticated when logging into Cisco Unified MeetingPlace Express from a workstation. See the "About Username and Password Requirements for the Cisco Unified MeetingPlace Express Service for the Cisco Unified IP Phone" section on page 5-8.

Table 5-8 Fields for the User Password Parameter (continued)

Service Parameter Information Field	Action
Parameter is Required Specifies that this parameter is required for a Cisco Unified IP Phone to subscribe to the Cisco Unified MeetingPlace Express service.	Check **Parameter is Required**.
Parameter is a Password (mask contents) Masks the password on the screen as the user enters it.	Check **Parameter is a Password (mask contents)**

c. Click **Insert and Close** to add the User Password parameter and close the Configure Cisco IP Phone Service Parameter dialog.

Step 9 To apply the Cisco Unified IP Phone service and parameter changes, take one of the following actions:

- If the service was modified after subscriptions existed, then click **Update Subscriptions** to rebuild all user subscriptions. You must update subscriptions if you changed the service URL, removed a phone service parameter, or changed the name for a phone service parameter.
- If the service is new and you do not need to rebuild user subscriptions, then click **Update**.

Step 10 Subscribe Cisco Unified IP Phones to the Cisco Unified MeetingPlace Express service by taking one or both of the following actions:

- You can subscribe individual Cisco Unified IP Phones to the Cisco Unified IP Phone service through Cisco Unified CallManager. See the *Cisco Unified CallManager Administration Guide*.
- Notify end users that they can subscribe their Cisco Unified IP Phones to the Cisco Unified MeetingPlace Express service. See the *User Guide for Cisco Unified MeetingPlace Express*.

Related Topics

- About Cisco Unified IP Phone Services, page 5-7
- About Integration With Cisco Unified CallManager, page 5-1
- Enabling SSL for the End-User Interface, Administration Center, and Web Conferencing, page 10-4

About User Authentication By an External Directory

Note User authentication by an external directory is only supported with Cisco Unified CallManager.

You can simplify user profile administration by enabling an external directory to authenticate Cisco Unified MeetingPlace Express users. Cisco Unified MeetingPlace Express automatically creates a user profile in the local database when a new user attempts to log in and successfully authenticates through an external directory.

Each user profile in Cisco Unified MeetingPlace Express includes an authentication method setting (local or external) that affects the following:

- How the user is authenticated in future attempts to access Cisco Unified MeetingPlace Express.
- Which user profile parameters may be modified by either the system administrator or the end user through Cisco Unified MeetingPlace Express.

The authentication method for a user cannot be configured through the Administration Center. The authentication method can be modified only within a user profile import file. Set the isLocalUser field to one of the following values:

- Yes—User is authenticated locally by the Cisco Unified MeetingPlace Express database. This is the default setting for user profiles that are imported or manually created through the Administration Center.
- No—User is authenticated by an external directory. This is the default setting for user profiles that are automatically created when new users successfully authenticate through an external directory.

See the following topics:

- Requirements for User Authentication By an External Directory, page 5-15
- Restrictions for User Authentication By an External Directory, page 5-16
- User Profile Settings When Authenticated By an External Directory, page 5-17

To configure user authentication by an external directory, see one of the following topics, depending on which release of Cisco Unified CallManager you use:

- Configuring User Authentication By an External Directory—Cisco Unified CallManager Release 4.x, page 5-17
- Configuring User Authentication By an External Directory—Cisco Unified CallManager Release 5.x, page 5-18

Related Topics

- About the Methods of Adding User Profiles, page 6-8
- About Integration With Cisco Unified CallManager, page 5-1

Requirements for User Authentication By an External Directory

Cisco Unified CallManager Release 4.0 or a later release is required to use an external directory to authenticate Cisco Unified MeetingPlace Express users.

Table 5-9 lists the supported authentication methods and directories.

Table 5-9 Supported Authentication Methods By an External Directory

Cisco Unified CallManager Release	Authentication Method	Supported Directories
4.x	LDAP[1]	• Cisco Unified CallManager DC-Directory[2] • Any LDAP directory with the installed Cisco Customer Directory Configuration Plugin for Cisco Unified CallManager[3]
5.x	AXL SOAP API[4]	• Cisco Unified CallManager user directory • Any LDAP directory that is synchronized with Cisco Unified CallManager[5]

1. LDAP = Lightweight Directory Access Protocol
2. DC-Directory = Data Connection Directory, embedded LDAP directory in Cisco Unified CallManager
3. Supported LDAP directories depend on the specific Cisco Unified CallManager release. See *Installing the Cisco Customer Directory Configuration Plugin for Cisco CallManager* for your specific release of Cisco Unified CallManager.
4. AXL SOAP API = Administrative XML Layer Simple Object Access Protocol Application Programming Interface
5. For information about synchronizing Cisco Unified CallManager with an LDAP directory, see the system guide and administration guide for your specific release of Cisco Unified CallManager.

Related Topics

- About User Authentication By an External Directory, page 5-14

Restrictions for User Authentication By an External Directory

- Cisco Unified MeetingPlace Express Release 1.1.1 and earlier releases do not support AXL SOAP API user authentication over the phone.
- The following restrictions apply for each user profile that is automatically created during authentication by an external directory, or configured as requiring external authentication during an import process:
  – The user is always authenticated through the external directory. Therefore, if the connection fails between Cisco Unified MeetingPlace Express and the external directory, the user will not be able to log in to Cisco Unified MeetingPlace Express.
  – Because the user is authenticated through the external directory, the User ID, User password, Profile Number, and Profile Password fields cannot be modified through Cisco Unified MeetingPlace Express by the user or by the system administrator.
  – These password-expiration fields on the Usage Configuration page do not apply to users that are authenticated by an external directory: Change profile password (days) and Change user password (days).

Related Topics

- About User Authentication By an External Directory, page 5-14

User Profile Settings When Authenticated By an External Directory

The following Cisco Unified MeetingPlace Express user profile fields are populated with information from the external directory:

- First name
- Last name
- User ID
- User password
- Profile Number—Unique number based on the user's phone number.
- Profile Password—Numeric PIN used to access Cisco Unified MeetingPlace Express by phone.
- E-mail address
- Search order for "Find Me"

If any of the listed fields are not available in the external directory, then the field is left blank in the Cisco Unified MeetingPlace Express user profile.

All other user profile fields are populated with the values configured in the guest profile. See the "About the Guest Profile and Guest Users" section on page 6-28.

Related Topics

- About User Authentication By an External Directory, page 5-14

Configuring User Authentication By an External Directory—Cisco Unified CallManager Release 4.x

This topic describes how to configure user authentication by an external directory that is either embedded in or integrated with Cisco Unified CallManager Release 4.x.

Before You Begin

- Read the "About User Authentication By an External Directory" section on page 5-14.
- If you plan to authenticate Cisco Unified MeetingPlace Express users against an LDAP directory that is separate from the DC-Directory embedded in Cisco Unified CallManager, then complete the required tasks in *Installing the Cisco Customer Directory Configuration Plugin* for your specific Cisco Unified CallManager release.

Procedure

Step 1 Log in to Cisco Unified MeetingPlace Express.

Step 2 Click **Administration** at the top of the page.

Step 3 On the left side of the page:

a. Click **System Configuration**.

b. Click **Usage Configuration**.

Step 4 Configure the following fields, the full descriptions and examples of which are provided in the "About This Page: Usage Configuration" section on page B-188:

- Cisco CallManager version—Set this field to Cisco Unified CallManager Release 4.x.
- LDAP URL—Set this field as follows:
  - Make sure that this URL starts with ldap, not http. For example, **ldap:**//*server-ip-address:port*/
  - Make sure that there are no spaces after the URL.
- Directory username—Use the format of an LDAP distinguished name, for example: cn=Directory Manager, o=cisco.com
- Password—Use the password that was specified during Cisco Unified CallManager installation.
- Cisco base—Leave blank if you are not using the Cisco Unified CallManager DC-Directory to authenticate Cisco Unified MeetingPlace Express users.
- User base
- Directory type

Step 5 Click **Save**.

Related Topics

- About User Authentication By an External Directory, page 5-14
- About This Page: Usage Configuration, page B-188
- About the Methods of Adding User Profiles, page 6-8

Configuring User Authentication By an External Directory—Cisco Unified CallManager Release 5.x

To configure user authentication by an external directory that is either embedded in or integrated with Cisco Unified CallManager 5.x, complete both of the following tasks:

1. Configuring Cisco Unified CallManager to Support Authentication of Cisco Unified MeetingPlace Express Users, page 5-18
2. Configuring Cisco Unified MeetingPlace Express for External User Authentication By Cisco Unified CallManager 5.x, page 5-20

Configuring Cisco Unified CallManager to Support Authentication of Cisco Unified MeetingPlace Express Users

This topic describes how to create an application user in Cisco Unified CallManager 5.x that enables Cisco Unified MeetingPlace Express users to be authenticated by one of the following directories:

- User directory in Cisco Unified CallManager 5.x
- Any LDAP directory that is synchronized with Cisco Unified CallManager 5.x

For information about synchronizing Cisco Unified CallManager with an LDAP directory, see the system guide and administration guide for your specific release of Cisco Unified CallManager.

Before You Begin

- Read the "About User Authentication By an External Directory" section on page 5-14.
- Make sure that you enable the AXL web service on the Cisco Unified CallManager so that other applications, such as Cisco Unified MeetingPlace Express, can access it. See the administration documentation for your release of Cisco Unified CallManager.

Procedure

Step 1 Go to http://*ccm-server*/ccmadmin/main.asp, where *ccm-server* is the fully qualified domain name or IP address of the Cisco Unified CallManager server.

Step 2 Log in with your Cisco Unified CallManager administrator username and password.

Step 3 Create a Cisco Unified MeetingPlace Express LDAP application user by following these steps:

a. Select **User Management** from the main menu.

b. Select **Application User** from the drop-down list.

c. Enter a username for the new application user, such as mpeaxl, and assign a password.

Associate the new application user to the user group that has permission to access the Cisco Unified CallManager AXL database, by configuring the following:

Step 4 Create a role for AXL users by following these steps:

a. Select **User Management** from the main menu.

b. Select **Role Configuration** from the drop-down list.

c. In the Name field, enter “Standard AXL API Access.”

d. Under Resource Access Information, next to AXL Database API, make sure that checkbox next to Allow to use API is checked. This allows AXL database access.

Step 5 Create a user group by following these steps:

a. Select **User Management** from the main menu.

b. Select **User Group** from the drop-down list.

c. In the Name field, enter “Standard AXL Users.”

Note This user group may already exist. If it does, proceed to Step 6.

d. Click **Save** to create the group.

Step 6 Add application users to the group by following these steps:

a. Select **User Management** from the main menu.

b. Select **User Group Configuration** from the drop-down list.

c. Click **Add Application Users to Group**.

d. Check the checkbox next to the name of the application user you created in Step 3c.

e. Click **Add Selected**.

f. Click **Save**.

Step 7 Assign a role to the user group by following these steps:

a. Select **User Management** from the main menu.

b. Select **User Group** from the drop-down list.

c. Click the Role Information icon next to the Standard AXL API Users created in step 4b.

d. Click **Assign Role to Group**.

e. Select Standard AXL API Access and then click **Add Selected**.

f. Click **Save**.

Related Topics

- Configuring User Authentication By an External Directory—Cisco Unified CallManager Release 5.x, page 5-18
- About User Authentication By an External Directory, page 5-14
- About the Methods of Adding User Profiles, page 6-8

Configuring Cisco Unified MeetingPlace Express for External User Authentication By Cisco Unified CallManager 5.x

This topic describes how to configure Cisco Unified MeetingPlace Express to authenticate users through one of the following directories:

- User directory in Cisco Unified CallManager 5.x
- Any LDAP directory that is synchronized with Cisco Unified CallManager 5.x

For information about synchronizing Cisco Unified CallManager with an LDAP directory, see the system guide and administration guide for your specific release of Cisco Unified CallManager.

Before You Begin

- Read the "About User Authentication By an External Directory" section on page 5-14.
- Complete the task described in the "Configuring Cisco Unified MeetingPlace Express for External User Authentication By Cisco Unified CallManager 5.x" section on page 5-20.

Procedure

Step 1 Log in to Cisco Unified MeetingPlace Express.

Step 2 Click **Administration** at the top of the page.

Step 3 On the left side of the page:

a. Click **System Configuration**.

b. Click **Usage Configuration**.

Step 4 Configure the following fields, the descriptions and examples of which are provided in the "About This Page: Usage Configuration" section on page B-188:

- Cisco CallManager version—Set this field to Cisco Unified CallManager release 5.x.
- AXL username—Username for the Cisco Unified MeetingPlace Express application user that you configured in Cisco Unified CallManager.

  See Step 3 in the "Configuring Cisco Unified CallManager to Support Authentication of Cisco Unified MeetingPlace Express Users" section on page 5-18.
- AXL password—Password for the Cisco Unified MeetingPlace Express application user that you configured in Cisco Unified CallManager.
- See Step 3 in the "Configuring Cisco Unified CallManager to Support Authentication of Cisco Unified MeetingPlace Express Users" section on page 5-18.

Step 5 In the New AXL URL field:

a. Enter the URL or hostname of the AXL directory server.

b. Click **Add**.

Step 6 Verify that the URL or hostname correctly appears in the AXL URL field.

Step 7 Click **Save**.

Step 8 Proceed to the "Configuring Cisco Unified CallManager to Support Authentication of Cisco Unified MeetingPlace Express Users" section on page 5-18.

Related Topics

- Configuring User Authentication By an External Directory—Cisco Unified CallManager Release 5.x, page 5-18
- About User Authentication By an External Directory, page 5-14
- About This Page: Usage Configuration, page B-188
- About the Methods of Adding User Profiles, page 6-8

About Integration with Cisco Unified CallManager Express and Other Standards-Based H.323 Call Control Devices

Besides Cisco Unified CallManager, Cisco Unified MeetingPlace Express also supports integration with other standards-based H.323 call-control devices such as the following:

- Cisco Unified CallManager Express

  Cisco Unified CallManager Express is suitable for entry-level Cisco Unified MeetingPlace Express systems that support up to 200 ports. For Cisco Unified MeetingPlace Express systems that support a larger user base, we recommend Cisco Unified CallManager instead of Cisco Unified CallManager Express for call-control. See the "About Integration With Cisco Unified CallManager" section on page 5-1.
- Cisco IOS software voice-enabled routers
- Third-party standards-based H.323 call-control devices

This topic describes one method of integrating Cisco Unified MeetingPlace Express with these standards-based H.323 call-control devices. In this method, Cisco Unified MeetingPlace Express is configured as a gateway to the call-control device. No gatekeeper is used in this setup.

See the following sections:

- Prerequisites for Integrating with Cisco Unified CallManager Express and Other Standards-Based H.323 Call Control Devices, page 5-22
- Required Tasks for Integrating with Cisco Unified CallManager Express and Other Standards-Based H.323 Call Control Devices, page 5-22

Prerequisites for Integrating with Cisco Unified CallManager Express and Other Standards-Based H.323 Call Control Devices

- Verify that the versions of your call-control device and Cisco Unified MeetingPlace Express are compatible. See the *Release Notes for Cisco Unified MeetingPlace Express*.
- Verify that your IP telephony network is already set up and working properly. For example:
  - Verify that the Cisco Unified IP Phones are properly connected and added to the database of your call-control device.
  - Verify that you can place and receive internal and external calls on the Cisco Unified IP Phones.

Related Topics

- About Integration with Cisco Unified CallManager Express and Other Standards-Based H.323 Call Control Devices, page 5-21

Required Tasks for Integrating with Cisco Unified CallManager Express and Other Standards-Based H.323 Call Control Devices

The tasks you need to complete depends on the type of call-control device you are using. See Table 5-10 for a task roadmap.

Table 5-10 Roadmap to Configuring Cisco Unified MeetingPlace Express With a Standards-Based Call-Control Device in an H.323 Environment

Task	Reference
1. Configure your particular call-control device.	• If you are configuring Cisco Unified CallManager Express or a Cisco IOS software voice-enabled router, complete the tasks described in the "Configuring a Cisco Call-Control Device in an H.323 Environment" section on page 5-23. • If you are using a third-party standards-based H.323 call-control device, see the product documentation that came with your device for configuration instructions.
2. Configure Cisco Unified MeetingPlace Express to connect to your call-control device.	See the "Configuring Cisco Unified MeetingPlace Express: Connecting to a Standards-Based H.323 Call-Control Device" section on page 5-26.

Related Topics

- About Integration with Cisco Unified CallManager Express and Other Standards-Based H.323 Call Control Devices, page 5-21

Configuring a Cisco Call-Control Device in an H.323 Environment

This topic describes how to add Cisco Unified MeetingPlace Express as an H.323 gateway to a Cisco call-control device other than Cisco Unified CallManager. This includes Cisco Unified CallManager Express and Cisco IOS software voice-enabled routers.

This topic consists of two tasks:

- Configuring a Cisco Call-Control Device in an H.323 Environment: Adding the Gateway, page 5-23
- Configuring a Cisco Call-Control Device in an H.323 Environment: Configuring the Dial Peer, page 5-24

Configuring a Cisco Call-Control Device in an H.323 Environment: Adding the Gateway

Before You Begin

- Read the following sections:
  - About Integration with Cisco Unified CallManager Express and Other Standards-Based H.323 Call Control Devices, page 5-21
  - Prerequisites for Integrating with Cisco Unified CallManager Express and Other Standards-Based H.323 Call Control Devices, page 5-22
- This task is performed in the Cisco IOS command-line interface (CLI) of the router. For more information about the Cisco IOS commands used in this procedure, see the *Cisco IOS Commands Master List* for your Cisco IOS software major release.

Procedure

Step 1 On the Cisco router, enter privileged EXEC mode or any other security level set by a system administrator. Enter your password if prompted.

```
Router# enable
```

Step 2 Enter global configuration mode.

```
Router# configure terminal
```

Step 3 Enter interface configuration mode.

```
Router(config)# interface type number
```

Step 4 Configure the IP address and subnet mask used by this gateway.

```
Router(config-if)# ip address [ip-address] [subnet-mask]
```

Step 5 (Optional) If you configured a Fast Ethernet interface, configure duplex operation as auto, which specifies the autonegotiation capability. The gateway automatically operates at half or full duplex depending on environmental factors, such as the type of media and transmission speeds for the peer routers, hubs, and switches used in the network configuration.

```
Router(config-if)# duplex {full | half | auto}
```

Step 6 (Optional) If you configured a Fast Ethernet interface, configure the speed for this gateway.

```
Router(config-if)# speed {10 | 100 | auto}
```

Step 7 Set the source IP address to be used for this gateway. This command binds all H.323 messages from the gateway to this IP address.

```
Router(config-if)# h323-gateway voip bind srcaddr [ip-address]
```

Step 8 Exit the current mode.

```
Router(config-if)# exit
```

Step 9 Proceed to the "Configuring a Cisco Call-Control Device in an H.323 Environment: Configuring the Dial Peer" section on page 5-24.

Example

The following example displays an H.323 gateway configuration with an IP address of 10.10.10.1. Both duplex operation and interface speed are configured for autonegotiation and all H.323 messages are bound to this IP address.

```
!
interface FastEthernet0/0
 ip address 10.10.10.1 255.255.255.0
 duplex auto
 speed auto
 h323-gateway voip bind srcaddr 10.10.10.1
```

Related Topics

- Required Tasks for Integrating with Cisco Unified CallManager Express and Other Standards-Based H.323 Call Control Devices, page 5-22
- About Integration with Cisco Unified CallManager Express and Other Standards-Based H.323 Call Control Devices, page 5-21

Configuring a Cisco Call-Control Device in an H.323 Environment: Configuring the Dial Peer

This topic describes how to enable your call-control device to route calls to Cisco Unified MeetingPlace Express by configuring a dial peer. Configuring dial peers is the key to implementing dial plans and providing voice services over an IP packet network. Dial peers are used to identify call source and destination endpoints and to define the characteristics applied to each call leg in the call connection.

Before You Begin

- Read the following sections:
  - About Integration with Cisco Unified CallManager Express and Other Standards-Based H.323 Call Control Devices, page 5-21
  - Prerequisites for Integrating with Cisco Unified CallManager Express and Other Standards-Based H.323 Call Control Devices, page 5-22
- Complete the task described in the "Configuring a Cisco Call-Control Device in an H.323 Environment: Adding the Gateway" section on page 5-23.
- This task is performed in the Cisco IOS command-line interface (CLI) of the router. For more information about the Cisco IOS commands used in this procedure, see the *Cisco IOS Commands Master List* for your Cisco IOS software major release.

Procedure

Step 1 On the Cisco router, enter privileged EXEC mode or any other security level set by a system administrator. Enter your password if prompted.

```
Router# enable
```

Step 2 Enter global configuration mode.

```
Router# configure terminal
```

Step 3 Enter dial peer voice configuration mode and define a remote voice over IP (VoIP) dial peer.

```
Router(config)# dial-peer voice number voip
```

- *number*—is one or more digits that identify the dial peer. Valid entries are from 1 to 2147483647.
- **voip**—indicates a VoIP peer that uses voice encapsulation on the IP network.

Step 4 (Optional) Provide a comment or a description to help you remember what is attached to this interface.

```
Router(config-dialpeer)# description string
```

Step 5 Route calls to the Cisco Unified MeetingPlace Express server.

```
Router(config-dialpeer)# destination-pattern digits
```

- *digits*—indicates the numbers that match the destination pattern.

Step 6 Configure the IP address of the Cisco Unified MeetingPlace Express server.

```
Router(config-dialpeer)# session target ipv4:ip-address
```

Step 7 Configure the router to use dual tone multifrequency (DTMF) relay to transport DTMF digits.

```
Router(config-dialpeer)# dtmf-relay h245-alphanumeric
```

Step 8 Configure the router to use a particular codec.

```
Router(config-dialpeer)# codec [g711ulaw | g711alaw]
```

Step 9 Disable voice activity detection (VAD) for the calls using this dial peer.

```
Router(config-dialpeer)# [no] vad
```

Step 10 Exit the current mode.

```
Router(config-dialpeer)# exit
```

Step 11 Proceed to the "Configuring Cisco Unified MeetingPlace Express: Connecting to a Standards-Based H.323 Call-Control Device" section on page 5-26.

Example

The following example displays dial peers that were configured to direct calls to a primary Cisco Unified MeetingPlace Express number and an alternate Cisco Unified MeetingPlace Express number. The Cisco Unified MeetingPlace Express IP address is configured as 10.10.10.4.

```
!
dial-peer voice 1 voip
 description MP Express main number
 destination-pattern 7777
 session target ipv4:10.10.10.4
 dtmf-relay h245-alphanumeric
```

```
 codec g711ulaw
 no vad
!
dial-peer voice 2 voip
 description MP Express alternate number
 destination-pattern 7000
 session target ipv4:10.10.10.4
 dtmf-relay h245-alphanumeric
 codec g711ulaw
 no vad
!
```

Related Topics

- Required Tasks for Integrating with Cisco Unified CallManager Express and Other Standards-Based H.323 Call Control Devices, page 5-22
- About Integration with Cisco Unified CallManager Express and Other Standards-Based H.323 Call Control Devices, page 5-21

Configuring Cisco Unified MeetingPlace Express: Connecting to a Standards-Based H.323 Call-Control Device

This topic describes how to configure Cisco Unified MeetingPlace Express to connect directly to a call-control device in an H.323 environment. This topic supports the following call-control devices:

- Cisco Unified CallManager Express
- Cisco IOS software voice-enabled router
- Third-party standards-based H.323 call-control devices

Before You Begin

- See the Prerequisites for Integrating with Cisco Unified CallManager Express and Other Standards-Based H.323 Call Control Devices and other information in the "About Integration with Cisco Unified CallManager Express and Other Standards-Based H.323 Call Control Devices" section on page 5-21.
- If you are integrating with Cisco Unified CallManager Express or a Cisco IOS software voice-enabled router, complete the tasks described in the "Configuring a Cisco Call-Control Device in an H.323 Environment" section on page 5-23.
- If you are using a third-party standards-based H.323 call-control device, configure your device as outlined in your product documentation.

Procedure

Step 1 Log in to Cisco Unified MeetingPlace Express.

Step 2 Click **Administration** at the top of the page.

Step 3 On the left side of the page:

a. Click **System Configuration**.

b. Click **Call Configuration**.

c. Click **H.323 Configuration**.

Step 4 In the H.323 Configuration page, configure the fields in Table 5-11.

Table 5-11 Required Configuration for H.323 Configuration Page on Cisco Unified MeetingPlace Express for Integration With a Call-Control Device Without a Gatekeeper

H.323 Configuration Page Field	Required Value
H.323 enabled	Yes
Local H.323 port	1720 (default)
Use gatekeeper	No
H.323 gateway 1	IP address of your call-control device. If you have a cluster of call-control devices, then enter the IP address of the primary call-processing device in the cluster.
H.323 gateway 2 H.323 gateway 3 H.323 gateway 4 H.323 gateway 5	IP addresses of other call-control devices in the cluster, if any. **Note** If the primary call-control device goes down, Cisco Unified MeetingPlace Express cannot complete dialed-out calls without a gatekeeper. These fields enable only incoming calls to be routed by the failover call-control devices.

Step 5 Click **Save**.

Step 6 On the left side of the page:

a. Click **System Configuration**.

b. Click **Call Configuration**.

c. Click **Dial Configuration**.

Step 7 In the Dial Configuration Page, configure the Outdials field to **H.323**.

Step 8 Click **Save**.

Step 9 Test this integration by placing a call from any phone to the phone number that is used to access the Cisco Unified MeetingPlace Express system. You should hear the “Welcome to Cisco Unified MeetingPlace Express” greeting.

Related Topics

- About Integration with Cisco Unified CallManager Express and Other Standards-Based H.323 Call Control Devices, page 5-21
- About This Page: H.323 Configuration, page B-100
- About This Page: Dial Configuration, page B-53

About Integration With a Call-Control Device Through a Gatekeeper

This topic describes the preferred method of integrating Cisco Unified MeetingPlace Express with a call-control device in an H.323 environment. This method involves using an external gatekeeper device to perform call admission control (CAC), bandwidth allocation, and dial pattern resolution, and to maintain a registry of devices in the multimedia network.

If you have a cluster of call-control devices that provide call-processing redundancy, then a gatekeeper enables Cisco Unified MeetingPlace Express to complete dialed-out calls even if the primary call-control device goes down.

Cisco Unified MeetingPlace Express supports integration with the following call-control devices through a gatekeeper:

- Cisco Unified CallManager
- Cisco Unified CallManager Express
- Cisco IOS software voice-enabled router

See the following sections:

- Prerequisites for Integrating With a Call-Control Device Through a Gatekeeper, page 5-28
- Required Tasks for Integrating With a Call-Control Device Through a Gatekeeper, page 5-28

Prerequisites for Integrating With a Call-Control Device Through a Gatekeeper

- The gatekeeper must be a Cisco router with Cisco IOS software that supports the Gatekeeper or Multimedia Conference Manager feature.
- Verify that your Cisco Unified MeetingPlace Express and call-control device versions are compatible. See the *Release Notes for Cisco Unified MeetingPlace Express*.
- Verify that your IP telephony network is working properly. For example, verify that the Cisco Unified IP Phones are connected and added to the database of your call control application. Also verify that you can place and receive internal and external calls on the Cisco Unified IP Phones.

Related Topics

- About Integration With a Call-Control Device Through a Gatekeeper, page 5-28

Required Tasks for Integrating With a Call-Control Device Through a Gatekeeper

The tasks you need to complete depend on the type of call-control device you are using. See Table 5-12 for a task roadmap.

Table 5-12 Roadmap to Configuring Cisco Unified MeetingPlace Express With a Call-Control Device Through a Gatekeeper

Task	Reference
1. (Optional) If your gatekeeper is not yet configured or defined in your call-control device, then configure your gatekeeper.	See the "Configuring the Gatekeeper: Identifying the Hostname, Zone, and Technology Prefix" section on page 5-29.
2. Configure your particular call-control device.	• If you are integrating with Cisco Unified CallManager, see the "Configuring Cisco Unified CallManager: Adding the Gatekeeper, Trunk, and Route Pattern" section on page 5-32 • If you are integrating with Cisco Unified CallManager Express or a Cisco IOS software voice-enabled router, see the "Configuring Cisco Unified CallManager Express and Other Standards-Based H.323 Call-Control Devices" section on page 5-34
3. Configure Cisco Unified MeetingPlace Express to integrate with your chosen call-control device.	See the "Configuring Cisco Unified MeetingPlace Express: Adding the Gatekeeper" section on page 5-38.

Related Topics

- About Integration With a Call-Control Device Through a Gatekeeper, page 5-28

Configuring the Gatekeeper: Identifying the Hostname, Zone, and Technology Prefix

This topic describes how to configure the gatekeeper to integrate Cisco Unified MeetingPlace Express with a call-control device.

Before You Begin

- Read the following sections:
  - About Integration With a Call-Control Device Through a Gatekeeper, page 5-28
  - Prerequisites for Integrating With a Call-Control Device Through a Gatekeeper, page 5-28
- If your gatekeeper is already configured with a hostname, zone, and technology prefix, then do not perform this task. Instead, take one of the following actions:
  - If you are integrating with Cisco Unified CallManager, proceed to the "Configuring Cisco Unified CallManager: Adding the Gatekeeper, Trunk, and Route Pattern" section on page 5-32.
  - If you are integrating with Cisco Unified CallManager Express or a Cisco IOS software voice-enabled router, proceed to the "Configuring Cisco Unified CallManager Express and Other Standards-Based H.323 Call-Control Devices" section on page 5-34.
- This task is performed in the Cisco IOS command-line interface (CLI) of the gatekeeper.

Procedure

Step 1 On the Cisco router, enter privileged EXEC mode or any other security level set by a system administrator. Enter your password if prompted.

```
Router# enable
```

Step 2 Enter global configuration mode.

```
Router# configure terminal
```

Step 3 Enter interface configuration mode.

```
Router(config)# interface type number
```

Step 4 Configure the IP address and subnet mask used by this gateway.

```
Router(config-if)# ip address [ip-address] [subnet-mask]
```

Step 5 (Optional) If you configured a Fast Ethernet interface, configure duplex operation as auto, which specifies the autonegotiation capability. The gateway automatically operates at half or full duplex depending on environmental factors, such as the type of media and transmission speeds for the peer routers, hubs, and switches used in the network configuration.

```
Router(config-if)# duplex auto
```

Step 6 (Optional) If you configured a Fast Ethernet interface, configure the speed for this gateway.

```
Router(config-if)# speed auto
```

Step 7 Configure this interface as an H.323 gateway interface.

```
Router(config-if)# h323-gateway voip interface
```

Step 8 Define the name and location of the gatekeeper for this gateway.

```
Router(config-if)# h323-gateway voip id [hostname] ipaddr [ip-address]
```

Step 9 Configure the H.323 name of the gateway identifying this gateway to its associated gatekeeper.

```
Router(config-if)# h323-gateway voip h323-id hostname
```

Step 10 Define the technology prefix that the gateway will register with the gatekeeper.

```
Router(config-if)# h323-gateway voip tech-prefix prefix
```

- *prefix*—Defines the numbers used as the technology prefixes. Each technology prefix can contain up to 11 characters. Although not strictly necessary, a pound (#) symbol is frequently used as the last digit in a technology prefix. Valid characters are 0 though 9, the pound (#) symbol, and the asterisk (*).

Step 11 Exit the current mode.

```
Router(config-if)# exit
```

Step 12 Enter gatekeeper configuration mode.

```
Router(config)# gatekeeper
```

Step 13 Configure the zone name and the name of the domain served by this gatekeeper.

```
Router(config-gk)# zone local zone-name domain-name [ip-address] [port]
```

- Optionally, you may specify the IP address of one of the interfaces on the gatekeeper. When the gatekeeper responds to gatekeeper discovery messages, it signals the endpoint or gateway to use this address in future communications.
- Set the gatekeeper port for Registration, Admission, and Status (RAS) signaling to 1719. This is the default.

Step 14 Configure the technology prefix, also called the gateway-type prefix.

```
Router(config-gk)# gw-type-prefix type-prefix
```

Step 15 End your configuration session by exiting to privileged EXEC mode.

```
Router(config-gk)# end
```

Step 16 Proceed to the next step in your integration configuration:

- If you are integrating with Cisco Unified CallManager, proceed to the "Configuring Cisco Unified CallManager: Adding the Gatekeeper, Trunk, and Route Pattern" section on page 5-32.
- If you are integrating with Cisco Unified CallManager Express or a Cisco IOS software voice-enabled router, proceed to the "Configuring Cisco Unified CallManager Express and Other Standards-Based H.323 Call-Control Devices" section on page 5-34.

Example

In the following Cisco IOS configuration example, the gatekeeper router hostname gk-1.example.com is also used as the zone name. Matching these names is not necessary but simplifies administration.

```
!
interface FastEthernet0/0
 ip address 1.1.100.200 255.255.0.0
 duplex auto
 speed auto
 h323-gateway voip interface
 h323-gateway voip id gk-1 ipaddr 10.1.100.200 1719
 h323-gateway voip h323-id ipipgw_core
 h323-gateway voip tech-prefix 1#
!
gatekeeper
 zone local gk-1.example.com 1.1.100.200
 gw-type-prefix 1#* default-technology
!
```

Tips

- For more information about the Cisco IOS commands used in this procedure, see the *Cisco IOS Commands Master List* for your Cisco IOS software major release.
- For conceptual information about gatekeepers, zones, and technology prefixes, see the *Cisco IOS H.323 Configuration Guide* for your Cisco IOS software major release.

Related Topics

- Required Tasks for Integrating With a Call-Control Device Through a Gatekeeper, page 5-28
- About Integration With a Call-Control Device Through a Gatekeeper, page 5-28

Configuring Cisco Unified CallManager: Adding the Gatekeeper, Trunk, and Route Pattern

This topic describes how to configure Cisco Unified CallManager in an H.323 environment to integrate with Cisco Unified MeetingPlace Express through a gatekeeper.

Before You Begin

- If the gatekeeper is already defined in Cisco Unified CallManager, then do not perform this task. Proceed to the "Configuring Cisco Unified MeetingPlace Express: Adding the Gatekeeper" section on page 5-38.
- See the Prerequisites for Integrating With a Call-Control Device Through a Gatekeeper and other information in the "About Integration With a Call-Control Device Through a Gatekeeper" section on page 5-28.
- Complete the task described in the "Configuring the Gatekeeper: Identifying the Hostname, Zone, and Technology Prefix" section on page 5-29.
- This task is performed in the Cisco Unified CallManager Administration pages. Because the pages and menus vary by Cisco Unified CallManager release, you may need to see the Cisco Unified CallManager Administration online help for more accurate step-by-step instructions than those provided in this procedure. The following procedure refers to Cisco Unified CallManager release 4.1.

Procedure

Step 1 Go to http://*ccm-server*/ccmadmin/main.asp, where *ccm-server* is the fully qualified domain name or IP address of the Cisco Unified CallManager server.

Step 2 Log in with your Cisco Unified CallManager administrator username and password.

Step 3 Add the gatekeeper to the Cisco Unified CallManager database by completing the following actions:

a. In the menu at the top of the Cisco Unified CallManager Administration page, click **Device** > **Gatekeeper**.

b. In the top right corner, click **Add a New Gatekeeper**.

c. In the Gatekeeper Configuration page, configure the fields described in Table 5-13.

Table 5-13 Fields for Adding A New Gatekeeper to Cisco Unified CallManager

Gatekeeper Configuration Field	Action
Host Name/IP Address IP address or hostname of the gatekeeper.	Enter one of the following values: • Hostname of the gatekeeper • IP address of the gatekeeper, as entered in Step 13 in the "Configuring the Gatekeeper: Identifying the Hostname, Zone, and Technology Prefix" section on page 5-29.
Enable Device Enables the registration of this gatekeeper with Cisco Unified CallManager.	Make sure that this checkbox is checked.

d. Click **Insert** to add the new gatekeeper to the Cisco Unified CallManager database.

e. Click **Reset Gatekeeper** to have the changes take effect.

Step 4 Add a new trunk to the gatekeeper by completing the following actions:

a. In the menu at the top of the Cisco Unified CallManager Administration page, click **Device > Trunk**.

b. In the top right corner, click **Add a New Trunk**.

c. In the Trunk type field, select **H.225 Trunk (Gatekeeper Controlled)**.

d. In the Device Protocol field, select **H.225**.

e. Click **Next**.

f. In the Trunk Configuration page, configure the fields described in Table 5-14.

Table 5-14 Fields for Adding A New Trunk to Cisco Unified CallManager

Trunk Configuration Field	Action
Device Information	
Device Name	Enter a unique identifier for this trunk, such as the name or IP address of the Cisco Unified MeetingPlace Express server.
Gatekeeper Information	
Gatekeeper Name	Select the name or IP address you entered in the Host Name/IP Address field in Step 3c.
Terminal Type	Select **Gateway**.
Technology Prefix	Enter the same value you entered for the *type-prefix* argument in Step 14 of the "Configuring the Gatekeeper: Identifying the Hostname, Zone, and Technology Prefix" section on page 5-29.
Zone	Enter the same value you entered for the *zone-name* argument in Step 13 of the "Configuring the Gatekeeper: Identifying the Hostname, Zone, and Technology Prefix" section on page 5-29.

g. For all other required fields on the Trunk Configuration page, configure the fields appropriately for the current Cisco Unified CallManager deployment. For information about each field, see the Cisco Unified CallManager online help or see the administration guide for your specific Cisco Unified CallManager release.

h. Click **Insert** to add the new trunk to the gatekeeper.

i. Click **Reset Trunk** to have the changes take effect.

Step 5 Add a new route pattern to Cisco Unified MeetingPlace Express through the gatekeeper by completing the following actions:

a. In the menu at the top of the Cisco Unified CallManager Administration page, click **Route Plan** > **Route/Hunt > Route Pattern**.

b. In the top right corner, click **Add a New Route Pattern**.

c. In the Route Pattern Configuration page, configure the fields described in Table 5-15.

Table 5-15 Fields for Adding A New Route Pattern to Cisco Unified CallManager

Route Pattern Configuration Field	Action
Route Pattern	Enter the phone number that users will use to call in to Cisco Unified MeetingPlace Express. Do not enter any spaces in this field.
Gateway or Route List	Select the value that matches the Device Name you entered for the trunk in Step 4f.

d. Click **Insert** to add the route pattern to the gatekeeper.

Step 6 (Optional) If you have multiple Cisco Unified MeetingPlace Express access numbers, repeat Step 7 for each access number. This includes phone numbers entered on the Usage Configuration page as well as direct meeting dial-in numbers.

Step 7 Proceed to the "Configuring Cisco Unified MeetingPlace Express: Adding the Gatekeeper" section on page 5-38.

Related Topics

- Required Tasks for Integrating With a Call-Control Device Through a Gatekeeper, page 5-28
- About Integration With a Call-Control Device Through a Gatekeeper, page 5-28

Configuring Cisco Unified CallManager Express and Other Standards-Based H.323 Call-Control Devices

This topic describes how to configure Cisco Unified CallManager Express and other Cisco IOS software voice-enabled routers to integrate with Cisco Unified MeetingPlace Express through a gatekeeper.

This topic is divided into two tasks:

- Configuring Cisco Unified CallManager Express and Other Standards-Based H.323 Call-Control Devices: Adding the Gatekeeper, page 5-35
- Configuring Cisco Unified CallManager Express and Other Standards-Based H.323 Call-Control Devices: Configuring the Dial Peer, page 5-36

Related Topics

- Required Tasks for Integrating With a Call-Control Device Through a Gatekeeper, page 5-28
- About Integration With a Call-Control Device Through a Gatekeeper, page 5-28

Configuring Cisco Unified CallManager Express and Other Standards-Based H.323 Call-Control Devices: Adding the Gatekeeper

Before You Begin

- If the gatekeeper is already defined in your Cisco IOS router, then do not perform this task. Proceed to the "Configuring Cisco Unified MeetingPlace Express: Adding the Gatekeeper" section on page 5-38.
- See the Prerequisites for Integrating With a Call-Control Device Through a Gatekeeper and other information in the "About Integration With a Call-Control Device Through a Gatekeeper" section on page 5-28.
- Complete the task described in the "Configuring the Gatekeeper: Identifying the Hostname, Zone, and Technology Prefix" section on page 5-29.
- This task is performed in the Cisco IOS command)-line interface (CLI) of the Cisco router.

Procedure

Step 1 On the Cisco router, enter privileged EXEC mode or any other security level set by a system administrator. Enter your password if prompted.

```
Router# enable
```

Step 2 Enter global configuration mode.

```
Router# configure terminal
```

Step 3 Enter interface configuration mode.

```
Router(config)# interface type number
```

Step 4 Configure the IP address and subnet mask used by this gateway.

```
Router(config-if)# ip address [ip-address] [subnet-mask]
```

Step 5 (Optional) If you configured a Fast Ethernet interface, configure duplex operation as auto, which specifies the autonegotiation capability. The gateway automatically operates at half or full duplex depending on environmental factors, such as the type of media and transmission speeds for the peer routers, hubs, and switches used in the network configuration.

```
Router(config-if)# duplex auto
```

Step 6 (Optional) If you configured a Fast Ethernet interface, configure the speed for this gateway.

```
Router(config-if)# speed auto
```

Step 7 Configure this interface as an H.323 gateway interface.

```
Router(config-if)# h323-gateway voip interface
```

Step 8 Define the name and location of the gatekeeper for this gateway.

```
Router(config-if)# h323-gateway voip id [hostname] ipaddr [ip-address]
```

Step 9 Configure the H.323 name of the gateway identifying this gateway to its associated gatekeeper.

```
Router(config-if)# h323-gateway voip h323-id hostname
```

Step 10 Define the technology prefix that the gateway will register with the gatekeeper.

```
Router(config-if)# h323-gateway voip tech-prefix prefix
```

- *prefix*—Defines the numbers used as the technology prefixes. Each technology prefix can contain up to 11 characters. Although not strictly necessary, a pound (#) symbol is frequently used as the last digit in a technology prefix. Valid characters are 0 though 9, the pound (#) symbol, and the asterisk (*).

Step 11 Set the source IP address to be used for this gateway. This command binds all H.323 messages from the gateway to this IP address.

```
Router(config-if)# h323-gateway voip bind srcaddr [ip-address]
```

Step 12 Exit the current mode.

```
Router(config-if)# exit
```

Step 13 Proceed to the "Configuring Cisco Unified CallManager Express and Other Standards-Based H.323 Call-Control Devices: Configuring the Dial Peer" section on page 5-36.

Example

The following example displays integration between a Cisco IOS voice gateway with an IP address of 1.1.100.1 and a gatekeeper named gk-1 that has an IP address of 1.1.100.200.

```
interface FastEthernet0/0
 ip address 1.1.100.1 255.255.255.0
 duplex auto
 speed auto
 h323-gateway voip interface
 h323-gateway voip id gk-1 ipaddr 1.1.100.200 1719
 h323-gateway voip h323-id cme1
 h323-gateway voip tech-prefix 1#
 h323-gateway voip bind srcaddr 1.1.100.1
```

Related Topics

- Configuring Cisco Unified CallManager Express and Other Standards-Based H.323 Call-Control Devices, page 5-34

Configuring Cisco Unified CallManager Express and Other Standards-Based H.323 Call-Control Devices: Configuring the Dial Peer

This topic describes how to enable your call-control device to route calls to Cisco Unified MeetingPlace Express by configuring a dial peer. Configuring dial peers is the key to implementing dial plans and providing voice services over an IP packet network. Dial peers are used to identify call source and destination endpoints and to define the characteristics applied to each call leg in the call connection.

Before You Begin

- See the Prerequisites for Integrating with Cisco Unified CallManager Express and Other Standards-Based H.323 Call Control Devices and other information in the "About Integration With a Call-Control Device Through a Gatekeeper" section on page 5-28.
- Complete the task described in the "Configuring Cisco Unified CallManager Express and Other Standards-Based H.323 Call-Control Devices: Adding the Gatekeeper" section on page 5-35.

- This task is performed in the Cisco IOS command-line interface (CLI) of the router. For more information about the Cisco IOS commands used in this procedure, see the *Cisco IOS Commands Master List* for your Cisco IOS software major release.

Procedure

Step 1 On the Cisco router, enter privileged EXEC mode or any other security level set by a system administrator. Enter your password if prompted.

```
Router# enable
```

Step 2 Enter global configuration mode.

```
Router# configure terminal
```

Step 3 Enter dial peer voice configuration mode and define a remote voice over IP (VoIP) dial peer.

```
Router(config)# dial-peer voice number voip
```

- *number*—is one or more digits that identify the dial peer. Valid entries are from 1 to 2147483647.
- **voip**—indicates a VoIP peer that uses voice encapsulation on the IP network.

Step 4 Route calls to the Cisco Unified MeetingPlace Express server.

```
Router(config-dialpeer)# destination-pattern digits
```

- *digits*—indicates the numbers that match the destination pattern.

Step 5 (Optional) Provide a comment or a description to help you remember what is attached to this interface.

```
Router(config-dialpeer)# description string
```

Step 6 Specify the network specific address for this dial peer to use Registration, Admission, and Status (RAS) signaling

```
Router(config-dialpeer)# session target ras
```

Step 7 Configure the router to use dual tone multifrequency (DTMF) relay to transport DTMF digits.

```
Router(config-dialpeer)# dtmf-relay h245-alphanumeric
```

Step 8 Configure the router to use a particular codec.

```
Router(config-dialpeer)# codec [g711ulaw | g711alaw]
```

Step 9 Disable voice activity detection (VAD) for the calls using this dial peer.

```
Router(config-dialpeer)# [no] vad
```

Step 10 Exit the current mode.

```
Router(config-dialpeer)# exit
```

Step 11 Proceed to the "Configuring Cisco Unified MeetingPlace Express: Adding the Gatekeeper" section on page 5-38.

Example

The following example displays dial peers that were configured to direct calls to a primary Cisco Unified MeetingPlace Express number and an alternate Cisco Unified MeetingPlace Express number using RAS.

```
!
dial-peer voice 1 voip
 destination-pattern 7777
 description MP express main number
 session target ras
 dtmf-relay h245-alphanumeric
 codec g711ulaw
 no vad
!
dial-peer voice 2
 voip destination-pattern 7000
 description MP express alternate number
 session target ras
 dtmf-relay h245-alphanumeric
 codec g711ulaw
 no vad
```

Related Topics

- Configuring Cisco Unified CallManager Express and Other Standards-Based H.323 Call-Control Devices, page 5-34

Configuring Cisco Unified MeetingPlace Express: Adding the Gatekeeper

This topic describes how to add a gatekeeper to Cisco Unified MeetingPlace Express in an H.323 environment.

Before You Begin

- Read the following sections:
  - About Integration With a Call-Control Device Through a Gatekeeper, page 5-28
  - Prerequisites for Integrating With a Call-Control Device Through a Gatekeeper, page 5-28
- Configure your call-control device by completing one of the following tasks:
  - If you are integrating with Cisco Unified CallManager, see the "Configuring Cisco Unified CallManager: Adding the Gatekeeper, Trunk, and Route Pattern" section on page 5-32.
  - If you are integrating with Cisco Unified CallManager Express or a Cisco IOS software voice-enabled router, see the "Configuring Cisco Unified CallManager Express and Other Standards-Based H.323 Call-Control Devices" section on page 5-34.
- This task is completed in the Cisco Unified MeetingPlace Express Administration Center.

Procedure

Step 1 Log in to Cisco Unified MeetingPlace Express.

Step 2 Click **Administration** at the top of the page.

Step 3 On the left side of the page:

a. Click **System Configuration**.

b. Click **Call Configuration**.

c. Click **H.323 Configuration**.

Step 4 In the H.323 Configuration page, configure the fields in Table 5-16.

Table 5-16 Required H.323 Configuration on Cisco Unified MeetingPlace Express for Integration With a Gatekeeper

H.323 Configuration Page Field	Required Value
H.323 enabled	Yes
Local H.323 port	1720
Use gatekeeper	Yes
Gatekeeper	Gatekeeper IP address

Step 5 Click **Save**.

Step 6 On the left side of the page:

a. Click **System Configuration**.

b. Click **Call Configuration**.

c. Click **Dial Configuration**.

Step 7 In the Dial Configuration Page, configure the Outdials field to **H.323**.

Step 8 Click **Save**.

Step 9 Test this integration by placing a call from any phone to the phone number that is used to access the Cisco Unified MeetingPlace Express system. You should hear the “Welcome to Cisco Unified MeetingPlace Express” greeting.

Related Topics

- About Integration With a Call-Control Device Through a Gatekeeper, page 5-28
- About This Page: H.323 Configuration, page B-100
- About This Page: Dial Configuration, page B-53

About Integration in a SIP Environment

Cisco Unified MeetingPlace Express supports integration with various devices in a SIP call-control environment. To deploy Cisco Unified MeetingPlace Express in a SIP environment, your network must have one of the following applications to route calls:

- Cisco Unified CallManager
- Cisco Unified CallManager Express
- Cisco IOS software voice-enabled router
- Cisco SIP Proxy Server

See the following sections:

- Prerequisites for Integration in a SIP Environment, page 5-40
- Cisco Unified CallManager Restrictions for Integration in a SIP Environment, page 5-40
- Required Tasks for Integration in a SIP Environment, page 5-41

Prerequisites for Integration in a SIP Environment

- Verify that the versions of your call-control device and Cisco Unified MeetingPlace Express are compatible. See the *Release Notes for Cisco Unified MeetingPlace Express*.
- Verify that your IP telephony network is working properly. For example:
  – Verify that the Cisco Unified IP Phones are connected and added to the database of your call-control device.
  – Verify that you can place and receive internal and external calls on the Cisco Unified IP Phones.

Related Topics

- About Integration in a SIP Environment, page 5-39

Cisco Unified CallManager Restrictions for Integration in a SIP Environment

- The number of simultaneous calls through the SIP trunk is limited by the number of available Media Termination Point (MTP) resources. This limitation exists because SIP uses in-band RFC 2833 for DTMF tones while H.323, MGCP, TAPI/JTAPI and SCCP all use out-of-band DTMF in Cisco Unified CallManager. The following restrictions and conditions apply:
  – Do not configure more than 48 MTP resources on the Cisco Unified CallManager server. Since each call requires two streams to the MTP device, with one Cisco Unified CallManager server and no external hardware MTP resources, the SIP trunk can support only up to 24 calls.
  – If you have a cluster of Cisco Unified CallManager servers, then each server in the cluster can provide up to 48 MTP resources to support calls through the SIP trunk. For example, a cluster with two Cisco Unified CallManager servers can support up to 48 calls over the SIP trunk.
  – For external hardware MTP resources, you can use a Cisco Catalyst 6500 Series Communication Media Module (CMM) with at least one Ad-Hoc Conferencing and Transcoding (ACT) Port Adapter. This combination provides 512 MTP resources.
  – You can avoid the MTP resource issue altogether by using an H.323 connection through a gatekeeper between Cisco Unified MeetingPlace Express and Cisco Unified CallManager. See the "About Integration With a Call-Control Device Through a Gatekeeper" section on page 5-28.
- Cisco Unified CallManager Release 4.x requires DSP resources to transcode calls in other codecs to G.711. Without the transcoding, only G.711 calls can be accepted by Cisco Unified MeetingPlace Express over the SIP trunk from Cisco Unified CallManager Release 4.x. For information about supported Cisco DSP resources for transcoding, see the *Cisco Unified CallManager System Guide*.

- Cisco Unified CallManager Release 4.x does not support RFC 3515, the SIP refer method of transferring calls. Therefore, if you use a SIP trunk to integrate Cisco Unified MeetingPlace Express with Cisco Unified CallManager Release 4.x, then calls cannot be transferred to the attendant. Instead, callers hear a busy signal in the situations listed in the "About Operator Assistance" section on page 3-1.

Related Topics

- About Integration in a SIP Environment, page 5-39

Required Tasks for Integration in a SIP Environment

The tasks you need to complete depend on the type of call-control device you are using. See Table 5-17 for a task roadmap.

Table 5-17 Roadmap to Configuring Cisco Unified MeetingPlace Express With a Call-Control Device in a SIP Environment

Task	Reference
1. Configure your particular call-control device.	See one of the following procedures depending on your call-control device: • If you are using Cisco Unified CallManager, see the "Configuring Cisco Unified CallManager: Adding the SIP Trunk and Route Pattern" section on page 5-41. • If you are using Cisco Unified CallManager Express or a Cisco IOS software voice-enabled router, see the "Configuring Cisco Unified CallManager Express and Other Cisco IOS Software Voice-Enabled Routers: Adding the SIP Gateway and Dial Peer" section on page 5-47. • If you are using Cisco SIP Proxy Server, see the "Configuring Cisco SIP Proxy Server" section on page 5-52
2. Configure Cisco Unified MeetingPlace Express to connect to your call-control device through a SIP trunk.	See the "Configuring Cisco Unified MeetingPlace Express: Connecting to a Call-Control Device Through a SIP Trunk" section on page 5-55.

Related Topics

- About Integration in a SIP Environment, page 5-39

Configuring Cisco Unified CallManager: Adding the SIP Trunk and Route Pattern

Perform one of the following tasks, depending on your Cisco Unified CallManager release:

- Adding the SIP Trunk and Route Pattern in Cisco Unified CallManager Release 4.1, page 5-42
- Adding the SIP Trunk and Route Pattern in Cisco Unified CallManager Release 5.x, page 5-44

Related Topics

- About Integration in a SIP Environment, page 5-39

Adding the SIP Trunk and Route Pattern in Cisco Unified CallManager Release 4.1

This topic describes how to add the SIP trunk to the Cisco Unified CallManager configuration database. This topic also describes how to enable Cisco Unified CallManager to route calls to Cisco Unified MeetingPlace Express by associating a phone number with the trunk. This association is called a route pattern.

Before You Begin

- Read the following sections:
  - About Integration in a SIP Environment, page 5-39
  - Prerequisites for Integration in a SIP Environment, page 5-40
  - Cisco Unified CallManager Restrictions for Integration in a SIP Environment, page 5-40
- This task is performed in the Cisco Unified CallManager Administration pages. Because the pages and menus vary by Cisco Unified CallManager release, you may need to see the Cisco Unified CallManager Administration online help for more accurate step-by-step instructions than those provided in this procedure.

Procedure

Step 1 Go to http://*ccm-server*/ccmadmin/main.asp, where *ccm-server* is the fully qualified domain name or IP address of the Cisco Unified CallManager server.

Step 2 Log in with your Cisco Unified CallManager administrator username and password.

Step 3 Add a new SIP trunk by completing the following actions:

a. In the menu at the top of the Cisco Unified CallManager Administration page, click **Device > Trunk**.

b. In the top right corner, click **Add a New Trunk**.

c. In the Trunk type field, select **SIP Trunk**.

d. In the Device Protocol field, select **SIP**.

e. Click **Next**.

f. In the Trunk Configuration page, configure the fields described in Table 5-18.

Table 5-18 Fields for Adding A New Trunk to Cisco Unified CallManager

Trunk Configuration Field	Action
Device Name	Enter a unique identifier for this trunk, such as the name or IP address of the Cisco Unified MeetingPlace Express server.
Device Pool	If no device pools are defined, then select **Default**. If the Cisco Unified CallManager deployment utilizes customer-defined device pools, then either create a new device pool or choose an existing device pool for a region with a codec that is compatible with the conferencing gateway. Currently, Cisco Unified MeetingPlace Express supports only the G.711 audio codec. The device pool specifies a collection of properties for this device including Cisco Unified CallManager Group, Date/Time Group, and Region.
Media Termination Point Required	Check this checkbox.
Destination Address	Enter the IP address of Port 1 (eth0) of the Cisco Unified MeetingPlace Express server.
Destination Port Incoming Port	Keep the default value of **5060**. If it becomes necessary for you to change this port number, then make sure that you configure the exact same port number in the Local SIP port: field in the SIP Configuration page of the Cisco Unified MeetingPlace Express Administration Center.
Outgoing Transport Type	Select **UDP**.

g. For all other required fields on the Trunk Configuration page, configure the fields appropriately for the current Cisco Unified CallManager deployment. For information about each field, see the Cisco Unified CallManager online help or see the administration guide for your specific Cisco Unified CallManager release.

h. Click **Insert** to add the new trunk.

Step 4 Add the route pattern to the Cisco Unified CallManager database by completing the following actions:

a. In the menu at the top of the Cisco Unified CallManager Administration page, click **Route Plan** > **Route/Hunt** > **Route Pattern**.

b. In the top right corner, click **Add a New Route Pattern**.

c. In the Route Pattern Configuration page, configure the fields described in Table 5-19.

Table 5-19 Fields for Adding A New Route Pattern to Cisco Unified CallManager

Route Pattern Configuration Field	Action
Route Pattern	Enter the phone number for users to call in to Cisco Unified MeetingPlace Express. This number must match the value configured in the Username: field in the SIP Configuration page of the Cisco Unified MeetingPlace Express Administration Center. Do not enter any spaces in this field.
Gateway or Route List	Select the value that matches the Device Name you entered for the gateway in Step 3f.

d. Click **Insert** to add the route pattern to Cisco Unified MeetingPlace Express.

Related Topics

- About Integration in a SIP Environment, page 5-39
- About This Page: SIP Configuration, page B-167
- About Integration With Cisco Unified CallManager, page 5-1

Adding the SIP Trunk and Route Pattern in Cisco Unified CallManager Release 5.x

This topic describes how to add the SIP trunk to the Cisco Unified CallManager configuration database. This topic also describes how to enable Cisco Unified CallManager to route calls to Cisco Unified MeetingPlace Express by associating a phone number with the trunk. This association is called a route pattern.

Note The information in this topic only applies to Cisco Unified MeetingPlace Express Release 1.1.2 and later.

Before You Begin

- Read the following sections:
  - About Integration in a SIP Environment, page 5-39
  - Prerequisites for Integration in a SIP Environment, page 5-40
  - Cisco Unified CallManager Restrictions for Integration in a SIP Environment, page 5-40
- This task is performed in the Cisco Unified CallManager Administration pages. Because the pages and menus vary by Cisco Unified CallManager release, you may need to see the Cisco Unified CallManager Administration online help for more accurate step-by-step instructions than those provided in this procedure.

Procedure

Step 1 Go to http://*ccm-server*/ccmadmin/main.asp, where *ccm-server* is the fully qualified domain name or IP address of the Cisco Unified CallManager server.

Step 2 Log in with your Cisco Unified CallManager administrator username and password.

Step 3 Add a new SIP trunk by completing the following actions:

a. In the menu at the top of the Cisco Unified CallManager Administration page, click **Device > Trunk**.

b. Click **Add New**.

c. In the Trunk type field, select **SIP Trunk**.

d. In the Device Protocol field, select **SIP** if it is not automatically selected for you.

e. Click **Next**.

f. In the Trunk Configuration page, configure the fields described in Table 5-20.

Table 5-20 Fields for Adding A New Trunk to Cisco Unified CallManager

Trunk Configuration Field	Action
Device Name	Enter a unique identifier for this trunk, such as the name or IP address of the Cisco Unified MeetingPlace Express server.
Device Pool	If no device pools are defined, then select **Default**. If the Cisco Unified CallManager deployment utilizes customer-defined device pools, then either create a new device pool or choose an existing device pool for a region with a codec that is compatible with the conferencing gateway. Currently, Cisco Unified MeetingPlace Express supports only the G.711 audio codec. The device pool specifies a collection of properties for this device including Cisco Unified CallManager Group, Date/Time Group, and Region.
Media Termination Point Required	Check this checkbox.
Destination Address	Enter the IP address of Port 1 (eth0) of the Cisco Unified MeetingPlace Express server.
Destination Port	Keep the default value of **5060**. If it becomes necessary for you to change this port number, then make sure that you configure the exact same port number in the Local SIP port: field in the SIP Configuration page of the Cisco Unified MeetingPlace Express Administration Center.
SIP Trunk Security Profile	Select **Non Secure SIP Trunk Profile**.
SIP Profile	Select **Standard SIP Profile**.
DTMF Signaling Method	Select **RFC 2833**.

g. For all other required fields on the Trunk Configuration page, configure the fields appropriately for the current Cisco Unified CallManager deployment. For information about each field, see the Cisco Unified CallManager online help or see the administration guide for your specific Cisco Unified CallManager release.

h. Click **Save** to add the new trunk.

Step 4 Configure the standard SIP profile by completing the following actions:

a. In the menu at the top of the Cisco Unified CallManager Administration page, click **Device** > **Device Settings** > **SIP Profile**.

b. To list all SIP profiles, click **Find** without entering anything in the Search Options fields.

c. Under Search Results, click **Standard SIP Profile**.

d. On the SIP Profile Configuration page, configure the fields described in Table 5-21.

Table 5-21 Fields for Configuring the Standard SIP Profile in Cisco Unified CallManager

SIP Profile Field	Action
Default MTP Telephony Event Payload Type	Keep the default value of **101**. If it becomes necessary for you to change this number, then make sure that you configure the exact same number in the RFC2833 payload type field on the SIP Configuration page of the Cisco Unified MeetingPlace Express Administration Center.
Disable Early Media on 180	Ensure that this checkbox is not selected.

e. For all other required fields on the SIP Profile Configuration page, configure the fields appropriately for the current Cisco Unified CallManager deployment. For information about each field, see the Cisco Unified CallManager online help or see the administration guide for your specific Cisco Unified CallManager release.

f. Click **Save**.

Step 5 Add the route pattern to the Cisco Unified CallManager database by completing the following actions:

a. In the menu at the top of the Cisco Unified CallManager Administration page, click **Route Plan** > **Route/Hunt** > **Route Pattern**.

b. Click **Add New**.

c. In the Route Pattern Configuration page, configure the fields described in Table 5-22.

Table 5-22 Fields for Adding A New Route Pattern to Cisco Unified CallManager

Route Pattern Configuration Field	Action
Route Pattern	Enter the phone number for users to call in to Cisco Unified MeetingPlace Express. This number must match the value configured in the Username: field in the SIP Configuration page of the Cisco Unified MeetingPlace Express Administration Center. Do not enter any spaces in this field.
Gateway or Route List	Select the value that matches the Device Name you entered for the gateway in Step 3f.

d. For all other required fields on the Route Pattern Configuration page, configure the fields appropriately for the current Cisco Unified CallManager deployment. For information about each field, see the Cisco Unified CallManager online help or see the administration guide for your specific Cisco Unified CallManager release.

e. Click **Save** to add the route pattern to Cisco Unified MeetingPlace Express.

f. Click **OK** to any pop-up dialog box messages that you see.

Related Topics

- About Integration in a SIP Environment, page 5-39
- About This Page: SIP Configuration, page B-167
- About Integration With Cisco Unified CallManager, page 5-1

Configuring Cisco Unified CallManager Express and Other Cisco IOS Software Voice-Enabled Routers: Adding the SIP Gateway and Dial Peer

This topic describes how to configure Cisco call-control devices other than Cisco Unified CallManager for the SIP voice over IP (VoIP) service.

For further information about Cisco IOS and SIP configuration, see the *Cisco IOS SIP Configuration Guide* for your Cisco IOS software major release.

This topic is divided in to four tasks:

- Shutting Down or Enabling Voice over IP (VoIP) Service on the Cisco Gateway, page 5-47
- Configuring SIP Server Support, page 5-48
- Verifying SIP Gateway Status, page 5-49
- Configuring SIP Support for Voice Dial Peers, page 5-50

Related Topics

- About Integration in a SIP Environment, page 5-39

Shutting Down or Enabling Voice over IP (VoIP) Service on the Cisco Gateway

Before You Begin

- Read the following sections:
  - About Integration in a SIP Environment, page 5-39
  - Prerequisites for Integration in a SIP Environment, page 5-40
- This task is performed in the Cisco IOS command-line interface (CLI) of the Cisco router. For more information about the Cisco IOS commands used in this procedure, see the *Cisco IOS Commands Master List* for your Cisco IOS software major release.

Procedure

Step 1 On the Cisco router, enter privileged EXEC mode or any other security level set by a system administrator. Enter your password if prompted.

```
Router# enable
```

Step 2 Enter global configuration mode.

```
Router# configure terminal
```

Step 3 Enter voice-service VoIP configuration mode.

```
Router(config)# voice service voip
```

Step 4 Enter SIP configuration mode.

```
Router(config-voi-serv)# sip
```

Step 5 Shut down or enable VoIP call services for the selected submode.

```
Router(config-serv-sip)# [no] call service stop [forced] [maintain-registration]
```

- To stop SIP service without killing active calls, choose the maintain registration attribute, that is:

```
Router(config-serv-sip)# call service stop maintain-registration
```

- To stop SIP service and tear down active calls, choose the forced argument, that is:

```
Router(config-serv-sip)# call service stop forced
```

Step 6 Exit the current mode.

```
Router(config-serv-sip)# exit
```

Step 7 Proceed to the "Configuring SIP Server Support" section on page 5-48.

Related Topics

- About Integration in a SIP Environment, page 5-39
- Configuring Cisco Unified CallManager Express and Other Cisco IOS Software Voice-Enabled Routers: Adding the SIP Gateway and Dial Peer, page 5-47

Configuring SIP Server Support

Before You Begin

- Read the following sections:
  - About Integration in a SIP Environment, page 5-39
  - Prerequisites for Integration in a SIP Environment, page 5-40
- Complete the task described in the "Shutting Down or Enabling Voice over IP (VoIP) Service on the Cisco Gateway" section on page 5-47.
- This task is performed in the Cisco IOS command-line interface (CLI) of the Cisco router. For more information about the Cisco IOS commands used in this procedure, see the *Cisco IOS Commands Master List* for your Cisco IOS software major release.

Procedure

Step 1 On the Cisco router, enter privileged EXEC mode or any other security level set by a system administrator. Enter your password if prompted.

```
Router# enable
```

Step 2 Enter global configuration mode.

```
Router# configure terminal
```

Step 3 Enter SIP user-agent configuration mode.

```
Router(config)# sip-ua
```

Step 4 Register E.164 numbers with an external SIP proxy or SIP registrar server.

```
Router(config-sip-ua)# registrar [dns:address] [ipv4:ip-address] expires seconds [tcp]
[secondary]
```

- **dns**:*address*—Domain-name server that resolves the name of the dial peer to receive calls.
- **ipv4**:*ip-address*:—IP address of the dial peer to receive calls.
- **expires** *seconds*—Default registration time, in seconds.
- **tcp**—Sets transport layer protocol to TCP. UDP is the default.
- **secondary**—(Optional) Specifies registration with a secondary SIP proxy or registrar for redundancy purposes.

Step 5 Specify the network address (IP address or hostname) of the SIP proxy server.

```
Router(config-sip-ua)# sip-server [ipv4:ip-address] [dns:address]
```

Step 6 Exit the current mode.

```
Router(config-sip-ua)# exit
```

Step 7 Proceed to the "Verifying SIP Gateway Status" section on page 5-49.

Related Topics

- About Integration in a SIP Environment, page 5-39
- Configuring Cisco Unified CallManager Express and Other Cisco IOS Software Voice-Enabled Routers: Adding the SIP Gateway and Dial Peer, page 5-47

Verifying SIP Gateway Status

Before You Begin

- Complete the task described in the "Configuring SIP Server Support" section on page 5-48.
- This task is performed in the Cisco IOS command-line interface (CLI) of the Cisco router. For more information about the Cisco IOS commands used in this procedure, see the *Cisco IOS Commands Master List* for your Cisco IOS software major release.

Procedure

Step 1 Use the following commands on the Cisco router to verify your SIP gateway status:

Router # **show sip service**

Router # **show sip-ua register status**

Router # **show sip-ua statistics**

Router # **show sip-ua status**

Router # **show sip-ua timers**

Step 2 Proceed to the "Configuring SIP Support for Voice Dial Peers" section on page 5-50.

Related Topics

- About Integration in a SIP Environment, page 5-39
- Configuring Cisco Unified CallManager Express and Other Cisco IOS Software Voice-Enabled Routers: Adding the SIP Gateway and Dial Peer, page 5-47

Configuring SIP Support for Voice Dial Peers

This topic describes how to enable your call-control device to route calls to Cisco Unified MeetingPlace Express using SIP. Configuring dial peers is the key to implementing dial plans and providing voice services over an IP packet network. Dial peers are used to identify call source and destination endpoints and to define the characteristics applied to each call leg in the call connection.

Before You Begin

- Read the following sections:
  - About Integration in a SIP Environment, page 5-39
  - Prerequisites for Integration in a SIP Environment, page 5-40
- Complete the "Verifying SIP Gateway Status" section on page 5-49.
- This task is performed in the Cisco IOS command-line interface (CLI) of the router. For more information about the Cisco IOS commands used in this procedure, see the *Cisco IOS Commands Master List* for your Cisco IOS software major release.

Procedure

Step 1 On the Cisco router, enter privileged EXEC mode or any other security level set by a system administrator. Enter your password if prompted.

```
Router# enable
```

Step 2 Enter global configuration mode.

```
Router# configure terminal
```

Step 3 Enter dial peer voice configuration mode and define a remote voice over IP (VoIP) dial peer.

```
Router(config)# dial-peer voice number voip
```

- *number*—One or more digits that identify the dial peer. Valid entries are from 1 to 2147483647.
- **voip**—Indicates a VoIP peer that uses voice encapsulation on the IP network.

Step 4 Enter the session protocol type.

```
Router(config-dialpeer)# session protocol sipv2
```

- **sipv2**—Configures the dial peer to use IETF SIP.

Step 5 Configure the router to use a particular codec.

```
Router(config-dialpeer)# codec [g711ulaw | g711alaw]}
```

Step 6 Configure the router to use dual tone multifrequency (DTMF) relay to transport DTMF digits.

```
Router(config-dialpeer)# dtmf-relay rtp-nte
```

Step 7 Specify a network-specific address for a dial peer.

```
Router(config-dialpeer)# session target {sip-server | dns:[hostname]|
ipv4:ip-address:[port-num]}
```

- **sip-server**— Sets the session target to the global SIP server. Used when the sip-server command has already specified the host name or IP address of the SIP server interface.
- **dns**:*hostname*—Sets the global SIP server interface to a domain name server (DNS) host name. A valid DNS host name takes the following format: name.gateway.xyz.
- **ipv4**:*ip-address*:—Sets the IP address.
- *port-num*—(Optional) Sets the UDP port number for the SIP server.

Note Wildcards can be used when defining the session target for VoIP peers

Step 8 Disable voice activity detection (VAD) for the calls using this dial peer.

```
Router(config-dialpeer)# [no] vad
```

Step 9 Exit the current mode.

```
Router(config-dialpeer)# exit
```

Example

The following example displays a dial peer that was configured to direct calls to a Cisco Unified MeetingPlace Express number by using SIP. The Cisco Unified MeetingPlace Express IP address is configured as 10.8.17.42.

```
!
dial-peer voice 123 voip
 destination-pattern [12]...
 session protocol sipv2
 session target ipv4:10.8.17.42
 dtmf-relay rtp-nte
 codec g711ulaw
 no vad
```

Related Topics

- About Integration in a SIP Environment, page 5-39
- Configuring Cisco Unified CallManager Express and Other Cisco IOS Software Voice-Enabled Routers: Adding the SIP Gateway and Dial Peer, page 5-47

Configuring Cisco SIP Proxy Server

The Cisco SIP proxy server (Cisco SPS) is a call-control software package that enables service providers and others to build scalable, reliable packet voice networks and to provide call-session management in a VoIP network. It can also serve as a registrar or redirect server. It provides a full array of call-routing capabilities for maximizing network performance in both small and large packet voice networks.

Cisco SPS has the capabilities of an edge proxy server, performing such functions as authentication, accounting, registration, network-access control, and security. It can also has the capabilities of an infrastructure proxy server, performing such functions as next-hop routing based on received or translated destination URLs.

This topic describes how to configure Cisco SPS to recognize Cisco Unified MeetingPlace Express as an endpoint. It is comprised of two tasks:

- Configuring Cisco SIP Proxy Server: Adding Cisco Unified MeetingPlace Express as a Subscriber, page 5-52
- Configuring Cisco SIP Proxy Server: Configuring a Route, page 5-53

Note For detailed information about how to install, configure, and manage Cisco SPS, see the *Installation Guide* or *Administrator Guide* for your Cisco SIP proxy server release.

Related Topics

- Required Tasks for Integration in a SIP Environment, page 5-41
- About Integration in a SIP Environment, page 5-39

Configuring Cisco SIP Proxy Server: Adding Cisco Unified MeetingPlace Express as a Subscriber

A subscriber is a SIP endpoint that has static, configurable subscriber information. This topic describes how to add Cisco Unified MeetingPlace Express as a subscriber to Cisco SIP proxy server (SPS).

Before You Begin

Install and configure your Cisco SPS by using the instructions in the Cisco SIP Proxy Server documentation.

Procedure

Step 1 From the Cisco SPS main menu, click **Subscribers**.

Step 2 To add a new subscriber, do the following:

a. Click **Add**.

b. Enter Cisco Unified MeetingPlace Express information. Any field that has a red asterisk must have an entry.

c. Click **Submit**.

Related Topics

- Configuring Cisco SIP Proxy Server, page 5-52
- About Integration in a SIP Environment, page 5-39

Configuring Cisco SIP Proxy Server: Configuring a Route

You can add, change, or delete dynamic and static routes.

A dynamic route is a path through the network that is automatically calculated according to routing protocols and routing update messages. A static route is a fixed path through the network that you explicitly configure. Static routes take precedence over dynamic routes and are synchronized among farm members. Configurable route information includes the following:

- Destination pattern and type
- Next hop and next-hop port
- Transport protocol
- Priority and weight
- Tech-prefix action
- Allow less-specific route
- Route block
- In service
- Label

Define destination patterns for routes when setting up a static route with Cisco SIP Proxy Server (SPS) as follows:

- Use user=phone when routing based on the phone number in the user portion.

  `Example: +14085550122@cisco.com; user=phone` (where 140855501222 is an E.164 number)

  `Example: 50122@cisco.com; user=phone` (where 50122 is an unambiguous extension within the cisco.com domain)

- Use user=ip when routing based on the domain portion (also known as domain routing).

You can use any of the characters included in the following directive when specifying a destination pattern, with the following caveat:

- NumericUsernameCharacterSet—Set of characters that Cisco SPS uses to determine whether the user-information portion of a Request-URI is in a category that applies to the "NumericUsernameInterpretation" processing step. This set does not apply to any user-information parameters.

  Default is +0123456789.-() (global phone number combinations). For more information on this directive, see the sipd.conf file.

Caution Some characters are treated as visual separators (examples: ( ) . -). These characters are removed before looking in the route database. Do not include them when defining a route destination pattern.

Special characters for defining a route are as follows:

- * indicates a multiple-digit wildcard (example: `9*` matches `911`, `914085551212`)
- . indicates a single-digit wildcard (example: `9..` matches `911`, but not `9111`)
- * indicates an actual * character (example: `\*69` matches `*69`)

Before You Begin

- Install and configure your Cisco SPS by using the instructions in the Cisco SIP Proxy Server documentation.
- Complete the task described in the "Configuring Cisco SIP Proxy Server: Adding Cisco Unified MeetingPlace Express as a Subscriber" section on page 5-52.

Procedure

Step 1 From the Cisco SPS main menu, click **Routes**.

Step 2 Display existing routes by performing a search with the search tool.

Step 3 To add a new route, do the following:

a. Click **Add**.

b. Enter route information. Any field that has a red asterisk must have an entry.

c. Click **Submit**.

Related Topics

- Configuring Cisco SIP Proxy Server, page 5-52
- About Integration in a SIP Environment, page 5-39

Configuring Cisco Unified MeetingPlace Express: Connecting to a Call-Control Device Through a SIP Trunk

This topic describes how to configure Cisco Unified MeetingPlace Express to connect directly to a supported call-control device through a SIP trunk.

Before You Begin

- See the Prerequisites for Integration in a SIP Environment and Cisco Unified CallManager Restrictions for Integration in a SIP Environment in the "About Integration in a SIP Environment" section on page 5-39.
- Configure your call-control device:
  - If you are integrating with Cisco Unified CallManager, complete the "Configuring Cisco Unified CallManager: Adding the SIP Trunk and Route Pattern" section on page 5-41.
  - If you are integrating with Cisco Unified CallManager Express or a Cisco IOS software voice-enabled router, complete the tasks described in the "Configuring Cisco Unified CallManager Express and Other Cisco IOS Software Voice-Enabled Routers: Adding the SIP Gateway and Dial Peer" section on page 5-47.
  - If you are integrating with Cisco SIP Proxy Server, complete the "Configuring Cisco SIP Proxy Server" section on page 5-52.
- This task is completed in the Cisco Unified MeetingPlace Express Administration Center.

Procedure

Step 1 Log in to Cisco Unified MeetingPlace Express.

Step 2 Click **Administration** at the top of the page.

Step 3 On the left side of the page:

a. Click **System Configuration**.

b. Click **Call Configuration**.

c. Click **SIP Configuration**.

Step 4 In the SIP Configuration page, configure the fields in Table 5-23.

Table 5-23 Required Configuration for SIP Configuration Page on Cisco Unified MeetingPlace Express for Integration Through a SIP Trunk

SIP Configuration Page Field	Required Value
SIP enabled?	Yes
Local SIP port:	5060 (default)

Table 5-23 Required Configuration for SIP Configuration Page on Cisco Unified MeetingPlace Express for Integration Through a SIP Trunk (continued)

SIP Configuration Page Field	Required Value
SIP proxy server 1:	IP address of the call-control device or SIP proxy server. SIP calls initiated by Cisco Unified MeetingPlace Express are directed to this IP address. If you have a cluster of call-control devices, then enter the IP address of the primary call-processing server in the cluster.
SIP proxy server 2: SIP proxy server 3: SIP proxy server 4: SIP proxy server 5: SIP proxy server 6:	IP addresses of other call-control devices or SIP proxy servers in a cluster that provides call-processing redundancy, if any. **Note** If the primary call-control application goes down, Cisco Unified MeetingPlace Express cannot complete dialed-out calls. These fields enable only incoming calls to be routed by the failover call-control application.

Step 5 Click **Save**.

Step 6 On the left side of the page:

a. Click **System Configuration**.

b. Click **Call Configuration**.

c. Click **Dial Configuration**.

Step 7 In the Dial Configuration Page, configure the Outdials field to **SIP**.

Step 8 Click **Save**.

Step 9 Test this integration by placing a call from any phone to the phone number that is used to access the Cisco Unified MeetingPlace Express system. You should hear the “Welcome to Cisco Unified MeetingPlace Express” greeting.

Related Topics

- Required Tasks for Integration in a SIP Environment, page 5-41
- About Integration in a SIP Environment, page 5-39