Main page: Cisco Unified MeetingPlace, Release 8.0

Up one level: Configuration

- Prerequisites for Configuring Call Control
- How to Configure Call Control for Voice Conferencing
- How to Configure Call Control for Video Conferencing with H.323 Endpoints

Contents

Prerequisites for Configuring Call Control

- Learn the benefits and restrictions of each supported call-control deployment option, and choose the best option for your Cisco Unified MeetingPlace system. Understand your deployment so that you know ahead of time which call-control devices you need to configure. See the *Planning Guide for Cisco Unified MeetingPlace* at http://docwiki.cisco.com/wiki/Cisco_Unified_MeetingPlace%2C_Release_8.0_--_Planning_Your_Deployment.
- Verify that the versions of your call-control devices and Cisco Unified MeetingPlace are compatible. See the *System Requirements for Cisco Unified MeetingPlace* at http://docwiki.cisco.com/wiki/Cisco_Unified_MeetingPlace_Release_8.0_--_System_Requirements_for_Cisco_U
- Install the call-control devices as described in the installation documentation for those devices.
- Verify that the Cisco Unified IP Phones and other endpoints are connected and added to the database of your call-control devices. See the *Compatibility Matrix for Cisco Unified MeetingPlace* at http://docwiki.cisco.com/wiki/Cisco_Unified_MeetingPlace_Release_8.0_--_Compatibility_Matrix_for_Cisco_Un
- Verify that you can place and receive internal and external calls on the Cisco Unified IP Phones and other endpoints.
- Install Cisco Unified MeetingPlace as described in the Quick Start for Installing and Configuring Cisco Unified MeetingPlace Release 8.0 module.

Related Topics

- Configuring Access Phone Numbers and Notification Labels module
- Configuring Direct Inward Dial for Cisco Unified MeetingPlace module

How to Configure Call Control for Voice Conferencing

- Configuring SIP on Cisco Unified MeetingPlace
- Configuring Cisco Unified Communications Manager Release 6.x or Later: SIP Trunk to Cisco Unified MeetingPlace
- Configuring Cisco Unified Communications Manager Release 6.x or Later: Route Patterns
- Configuring SIP Trunks Between Cisco Unified Communications Manager Release 5.x and Cisco Unified MeetingPlace Release 6.1 or Later
- Configuring Inter-Cluster Trunks Between Cisco Unified Communications Manager Release 4.x or 5.x and Cisco Unified Communications Manager Release 6.1 or Later
- Configuring Cisco Unified Communications Manager Release 4.x or 5.x: H.323 Trunk to Cisco Unified Border Element
- Configuring Cisco Unified Communications Manager Release 4.x or 5.x: Route Patterns
- Configuring Cisco Unified Border Element: H.323 to SIP Conversion
- Configuring the Cisco IOS Gateway: Dial Peers to Cisco Unified MeetingPlace
- Verifying the Call-Control Configuration

Configuring SIP on Cisco Unified MeetingPlace

Complete this task to connect Cisco Unified MeetingPlace to supported call-control devices.

Before You Begin

Complete the Prerequisites for Configuring Call Control.

Procedure

1. Sign in to the Administration Center.
2. Select **System Configuration** > **Call Configuration** > **SIP Configuration**.
3. Configure the fields on the SIP Configuration Page.
4. Select **Save**.

Related Topics

- Table: Field Reference: SIP Configuration Page in the Administration Center Page References for Cisco Unified MeetingPlace (R - S pages)

What to Do Next

Perform one of these actions:

- If your call-control network includes Cisco Unified Communications Manager Release 6.x (or later), proceed to the Configuring Cisco Unified Communications Manager Release 6.x or Later: SIP Trunk to Cisco Unified MeetingPlace.
- If your call-control network uses Cisco Unified Communications Manager Release 4.x or 5.x *without* Cisco Unified Communications Manager Release 6.1 (or later), proceed to the Configuring Cisco Unified Communications Manager Release 4.x or 5.x: Route Patterns.
- If your call-control network does not include Cisco Unified Communications Manager, proceed to the Configuring the Cisco IOS Gateway: Dial Peers to Cisco Unified MeetingPlace.

Configuring Cisco Unified Communications Manager Release 6.x or Later: SIP Trunk to Cisco Unified MeetingPlace

Before You Begin

- Complete the Configuring SIP on Cisco Unified MeetingPlace.
- We recommend that you configure a Calling Search Space in Cisco Unified Communications Manager that does the following:
    - Allows dial-out calls to meeting participants and the help desk Attendant.
    - Prevents toll fraud by blocking unwanted dial-out calls, for example, to international or premium-rate phone numbers.

  See the Administration Guide for your release of Cisco Unified Communications Manager at http://www.cisco.com/en/US/products/sw/voicesw/ps556/prod_maintenance_guides_list.html.

- We recommend that you create a SIP trunk security profile in Cisco Unified Communications Manager specifically for Cisco Unified MeetingPlace.

See Configuring a SIP Trunk Security Profile in Cisco Unified Communications Manager for Cisco Unified MeetingPlace in the Integrating Cisco Unified MeetingPlace with Cisco Unified Communications Manager module.

- If you want to prevent conference disruption by music when a user places a call on hold, complete the Configuring Cisco Unified Communications Manager: Music On Hold task in the Integrating Cisco Unified MeetingPlace with Cisco Unified Communications Manager module.
- You perform this task in the Cisco Unified Communications Manager Administration pages. Because the pages and menus vary by release, you should check the Cisco Unified Communications Manager Administration online help for step-by-step instructions that are specific to your release.

Procedure

1. Go to **http://***ccm-server***/**, where *ccm-server* is the fully-qualified domain name or IP address of the Cisco Unified Communications Manager server.
2. Sign in to Cisco Unified Communications Manager Administration.
3. Select **Device > Trunk**.
4. Select **Add New**.
5. In the Trunk type field, select **SIP Trunk**.
6. Select **Next**.
7. Configure the fields described in Table: Fields for Adding a SIP Trunk in Cisco Unified Communications Manager Release 6.x (or Later).

Table: Fields for Adding a SIP Trunk in Cisco Unified Communications Manager Release 6.x (or Later)

Field	Action
Device Name	Enter a unique identifier for this trunk, such as the name or IP address of the Cisco Unified MeetingPlace server.
Device Pool	The device pool must use a codec that is compatible with the conferencing gateway (or bridge).
AAR Group	For security and toll-fraud prevention, use a device pool and an automatic alternate routing (AAR) group that will block any undesired phone numbers from being dialed out.
Media Resource Group List	(Optional) If Cisco Unified MeetingPlace-supported endpoints are registered to this Cisco Unified Communications Manager, we recommend that you choose one of the following to prevent conference calls from being disrupted by music whenever a user places a call on hold: • Default value of <None>. • A Media Resource Group List that does *not* contain music on hold resources.
Media Termination Point Required	Uncheck this check box.
Destination Address	The DNS hostname or IP address of the Cisco Unified MeetingPlace Application Server.

	In an Application Server Failover deployment, make sure you enter the shared hostname and IP address of eth0.
Destination Port	Keep the default value of **5060**.
SIP Trunk Security Profile	Select the SIP trunk security profile that you created specifically for Cisco Unified MeetingPlace. If you did not create a SIP trunk security profile, select the default **Non Secure SIP Trunk Profile**.
Rerouting Calling Search Space	Make sure to set this field appropriately to ensure call transfers (out to attendant or to other systems) are successful. Consult the Cisco Unified Communications Manager administrator for the appropriate calling search space (CSS) to use. Usually, this will be the same CSS as the one used for Inbound Calling.
DTMF Signaling Method	Select **No Preference**.

8. Configure all other required fields appropriately for your current deployment.

   If you configured a Calling Search Space to block unwanted dial-out calls, apply the Calling Search Space accordingly to the SIP trunk.

   Tip: For field descriptions, select **Help > This Page**.
9. Select **Save**.

Related Topics

- Configuring Application Server Failover for Cisco Unified MeetingPlace module
- Integrating Cisco Unified MeetingPlace with Cisco Unified Communications Manager module
- Configuring Cisco Unified Communications Manager: Music On Hold in the Integrating Cisco Unified MeetingPlace with Cisco Unified Communications Manager module

What to Do Next

Proceed to the Configuring Cisco Unified Communications Manager Release 6.x or Later: Route Patterns.

Configuring Cisco Unified Communications Manager Release 6.x or Later: Route Patterns

Route patterns enable Cisco Unified Communications Manager to route calls to Cisco Unified MeetingPlace by associating phone numbers with the SIP trunk.

Before You Begin

- Complete the Configuring Cisco Unified Communications Manager Release 6.x or Later: SIP Trunk to Cisco Unified MeetingPlace.
- Write down each of the phone numbers from the Cisco Unified MeetingPlace Administration Center:
    - Access phone numbers configured on the Usage Configuration Page
    - Direct Inward Dial (DID) numbers-only if you enable DID through the Route calls to meeting ID that matches DID field
- You perform this task in the Cisco Unified Communications Manager Administration pages. Because the pages and menus vary by release, you should check the Cisco Unified Communications Manager Administration online help for step-by-step instructions that are specific to your release.

Procedure

1. Go to **http://***ccm-server***/**, where *ccm-server* is the fully-qualified domain name or IP address of the Cisco Unified Communications Manager server.
2. Sign in to Cisco Unified Communications Manager Administration.
3. Select **Call Routing** > **Route/Hunt > Route Pattern**.
4. Select **Add New**.
5. Configure the fields described in Table: Fields for Adding a Route Pattern in Cisco Unified Communications Manager Release 6.x (or Later).

Table: Fields for Adding a Route Pattern in Cisco Unified Communications Manager Release 6.x (or Later)

Field	Action
Route Pattern	Enter the Cisco Unified MeetingPlace phone number. Requirements: • This number must not conflict with any other route pattern defined in this Cisco Unified Communications Manager cluster. • Do not enter any spaces in this field.
Gateway/Route List	Select the Device Name of the SIP trunk to Cisco Unified MeetingPlace.
Call Classification	Select **OnNet**.
Provide Outside Dial Tone	Uncheck the check box.

6. Configure all other required fields appropriately for your current deployment.

    Tip: For field descriptions, select **Help > This Page**.
7. Select **Save**.
8. Select **OK** to any pop-up dialog box messages that you see.
9. Repeat this procedure as necessary to route calls to each access phone number and DID number for your Cisco Unified MeetingPlace system.

Related Topics

- SIP Configuration Page in the Administration Center Page References for Cisco Unified MeetingPlace (R - S pages)
- Configuring Access Phone Numbers and Notification Labels module
- Configuring Direct Inward Dial for Cisco Unified MeetingPlace module

What to Do Next

If you are using Cisco Unified Communications Manager Release 6.1 (or later) to provide front-end signaling for Cisco Unified MeetingPlace, proceed to one of these sections:

- Configuring SIP Trunks Between Cisco Unified Communications Manager Release 5.x and Cisco Unified MeetingPlace Release 6.1 or Later
- Configuring Inter-Cluster Trunks Between Cisco Unified Communications Manager Release 4.x or 5.x and Cisco Unified Communications Manager Release 6.1 or Later

Otherwise, proceed to the Verifying the Call-Control Configuration.

Configuring SIP Trunks Between Cisco Unified Communications Manager Release 5.x and Cisco Unified MeetingPlace Release 6.1 or Later

Perform this task if you have already deployed Cisco Unified Communications Manager Release 5.x and are using Cisco Unified Communications Manager Release 6.1 (or later) to provide front-end signaling for Cisco Unified MeetingPlace.

Before You Begin

- You can instead choose to configure inter-cluster trunks (instead of SIP trunks) between Cisco Unified Communications Manager Release 5.x and Cisco Unified Communications Manager Release 6.1 (or later). If this is the case, do not perform this task. Instead, see the Configuring Inter-Cluster Trunks Between Cisco Unified Communications Manager Release 4.x or 5.x and Cisco Unified Communications Manager Release 6.1 or Later.
- Complete the Configuring Cisco Unified Communications Manager Release 6.x or Later: Route Patterns.
- Perform this task on *both* of these servers:
    - Cisco Unified Communications Manager Release 5.x
    - Cisco Unified Communications Manager Release 6.1 (or later)
- We recommend that you configure a Calling Search Space in Cisco Unified Communications Manager that does the following:
    - Allows dial-out calls to meeting participants and the help desk Attendant.
    - Prevents toll fraud by blocking unwanted dial-out calls, for example, to international or premium-rate phone numbers.

  See the Administration Guide for your release of Cisco Unified Communications Manager at http://www.cisco.com/en/US/products/sw/voicesw/ps556/prod_maintenance_guides_list.html.

- You perform this task in the Cisco Unified Communications Manager Administration pages. Because the pages and menus vary by release, you should check the Cisco Unified Communications Manager Administration online help for step-by-step instructions that are specific to your release.

Procedure

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 Communications Manager server.
2. Sign in to Cisco Unified Communications Manager Administration.
3. Select **Device > Trunk**.
4. Select **Add New**.
5. In the Trunk Type field, select **SIP Trunk**.
6. Select **Next**.
7. Configure the fields described in Table: Fields for Adding a SIP Trunk in Cisco Unified Communications Manager.

Table: Fields for Adding a SIP Trunk in Cisco Unified Communications Manager

Field	Action
Device Name	Enter a unique identifier for this trunk, for example: • If you are configuring Cisco Unified Communications Manager Release 5.x, enter the name or IP address of the Cisco Unified Communications Manager Release 6.1 (or later) server that provides front-end signaling for Cisco Unified MeetingPlace. • If you are configuring Cisco Unified Communications Manager Release 6.1 (or later), enter the name or IP address of the Cisco Unified Communications Manager Release 5.x server.
Device Pool	If no device pools are defined, select **Default**. If device pools are already defined, 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 (or bridge).
Media Resource Group List	(Optional) If Cisco Unified MeetingPlace-supported endpoints are registered to this Cisco Unified Communications Manager, we recommend that you choose one of the following to prevent conference calls from being disrupted by music whenever a user places a call on hold: • Default value of <None>. • A Media Resource Group List that does *not* contain music on hold resources.
Media Termination Point Required	Uncheck this check box.
Destination Address	Enter the destination IP address, specifically: • If you are configuring Cisco Unified Communications Manager Release 5.x, enter the IP address of the Cisco Unified

	Communications Manager Release 6.1 (or later) server that provides front-end signaling for Cisco Unified MeetingPlace. • If you are configuring Cisco Unified Communications Manager Release 6.1 (or later), enter the name or IP address of the Cisco Unified Communications Manager Release 5.x server.
Destination Port Incoming Port	Keep the default value of **5060**.

8. Configure all other required fields appropriately for your current deployment.
If you configured a Calling Search Space to block unwanted dial-out calls, apply the Calling Search Space accordingly to the SIP trunk.
Tip: For field descriptions, select **Help > This Page**.
9. Select **Save**.
10. Repeat this task so that *both* of these servers are configured with SIP trunks that point to each other:
    - Cisco Unified Communications Manager Release 5.x
    - Cisco Unified Communications Manager Release 6.1 (or later)

Related Topics

- Configuring Operator Assistance in the Configuring Attendant Settings for Cisco Unified MeetingPlace module
- Configuring Cisco Unified Communications Manager: Music On Hold in the Integrating Cisco Unified MeetingPlace with Cisco Unified Communications Manager module

What to Do Next

Proceed to the Configuring Cisco Unified Communications Manager Release 4.x or 5.x: Route Patterns.

Configuring Inter-Cluster Trunks Between Cisco Unified Communications Manager Release 4.x or 5.x and Cisco Unified Communications Manager Release 6.1 or Later

Perform this task if you already deployed Cisco Unified Communications Manager Release 4.x or 5.x and are using Cisco Unified Communications Manager Release 6.1 (or later) to provide front-end signaling for Cisco Unified MeetingPlace.

Before You Begin

- You can instead choose to configure SIP trunks (instead of inter-cluster trunks) between Cisco Unified Communications Manager Release 5.x and Cisco Unified MeetingPlace Release 6.1 (or later). If this is the case, do not perform this task. Instead, see the Configuring SIP Trunks Between Cisco Unified Communications Manager Release 5.x and Cisco Unified MeetingPlace Release 6.1 or Later.

- Complete the Configuring Cisco Unified Communications Manager Release 6.x or Later: Route Patterns.
- Perform this task on *both* of these servers:
    - Cisco Unified Communications Manager Release 4.x or 5.x
    - Cisco Unified Communications Manager Release 6.1 (or later)
- We recommend that you configure a Calling Search Space in Cisco Unified Communications Manager that does the following:
    - Allows dial-out calls to meeting participants and the help desk Attendant.
    - Prevents toll fraud by blocking unwanted dial-out calls, for example, to international or premium-rate phone numbers.

  See the Administration Guide for your release of Cisco Unified Communications Manager at http://www.cisco.com/en/US/products/sw/voicesw/ps556/prod_maintenance_guides_list.html.

- You perform this task in the Cisco Unified Communications Manager Administration pages. Because the pages and menus vary by release, you should check the Cisco Unified Communications Manager Administration online help for step-by-step instructions that are specific to your release.

Procedure

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 Communications Manager server.
2. Sign in to Cisco Unified Communications Manager Administration.
3. Select **Device > Trunk**.
4. Select **Add New**.
5. In the Trunk type field, select **Inter-Cluster Trunk (Non-Gatekeeper Controlled)**.
6. Select **Next**.
7. Configure the fields described in Table: Fields for Adding an Inter-Cluster Trunk in Cisco Unified Communications Manager.

Table: Fields for Adding an Inter-Cluster Trunk in Cisco Unified Communications Manager

Field	Action
Device Name	Enter a unique identifier for this trunk, for example: • If you are configuring Cisco Unified Communications Manager Release 4.x or 5.x, enter the name or IP address of the Cisco Unified Communications Manager Release 6.1 (or later) server that provides front-end signaling for Cisco Unified MeetingPlace. • If you are configuring Cisco Unified Communications Manager Release 6.1 (or later), enter the name or IP address of the Cisco Unified Communications Manager Release 4.x or 5.x server.
Device Pool	If no device pools are defined, select **Default**. If device pools are already defined, 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 (or bridge).

Media Resource Group List	(Optional) If Cisco Unified MeetingPlace-supported endpoints are registered to this Cisco Unified Communications Manager, we recommend that you choose one of the following to prevent conference calls from being disrupted by music whenever a user places a call on hold: • Default value of <None>. • A Media Resource Group List that does *not* contain music on hold resources.
Media Termination Point Required	Uncheck this check box.
Server 1 IP Address/Host Name	Identify the target server, specifically: • If you are configuring Cisco Unified Communications Manager Release 4.x or 5.x, specify the Cisco Unified Communications Manager Release 6.1 (or later) server that provides front-end signaling for Cisco Unified MeetingPlace. • If you are configuring Cisco Unified Communications Manager Release Release 6.1 (or later), specify the Cisco Unified Communications Manager Release 4.x or 5.x server.

8. Configure all other required fields appropriately for your current deployment.
If you configured a Calling Search Space to block unwanted dial-out calls, apply the Calling Search Space accordingly to the SIP trunk.
Tip: For field descriptions, select **Help > This Page**.
9. Select **Save**.
10. Repeat this task so that *both* of the servers are configured with SIP trunks that point to each other:
    - Cisco Unified Communications Manager Release 4.x or 5.x
    - Cisco Unified Communications Manager Release 6.1 (or later)

Related Topics

- Configuring Cisco Unified Communications Manager: Music On Hold in the Integrating Cisco Unified MeetingPlace with Cisco Unified Communications Manager module

What to Do Next

Proceed to the Configuring Cisco Unified Communications Manager Release 4.x or 5.x: Route Patterns.

Configuring Cisco Unified Communications Manager Release 4.x or 5.x: H.323 Trunk to Cisco Unified Border Element

Before You Begin

- Complete the Configuring SIP on Cisco Unified MeetingPlace.
- This task is performed in the Cisco Unified Communications Manager administration interface. Because the pages and menus vary by release, you might need to see the Cisco Unified Communications Manager online help for more accurate step-by-step instructions than those provided in this procedure.

Procedure

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 Communications Manager server.
2. Sign in to Cisco Unified Communications Manager Administration.
3. Select **Device > Trunk**.
4. Select **Add New**.
5. In the Trunk type field, select **Inter-Cluster Trunk (Non-Gatekeeper Controlled)**.
6. Select **Next**.
7. Configure the fields described in Table: Fields for Adding an Inter-Cluster Trunk in Cisco Unified Communications Manager Release 4.x or 5.x.

Table: Fields for Adding an Inter-Cluster Trunk in Cisco Unified Communications Manager Release 4.x or 5.x

Field	Action
Device Name	Enter a unique identifier for this trunk, such as the name or IP address of the Cisco Unified Communications Manager Release 6.1 (or later) server that provides front-end signaling for Cisco Unified MeetingPlace.
Device Pool	If no device pools are defined, select **Default**. If the Cisco Unified Communications Manager deployment uses customer-defined device pools, 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 (or bridge).
Media Termination Point Required	Uncheck this check box.
Server 1 IP Address/Host Name	Identify the Cisco Unified Communications Manager Release 6.1 (or later) server that provides front-end signaling for Cisco Unified MeetingPlace.

8. Configure all other required fields appropriately for your current deployment.

   Tip: For field descriptions, select **Help > This Page**.
9. Select **Save**.

What to Do Next

Proceed to the Configuring Cisco Unified Communications Manager Release 4.x or 5.x: Route Patterns.

Configuring Cisco Unified Communications Manager Release 4.x or 5.x: Route Patterns

Use this procedure to configure route patterns to enable Cisco Unified Communications Manager Release 4.x or 5.x to route calls that are placed to Cisco Unified MeetingPlace phone numbers. The route patterns associate the Cisco Unified MeetingPlace phone numbers with one of the following, depending on your deployment:

- Inter-cluster trunk to the Cisco Unified Communications Manager Release 6.1 (or later) server that provides front-end signaling for Cisco Unified MeetingPlace
- (Cisco Unified Communications Manager Release 5.x only) SIP trunk to the Cisco Unified Communications Manager Release 6.1 (or later) server that provides front-end signaling for Cisco Unified MeetingPlace
- Cisco IOS gateway with Cisco Unified Border Element

Before You Begin

- Complete the Configuring SIP on Cisco Unified MeetingPlace.
- Complete one of these items, depending on your deployment:
    - Configuring SIP Trunks Between Cisco Unified Communications Manager Release 5.x and Cisco Unified MeetingPlace Release 6.1 or Later
    - Configuring Inter-Cluster Trunks Between Cisco Unified Communications Manager Release 4.x or 5.x and Cisco Unified Communications Manager Release 6.1 or Later
    - Make sure that your Cisco Unified Communications Manager configuration database already includes a gateway entry for the Cisco Unified Border Element. See the Cisco Unified Communications Manager online help for information about finding or adding gateways in Cisco Unified Communications Manager.
- Write down each of the phone numbers from the Cisco Unified MeetingPlace Administration Center:
    - Access phone numbers configured on the Usage Configuration Page
    - Direct Inward Dial (DID) numbers-only if you enable DID through the Route calls to meeting ID that matches DID field
- You perform this task in the Cisco Unified Communications Manager Administration pages. Because the pages and menus vary by release, you should check the Cisco Unified Communications Manager Administration online help for step-by-step instructions that are specific to your release.

Procedure

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 Communications Manager server.
2. Sign in to Cisco Unified Communications Manager Administration.
3. Select **Call Routing** > **Route/Hunt > Route Pattern**.
4. Select **Add New**.
5. Configure the fields described in Table: Fields for Adding a Route Pattern in Cisco Unified Communications Manager Release 4.x or 5.x.

Table: Fields for Adding a Route Pattern in Cisco Unified Communications Manager Release 4.x or 5.x

Field	Action
Route Pattern	Enter the Cisco Unified MeetingPlace phone number. Requirements: • This number must not conflict with any other route pattern defined in this Cisco Unified Communications Manager cluster. • Do not enter any spaces in this field.
	Select the Device Name of one of the following, depending on your deployment:

Gateway/Route List	• Inter-cluster trunk to the Cisco Unified Communications Manager Release 6.1 (or later) server that provides front-end signaling for Cisco Unified MeetingPlace. • (Cisco Unified Communications Manager Release 5.x only) SIP trunk to the Cisco Unified Communications Manager Release 6.1 (or later) server that provides front-end signaling for Cisco Unified MeetingPlace. Cisco IOS gateway with Cisco Unified Border Element
Call Classification	Select **OffNet**.

6. Configure all other required fields appropriately for your current deployment.

   Tip: For field descriptions, select **Help > This Page**.
7. Select **Save**.
8. Select **OK** to any pop-up dialog box messages that you see.
9. Repeat this procedure as necessary to route calls to each access phone number and DID number for your Cisco Unified MeetingPlace system.

Related Topics

- Configuring Access Phone Numbers and Notification Labels module
- Configuring Direct Inward Dial for Cisco Unified MeetingPlace module

What to Do Next

Take one of these actions:

- If your route patterns direct calls to Cisco Unified Communications Manager Release 6.1 (or later), proceed to the Verifying the Call-Control Configuration.
- If your route patterns direct calls to a Cisco IOS gateway with Cisco Unified Border Element, proceed to the Configuring Cisco Unified Border Element: H.323 to SIP Conversion.
- Proceed to the Verifying the Call-Control Configuration.

Configuring Cisco Unified Border Element: H.323 to SIP Conversion

Perform this task to support H.323 endpoints if your call-control network does not include Cisco Unified Communications Manager 6.1 (or later).

Note: Cisco Unified Border Element was previously known as the Cisco Multiservice IP-to-IP Gateway.

Before You Begin

- Complete the Configuring Cisco Unified Communications Manager Release 4.x or 5.x: Route Patterns.
- 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 one of these documents:

    - *Cisco Unified Border Element Configuration Guide* for your Cisco IOS release
    - *Cisco IOS Commands Master List* for your Cisco IOS release

Restriction

CUBE is not compatible with Cisco Unified MeetingPlace systems that include one or more video blades.

Procedure

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
   ```
2. Enter global configuration mode.
   ```
   Router# configure terminal
   ```
3. Enable basic CUBE functionality on the router. This functionality terminates an incoming VoIP call and re-originates it with the use of an outbound VoIP dial peer. The calls can be H.323 to SIP or SIP to SIP.
   ```
   Router(config)# voice service voip
   Router(config-voi-serv)# allow-connections h323 to sip
   Router(config-voi-serv)# allow-connections sip to h323
   Router(config-voi-serv)# allow-connections sip to sip
   Router(config-voi-serv)# allow-connections h323 to h323
   ```

What to Do Next

Proceed to the Configuring the Cisco IOS Gateway: Dial Peers to Cisco Unified MeetingPlace.

Configuring the Cisco IOS Gateway: Dial Peers to Cisco Unified MeetingPlace

Use this procedure to enable the Cisco IOS Gateway to route calls to Cisco Unified MeetingPlace by configuring dial peers. 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

- Complete the Configuring SIP on Cisco Unified MeetingPlace.
- Make sure that your Cisco Unified Communications Manager configuration database already includes the Cisco IOS gateway. See the Cisco Unified Communications Manager online help for information about finding or adding gateways in Cisco Unified Communications Manager.
- Write down each of the phone numbers from the Cisco Unified MeetingPlace Administration Center:
    - Access phone numbers configured on the Usage Configuration Page
    - Direct Inward Dial (DID) numbers-only if you enable DID through the Route calls to meeting ID that matches DID field
- You perform this task 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

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`
2. Enter global configuration mode.
   `Router# configure terminal`
3. Enter dial peer voice configuration mode and define a remote voice over IP (VoIP) dial peer.
   `Router(config)# dial-peer voice number voip`
   The *number* is one or more digits that identify the dial peer. Valid entries are from 1 to 2147483647.
4. (Optional) Provide a comment or description to help you distinguish this particular dial peer from other dial peer configurations.
   `Router(config-dialpeer)# description string`
5. Route calls to the Cisco Unified MeetingPlace Application Server.
   `Router(config-dialpeer)# destination-pattern digits`
   For *digits*, enter the Cisco Unified MeetingPlace phone number.
6. Configure the dial peer to use SIP.
   `Router(config-dialpeer)# session protocol sipv2`
7. Configure the IP address of the Cisco Unified MeetingPlace Application Server.
   `Router(config-dialpeer)# session target ipv4:ip-address`
   In an Application Server failover deployment, enter the shared IP address of eth0.
8. Configure the router to forward dual tone multifrequency (DTMF) tones by using Real-Time Transport Protocol (RTP) with the Named Telephone Event (NTE) payload type.
   `Router(config-dialpeer)# dtmf-relay rtp-nte`
9. Configure the router to use a particular codec.
   `Router(config-dialpeer)# codec [g711ulaw | g711alaw | g729 | g722-64 | ilbc]`
10. Disable voice activity detection (VAD) for the calls using this dial peer.
    `Router(config-dialpeer)# no vad`
11. Exit the current mode.
    `Router(config-dialpeer)# exit`
12. Repeat this procedure as necessary to route calls to each access phone number and DID number for your Cisco Unified MeetingPlace system.

Example

This example displays dial peers that were configured to direct calls to two Cisco Unified MeetingPlace access phone numbers. The Cisco Unified MeetingPlace Application Server IP address is 10.10.10.4.

```
 !

dial-peer voice 1 voip

     description Cisco Unified MeetingPlace access phone number 1

     destination-pattern 50111

     session protocol sipv2

     session target ipv4:10.10.10.4
```

```
      dtmf-relay rtp-nte

      codec g711ulaw

      no vad

 !

dial-peer voice 2 voip

      description Cisco Unified MeetingPlace access phone number 2

      destination-pattern 50123

      session protocol sipv2

      session target ipv4:10.10.10.4

      dtmf-relay rtp-nte

      codec g711ulaw

      no vad

 !
```

Related Topics

- Configuring Access Phone Numbers and Notification Labels module
- Configuring Direct Inward Dial for Cisco Unified MeetingPlace module
- Configuring Application Server Failover for Cisco Unified MeetingPlace module

What to Do Next

Proceed to the Verifying the Call-Control Configuration.

Verifying the Call-Control Configuration

Procedure

1. Call one of the Cisco Unified MeetingPlace access phone numbers configured on the Usage Configuration Page of the Administration Center.
2. Verify that you hear the Cisco Unified MeetingPlace voice prompts.

Troubleshooting Tips

See the Troubleshooting Phone Issues for Cisco Unified MeetingPlace module.

Related Topics

- Verifying Basic Voice and Video Conferencing Using the Telephone User Interface in the Quick Start Configuration for Cisco Unified MeetingPlace Basic Voice and Video Conferencing module
- Verifying Basic Voice and Video Conferencing Using the Web User Portal in the Quick Start Configuration for Cisco Unified MeetingPlace Basic Voice and Video Conferencing module
- Configuring Access Phone Numbers and Notification Labels module
- Configuring Direct Inward Dial for Cisco Unified MeetingPlace module

What To Do Next

- If your network includes H.323 video endpoints, proceed to the How to Configure Call Control for Video Conferencing with H.323 Endpoints.
- For Cisco Unified Communications Manager environments, we recommend disabling the Music on Hold (MoH) feature for Cisco Unified MeetingPlace. See Configuring Cisco Unified Communications Manager: Music On Hold in the Integrating Cisco Unified MeetingPlace with Cisco Unified Communications Manager module.

How to Configure Call Control for Video Conferencing with H.323 Endpoints

- How to Configure the Cisco IOS Gatekeeper
- Configuring Cisco Unified Communications Manager Release 6.1 or Later: H.323 Endpoints

How to Configure the Cisco IOS Gatekeeper

Perform one of these tasks, depending on which device provides dial plan resolution for your network:

- Configuring the Cisco IOS Gatekeeper: Dial Plan Resolution Provided By Gatekeeper
- Configuring the Cisco IOS Gatekeeper: Dial Plan Resolution Provided by Cisco Unified Communications Manager

Configuring the Cisco IOS Gatekeeper: Dial Plan Resolution Provided By Gatekeeper

Choose this configuration option if the Cisco IOS gatekeeper provides dial plan resolution for all devices in your network.

Before You Begin

- Configure voice call control. See the How to Configure Call Control for Voice Conferencing.
- Configure the gatekeeper Cisco Unified Communications Manager. For instructions on adding the gatekeeper, see the Cisco Unified Communications Manager online help.
- You perform this task 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 release.

Procedure

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`**
2. Enter global configuration mode.
   `Router#` **`configure terminal`**
3. Enter gatekeeper configuration mode.
   `Router#` **`gatekeeper`**
4. Define the zone controlled by the gatekeeper.
   `Router(config-gk)#` **`zone local`** *`gk-zone-name domain-name`*
5. Specify which subnets the gatekeeper will accept discovery and registration messages sent by endpoints in those subnets.
   `Router(config-gk)#` **`no zone subnet`** *`gk-zone-name`* **`default enable`**
   `Router(config-gk)#` **`zone subnet`** *`gk-zone-name subnet1-address`*`{`**`/`***`bits-in-mask`*` | `*`mask-address`*`}` **`enable`**
   `Router(config-gk)#` **`zone subnet`** *`gk-zone-name subnet2-address`*`{`**`/`***`bits-in-mask`*` | `*`mask-address`*`}` **`enable`**
6. Define a technology prefix, which is stripped before checking for the zone prefix. Configure calls to hop off at the gatekeeper, regardless of the zone prefix in the destination address. The **default-technology** option specifies to use gateways registering with this prefix option as the default for routing any addresses that are otherwise unresolved.
   `Router(config-gk)#` **`gw-type-prefix`** *`type-prefix`* **`hopoff`** *`gk-zone-name`* **`default-technology`**
7. Disable proxy communications with local terminals for calls between local and remote zones.
   `Router(config-gk)#` **`no use-proxy`** *`gk-zone-name`* **`default inbound-to terminal`**
   `Router(config-gk)#` **`no use-proxy`** *`gk-zone-name`* **`default outbound-from terminal`**
8. Enable the gatekeeper.
   `Router(config-gk)#` **`no shutdown`**

Example

```
 !

gatekeeper

     zone local mp2-video example.com

     no zone subnet mp2-video default enable
```

```
      zone subnet mp2-video 10.20.120.50/32 enable

      zone subnet mp2-video 10.10.1.0/24 enable

      gw-type-prefix 2#* hopoff mp2-video default-technology

      no use-proxy mp2-video default inbound-to terminal

      no use-proxy mp2-video default outbound-from terminal

      no shutdown

 !
```

What to Do Next

Proceed to one of these sections in the Quick Start Configuration for Cisco Unified MeetingPlace Basic Voice and Video Conferencing module:

- Verifying Basic Voice and Video Conferencing Using the Telephone User Interface
- Verifying Basic Voice and Video Conferencing Using the Web User Portal

Configuring the Cisco IOS Gatekeeper: Dial Plan Resolution Provided by Cisco Unified Communications Manager

Choose this configuration option if the following are true:

- All H.323 video endpoints register to this Cisco IOS gatekeeper.
- Cisco Unified Communications Manager provides dial plan resolution and becomes the master call-control point for all devices in your network.

Before You Begin

- Configure voice call control. See the How to Configure Call Control for Voice Conferencing.
- Add this gatekeeper to Cisco Unified Communications Manager. For instructions on adding the gatekeeper, see the Cisco Unified Communications Manager online help.
- You perform this task 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 release.

Procedure

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`
2. Enter global configuration mode.
   `Router# configure terminal`

3. Enter gatekeeper configuration mode.
   Router# **gatekeeper**
4. Define the zone controlled by the gatekeeper. Specify which gatekeeper interface to use for Registration, Admission, and Status (RAS) signaling. Force all intra-zone calls, in addition to calls that enter and leave the zone, to use this gatekeeper.
   Router(config-gk)# **zone local** *gk-zone-name domain-name ras-IP-address* **invia** *gk-zone-name* **outvia** *gk-zone-name* **enable-intrazone**
5. Define a technology prefix that matches what you configure for the H.323 endpoints in Cisco Unified Communications Manager. The **default-technology** option specifies to use gateways registering with this prefix option as the default for routing any addresses that are otherwise unresolved.
   Router(config-gk)# **gw-type-prefix** *type-prefix* **default-technology**
6. Disable proxy communications with local terminals for calls between local and remote zones.
   Router(config-gk)# **no use-proxy** *local-zone-name* **default inbound-to terminal**
   Router(config-gk)# **no use-proxy** *local-zone-name* **default outbound-from terminal**
7. Enable the gatekeeper.
   Router(config-gk)# **no shutdown**

Example

```
 !

gatekeeper

     zone local MP-Zone1 example.net 192.168.2.50 invia MP-Zone1 outvia
     MP-Zone1 enable-intrazone

     gw-type-prefix 1#* default-technology

     no use-proxy MP-Zone1 default inbound-to terminal

     no use-proxy MP-Zone1 default outbound-from terminal

     no shutdown

 !
```

What To Do Next

Proceed to the Configuring Cisco Unified Communications Manager Release 6.1 or Later: H.323 Endpoints.

Configuring Cisco Unified Communications Manager Release 6.1 or Later: H.323 Endpoints

Perform this task only if Cisco Unified Communications Manager provides dial plan resolution for your network.

Before You Begin

- Complete the Configuring the Cisco IOS Gatekeeper: Dial Plan Resolution Provided by Cisco Unified Communications Manager.
- You perform this task in the Cisco Unified Communications Manager Administration pages. Because the pages and menus vary by release, you should check the Cisco Unified Communications Manager Administration online help for step-by-step instructions that are specific to your release.

Procedure

1. Go to **http://***ccm-server***/**, where *ccm-server* is the fully-qualified domain name or IP address of the Cisco Unified Communications Manager server.
2. Sign in to Cisco Unified Communications Manager Administration.
3. Select **Device > Phone**.
4. (Optional) To display a list of existing phone entries, select **Find**.
5. Select **Add New**.
6. Select **H.323 Client** in the Phone Type field.
7. Select **Next**.
8. Configure the fields described in Table: Fields for Adding an H.323 Endpoint in Cisco Unified Communications Manager Release 6.1 (or Later).

Table: Fields for Adding an H.323 Endpoint in Cisco Unified Communications Manager Release 6.1 (or Later)

Field	Action
Device Name	Enter the IP address of the H.323 endpoint.
Device Pool	Select **Default** or your configured device pool.
Outgoing Caller ID Pattern	Enter the extension or phone number of the H.323 endpoint.
Gatekeeper Name	Select the gatekeeper that you configured in the Configuring the Cisco IOS Gatekeeper: Dial Plan Resolution Provided by Cisco Unified Communications Manager.
E.164	Enter the E.164 phone number used by the H.323 endpoint.
Technology Prefix	Enter the technology prefix (*type-prefix*) that you configured in Step 5 in the Configuring the Cisco IOS Gatekeeper: Dial Plan Resolution Provided by Cisco Unified Communications Manager.
Zone	Enter the zone name (*gk-zone-name*) that you configured in Step 4 in the Configuring the Cisco IOS Gatekeeper: Dial Plan Resolution Provided by Cisco Unified Communications Manager.

9. Configure all other required fields appropriately for your current deployment.

   Tip: For field descriptions, select **Help > This Page**.
10. Select **Save**.
11. Select **OK** to any pop-up dialog box messages that you see.
12. Repeat this procedure to add each H.323 endpoint to Cisco Unified Communications Manager.

What to Do Next

Proceed to one of these sections in the Quick Start Configuration for Cisco Unified MeetingPlace Basic Voice and Video Conferencing module:

- Verifying Basic Voice and Video Conferencing Using the Telephone User Interface
- Verifying Basic Voice and Video Conferencing Using the Web User Portal