Main page: Cisco Unified MeetingPlace Express, Release 2.x

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

Contents

Table: Task Roadmap

Task	Reference
Configure your particular call-control device.	See one of the following depending on your call-control device: • If you are using Cisco Unified Communications Manager, see the Configuring Cisco Unified Communications Manager: Adding the SIP Trunk and Route Pattern. • If you are using Cisco Unified Communications Manager Express or a Cisco IOS software voice-enabled router, see the Configuring Cisco Unified Communications Manager Express and Other Cisco IOS Software Voice-Enabled Routers: Adding the SIP Gateway and Dial Peer. • If you are using Cisco SIP Proxy Server, see the Configuring the Cisco SIP Proxy Server
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.

Configuring Cisco Unified Communications Manager: Adding the SIP Trunk and Route Pattern

Perform one of the following tasks, depending on which version of Cisco Unified Communications Manager you are using:

- Adding the SIP Trunk and Route Pattern in Cisco Unified CallManager Release 4.1
- Adding the SIP Trunk and Route Pattern in Cisco Unified Communications Manager Release 5.x

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 Release 4.1 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:
    - Integrating in a SIP Environment
    - Prerequisites for Integrating in a SIP Environment
    - Restrictions for Integrating in a SIP Environment

Restriction

- This task is performed in the Cisco Unified CallManager pages. Because the pages and menus vary by Cisco Unified CallManager release, you may need to see the Cisco Unified CallManager 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 CallManager server.
2. Log in with your Cisco Unified CallManager administrator username and password.
3. Add a new SIP trunk by completing the following actions:

a. 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. Configure the fields described in Table: Fields for Adding a New Trunk to Cisco Unified CallManager.

Table: 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	See Device Pool.
Media Termination Point Required	This box is always checked by default and cannot be unchecked. Video is not supported with Cisco Unified CallManager Release 4.x.
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**.

Incoming Port	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**.

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

a. Click **Route Plan** > **Route/Hunt > Route Pattern**.

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

c. Configure the fields described in Table: Fields for Adding a New Route Pattern to Cisco Unified CallManager.

Table: 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**.

Related Topics

- Field Reference: SIP Configuration

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

Note: The names for Cisco Unified CallManager Release 4.3, Release 5.1, and Release 6.0 have been changed to Cisco Unified Communications Manager Release 4.3, Release 5.1, and Release 6.0.

The names of Cisco Unified CallManager Release 4.0, Release 4.1, Release 4.2, and Release 5.0 have *not* changed and remain the same.

This topic describes how to add the SIP trunk to the Cisco Unified Communications Manager configuration database. This topic also describes how to enable Cisco Unified Communications Manager 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:
    - Integrating in a SIP Environment
    - Prerequisites for Integrating in a SIP Environment
    - Restrictions for Integrating in a SIP Environment

Restriction

- This task is performed in the Cisco Unified Communications Manager pages. Because the pages and menus vary by Cisco Unified Communications Manager release, you may need to see the Cisco Unified Communications Manager online help for more accurate step-by-step instructions than those provided in this procedure. The following procedure was created using Cisco Unified Communications Manager Release 5.0(3).

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. Log in with your Cisco Unified Communications Manager administrator username and password.
3. Add a new SIP trunk by completing the following actions:

a. 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. Configure the fields described in Table: Fields for Adding a New Trunk to Cisco Unified Communications Manager.

Table: Fields for Adding a New Trunk to Cisco Unified Communications Manager

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	See Device Pool.
Media Termination Point Required	Uncheck this check box if you want video to work. Video will not work if this box is checked. Checking this box prevents calls from supporting video. MTP should only be used when necessary.
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, 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 Communications Manager deployment. For information about each field, see the Cisco Unified Communications Manager online help or see the administration guide for your specific Cisco Unified Communications Manager release.

h. Click **Save**.

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

a. 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. Configure the fields described in Table: Fields for Configuring the Standard SIP Profile in Cisco Unified Communications Manager.

Table: Fields for Configuring the Standard SIP Profile in Cisco Unified Communications Manager

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 check box is not selected.

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

f. Click **Save**.

5. Configure the nonsecure SIP trunk profile by completing the following actions:

a. Click **System > Security Profile > SIP Trunk Security Profile**.

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

c. Under Search Results, click **Non Secure SIP Trunk Profile**.

d. Configure the fields described in Table: Fields for Configuring the Non Secure SIP Trunk Profile in Cisco Unified Communications Manager.

Table: Fields for Configuring the Non Secure SIP Trunk Profile in Cisco Unified Communications Manager

SIP Trunk Security Profile Information Field	Action
Incoming Transport Type	Keep the default value of **TCP+UDP**.
Outgoing Transport Type	Select **UDP**.

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

f. Click **Save**.

6. Add the route pattern to the Cisco Unified Communications Manager database by completing the following actions:

a. Click **Call Routing** > **Route/Hunt > Route Pattern**.

b. Click **Add New**.

c. Configure the fields described in Table: Fields for Adding a New Route Pattern to Cisco Unified Communications Manager.

Table: Fields for Adding a New Route Pattern to Cisco Unified Communications Manager

Route Pattern Configuration Field	Action
Route Pattern	See Route Pattern.
Gateway/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 Communications Manager deployment. For information about each field, see the Cisco Unified Communications Manager online help or see the administration guide for your specific Cisco Unified Communications Manager release.

e. Click **Save**.

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

Related Topics

- Field Reference: SIP Configuration

Configuring Cisco Unified Communications Manager 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 Communications Manager 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 three tasks:

- Shutting Down or Enabling Voice over IP (VoIP) Service on the Cisco Gateway
- Configuring SIP Server Support
- Configuring SIP Support for Voice Dial Peers

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

Before You Begin

- Read the following sections:
    - Integrating in a SIP Environment
    - Prerequisites for Integrating in a SIP Environment

Restriction

- 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

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 voice-service VoIP configuration mode.
   `Router(config)# voice service voip`
4. Enter SIP configuration mode.
   `Router(config-voi-serv)# sip`
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`
6. Exit the current mode.
   `Router(config-serv-sip)# exit`
7. Proceed to the Configuring SIP Server Support.

Configuring SIP Server Support

Before You Begin

- Complete the task described in the Shutting Down or Enabling Voice over IP (VoIP) Service on the Cisco Gateway.

Restriction

- 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

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 SIP user-agent configuration mode.
   `Router(config)#` **`sip-ua`**
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.
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`*`]`
6. Exit the current mode.
   `Router(config-sip-ua)#` **`exit`**
7. 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
   ```
8. Proceed to the Configuring SIP Support for Voice Dial Peers.

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:
  - Integrating in a SIP Environment
  - Prerequisites for Integrating in a SIP Environment
- Complete the Configuring SIP Server Support.

Restriction

- 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

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`
   - *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.
4. Enter the session protocol type.
   `Router(config-dialpeer)# session protocol sipv2`
   - **sipv2**-Configures the dial peer to use IETF SIP.
5. Configure the router to use a particular codec.
   `Router(config-dialpeer)# codec [g711ulaw | g711alaw | g729r8]}`
6. Configure the router to use dual tone multifrequency (DTMF) relay to transport DTMF digits.
   `Router(config-dialpeer)# dtmf-relay rtp-nte`
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.
8. Disable voice activity detection (VAD) for the calls using this dial peer.
   `Router(config-dialpeer)# [no] vad`
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
```

Configuring the 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
- Configuring Cisco SIP Proxy Server: Configuring a Dynamic or Static Route

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

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 SPS.

Before You Begin

- Install and configure your Cisco SPS by using the instructions in the Cisco SPS documentation.

Procedure

1. From the Cisco SPS main menu, click **Subscribers**.
2. Click **Add**.
3. Enter Cisco Unified MeetingPlace Express information.
4. Click **Submit**.

Configuring Cisco SIP Proxy Server: Configuring a Dynamic or Static Route

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 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 and 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 SPS documentation.
- Complete the task described in the Configuring Cisco SIP Proxy Server: Adding Cisco Unified MeetingPlace Express as a Subscriber.

Procedure

1. From the Cisco SPS main menu, click **Routes**.
2. Display existing routes by performing a search with the search tool.
3. To add a new route, do the following:
    1. Click **Add**.
    2. Enter route information.
    3. Click **Submit**.

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

- Read the following sections:
    - Prerequisites for Integrating in a SIP Environment
    - Restrictions for Integrating in a SIP Environment
- Configure your call-control device:
    - If you are integrating with Cisco Unified Communications Manager, complete the Configuring Cisco Unified Communications Manager: Adding the SIP Trunk and Route Pattern.
    - If you are integrating with Cisco Unified Communications Manager Express or a Cisco IOS software voice-enabled router, complete the tasks described in the Configuring Cisco Unified Communications Manager Express and Other Cisco IOS Software Voice-Enabled Routers: Adding the SIP Gateway and Dial Peer.
    - If you are integrating with Cisco SIP Proxy Server, complete the Configuring the Cisco SIP Proxy Server.

Procedure

1. Log in to Cisco Unified MeetingPlace Express and click **Administration**.
2. Click **System Configuration** > **Call Configuration** > **SIP Configuration**.
3. Configure the fields in Table: Required Configuration for SIP Configuration Page on Cisco Unified MeetingPlace Express for Integration Through a SIP Trunk.

Table: 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)
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.

4. Click **Save**.

5. Click **System Configuration** > **Call Configuration** > **Dial Configuration**.

6. Set the Outdials field to **SIP**.

7. Click **Save**.

8. 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.