APPENDIX A

Configuring an Example Environment

This appendix provides an example deployment scenario for setting up and configuring a Cisco Unified Application Environment.

The following section describes how to set up and configure an example environment having these properties:

- One Cisco Unified Application Server and one Cisco Unified Media Engine co-located on the same physical server
- SIP used for telephony integration
- One Cisco Unified Communications Manager cluster
- Test phones
- Sample applications used for integration

Note Actual IP addresses will differ in your own test environment.

The specific tasks required for setting up the Cisco Unified Application Environment will vary according to protocols and applications such as these:

- Number of Cisco Unified Application Servers and Cisco Unified Media Engines
- Number of Cisco Unified Communications Manager clusters
- Type of telephony protocol
- Types of applications used

Setting Up an Example Deployment and Performing Configuration Tasks

To set up your example deployment, you must perform these configuration tasks:

- Task 1: Log in to the Cisco Unified Application Environment Administration
- Task 2: Create a Cisco Unified Media Engine Connection
- Task 3: Create a SIP Connection to Cisco Unified Communications Manager
- Task 4: Create a SIP Trunk
- Task 5: Set Up a Route Pattern

- Task 6: Create Phones in Cisco Unified Communications Manager
- Task 7: Configure Your Phone to Connect to the Cisco Unified Communications Server
- Task 8: Configure the SIP Provider Plugin
- Task 9: Install, Configure, and Test Sample Applications

Task 1: Log in to the Cisco Unified Application Environment Administration

To log in to the Cisco Unified Application Environment Administration, follow these steps:

Procedure

Step 1 In the address bar of the web browser, enter the following URL: **http://**<*serverIPaddress*>**/cuaeadmin**.

Step 2 The Cisco Unified Application Environment Administration Login Screen appears.

Step 3 Enter your username and the password, and click **Login**.

Task 2: Create a Cisco Unified Media Engine Connection

The example applications in this guide use media capabilities. Therefore, you must identify at least one Cisco Unified Application Server that has Cisco Unified Media Engine software activated and licensed.

Note It is necessary to assign a Cisco Unified Media Engine to support media applications even if the Cisco Unified Application Server and Cisco Unified Media Engine are on the same hardware platform.

To assign a Cisco Unified Media Engine, follow these steps:

Procedure

Step 1 Choose **Connections > Add Connections** in the global navigation.

The Connection Wizard appears.

Step 2 Select **Media Engine** under Unified Application Environment Connections.

Step 3 Click **Next**.

The Adding Media Engine page appears.

Step 4 Enter the values as described in Table A-1.

Table A-1 Media Engine Fields

Field	Description/Recommendation
Media Engine Name	Name for Cisco Unified Media Engine.
IP Address	IP address of the server that hosts the Cisco Unified Media Engine.
Password/Verify Password	Password for access to the Cisco Unified Media Engine.

Step 5 Click **Save**.

Task 3: Create a SIP Connection to Cisco Unified Communications Manager

The Unified Communications Manager Cluster must contain at least one node corresponding to the IP address of a Cisco Unified Communications Manager server. By making this association, you dictate which Cisco Unified Communications Managers are signaled using SIP when an application makes a call.

To create a Cisco Unified Communications Manager Cluster, follow these steps:

Procedure

Step 1 Choose **Connections** > **Add Connection**.

The Connection Wizard appears.

Step 2 Select the **Cisco Unified Communication Manager Cluster** option under Unified Communication System Connections.

Step 3 Click **Next**.

The Add Unified Communication Manager Cluster page appears.

Step 4 Enter the values as described in Table A-2.

Table A-2 Unified Communications Manager Cluster Fields

Field	Description
Name	Name for the Cisco Unified Communications Manager.
Version	Version of the Cisco Unified Communications Manager you installed.

Table A-2 Unified Communications Manager Cluster Fields (continued)

Field	Description
Publisher Username	User name of the first Node (Publisher). The correct Publisher Username depends on the version of Cisco Unified Communications Manager you installed: • If version 3.x or 4.x, enter the username and password you use to log in to the Cisco Unified Communications Manager DeviceListX.asp report page. • If version 5.x, 6.x, 7x, enter the user name and password of a Cisco Unified Communications Application User with the Standard AXL API Access role. **Note** For more information about the Cisco Unified Communications Manager DeviceListX.asp report page or Cisco Unified Communications Manager Application Users, see the documentation provided with the version of Cisco Unified Communications Manager you have installed.
Publisher Password/ Verify Password	Password for the username entered in the Publisher Username field.
SNMP Community	SNMP community string you configured on the Cisco Unified Communications Manager Cluster (through Cisco Unified Serviceability). **Note** The SNMP Community string is required only if you are using Cisco Unified Communications Manager version 5.x, 6.x, 7x and you are using DeviceListX API calls. For more information about DeviceListX, see the Cisco Unified Application Environment API reference documentation.
Description	Description meaningful in your environment.

Step 5 Add nodes to replicate the structure of your Cisco Unified Communications cluster in the Unified Communications Manager Cluster Nodes section:

- Enter a name and IP address for each Cisco Unified Communications Manager node.
- Select the **Call Control** check box for all nodes that run the Cisco Unified Communications Manager service and that you want the Cisco Unified Application Environment to communicate with using SIP. Each node for which you check the Call Control check box is automatically placed into the Default SIP Call Route Group.

- Select the **CTI** check box for each Cisco Unified Communications Manager node that supports CTI services (for use with the JTAPI sample application.)

Note If CTI is not supported on any Cisco Unified Communications Manager nodes to which you have access, you cannot use JTAPI.

Step 6 Click **Save**.

Task 4: Create a SIP Trunk

The Cisco Unified Application Server appears as a device type in Cisco Unified Communications Manager. The device type it appears as is based on the protocol it uses to communicate with Cisco Unified Application Server. In the case of SIP, the Cisco Unified Application Server appears to the Cisco Unified Communications Manager as a SIP trunk connection.

The SIP trunk device name in Cisco Unified Communications Manager must correspond to the IP address or Domain Name System (DNS) name of the primary IP address of the Cisco Unified Application Server.

This section contains these subtasks:

- Create the SIP Trunk Security Profile, page A-5
- Create a SIP Profile, page A-6
- Create the SIP Trunk, page A-7

Create the SIP Trunk Security Profile

Creating a SIP Trunk Security Profile ensures that the Cisco Unified Communications Manager administrator can make changes when required without affecting anything other than connection between the Cisco Unified Communications Manager and the Cisco Unified Application Server.

To create the SIP Trunk Security Profile, which you will need later when you configure the SIP Trunk parameters, follow these steps:

Procedure

Step 1 Log in to the Cisco Unified Communications Manager administrative web interface.

Step 2 Choose **System > Security Profile > SIP Trunk Security Profile**.

Step 3 Click **Add New**.

The SIP Trunk Security Profile Configuration page appears.

Step 4 Enter the values for key fields as described in Table A-3.

Table A-3 SIP Trunk Security Profile Options

Field	Description
Name	Name that indicates that this SIP Trunk Security Profile is for use by the Cisco Unified Application Environment SIP Trunk.
Description	Description that indicates that this SIP Trunk Security Profile is for use by the Cisco Unified Application Environment SIP Trunk.
Device Security Mode	Non Secure.
Incoming Transport Type	TCP+UDP.
Outgoing Transport Type	TCP.
Enable Digest Authentication	Do not select check box.
X.509 Subject Name	Leave blank.
Incoming Port	5060.
Enable Application Level Authorization	Do not select check box.
Accept Presence Subscription	Do not select check box.
Accept Out-of-Dialog REFER	Do not select check box.
Accept Unsolicited Notification	Do not select check box.
Accept Replaces Header	Do not select check box.

Step 5 Click **Save.**

Create a SIP Profile

The SIP Profile is used specifically with the integration between Cisco Unified Communications Manager and the Unified Application Server.

To create the SIP Profile, which you will need later when you configure the SIP Trunk parameters, follow these steps:

Procedure

Step 1 Log in to the Cisco Unified Communications Manager administrative web interface.

Step 2 Choose **Device > Device Settings> SIP Profile**.

Step 3 Click **Add New**.

The SIP Profile Configuration page appears.

Step 4 Enter the values for key fields as described in Table A-4.

Table A-4 SIP Trunk Security Profile Options

Field	Description/Recommendation
Name	Name that indicates that this SIP Trunk Security Profile is for use by the Cisco Unified Application Environment SIP Trunk.
Description	Description that indicates that this SIP Trunk Security Profile is for use by the Cisco Unified Application Environment SIP Trunk.
Default MTP Telephony Event Payload Type	101.
Redirect By Application	Select this check box.
Disable Early Media on 180	Do not select check box.

Step 5 Click **Save.**

Create the SIP Trunk

The SIP trunk is used by the Cisco Unified Application Server to connect to the Cisco Unified Communications Manager.

To create a SIP trunk, follow these steps.

Procedure

Step 1 Log in to the Cisco Unified Communications Manager administrative web interface.

Step 2 Choose **Device** > **Trunk.**

Step 3 Click **Add New**.

Step 4 Select **SIP Trunk** as the trunk type.

The application sets SIP as the device protocol.

Step 5 Click **Next**.

The Trunk Configuration page appears.

Step 6 Enter the values for key fields as described in Table A-5.

Table A-5 SIP Trunk Fields

Field	Description/Recommendation
Device Information	
Device Name	Name that indicates this trunk is used for communicating with the Cisco Unified Application Environment.

Table A-5 SIP Trunk Fields

Field	Description/Recommendation (continued)
Device Pool	• If you are using a production environment, see your Cisco Unified Communications Manager administrator for assistance configuring devices. • If you are using the SDK version of the Cisco Unified Communications Manager, select **Default**.
Media Termination Point Required	• Do not select check box—If communication with the Cisco Unified Application Environment is required. • Select this check box—If your application uses SCCP phones and requires DTMF to be audible.
Retry Video Call as Audio	Select the check box.
Unattended Port	Do not select check box.
Call Routing Information - Inbound Calls	
Redirecting Diversion Header Delivery - Inbound	Select the check box.
Call Routing Information - Outbound Calls	
Redirecting Diversion Header Delivery – Outbound	Select the check box.
SIP Information	
Destination Address	The dual IP address given to the Cisco Unified Application Server.
SIP Trunk Security Profile	Security profile created in the "Create the SIP Trunk Security Profile" section on page A-5.
SIP Profile	SIP profile created in the "Create a SIP Profile" section on page A-6.

Step 7 Click **Save.**

Task 5: Set Up a Route Pattern

Creating a route pattern in Cisco Unified Communications Manager provides a route to the SIP trunk you defined in Task 4: Create a SIP Trunk, page A-5.

To set up a route pattern in Cisco Unified Communications Manager, follow these steps:

Procedure

Step 1 Log in to the Cisco Unified Communications Manager administrative web interface.

Step 2 Choose one of these:

- **Route Plan > Route Pattern** (3.3).
- **Route Plan > Route/Hunt > Route Pattern** (4.x)
- **Call Routing > Route/Hunt > Route Pattern** (5.x, 6.x, 7x)

Step 3 Click **Add a New Route Pattern**.

The Route Pattern Configuration page appears.

Step 4 Enter the values for key field as show in Table A-6.

Table A-6 Route Pattern Options

Field	Description/Recommendation
Route Pattern	Route pattern.
Description	Route pattern for the Cisco Unified Application Environment and/or a specific application installed on it.
Gateway/Route List	SIP Trunk created in the "Create the SIP Trunk" section on page A-7.

Step 5 Click **Save**.

Step 6 Choose **Accept** in the Authorization Codes warning.

Task 6: Create Phones in Cisco Unified Communications Manager

To define two test phones used in your network, follow these steps:

Note These instructions are specific to the IP Communicator. If you have a different phone you want to use for testing purposes, see the documentation for that phone.

Procedure

Step 1 Log in to the Cisco Unified Communications Manager administrative web interface.

Step 2 Choose **Device** > **Phone**.

Step 3 Click **Add New**.

Step 4 Select the phone type appropriate to the phone you have. For example, Cisco IP Communicator.

Step 5 Click **Next**.

Step 6 Select SIP or SCCP for the Device Protocol.

Step 7 Click **Next**.

The Phone Configuration page appears.

Step 8 Enter the value for key fields as shown in Table A-7.

Table A-7 Phone Configuration Options

Field	Description/Recommendation
Device Name	If your phone is a Cisco IP Communicator, enter the Device Name from the IP Communicator Network Preferences tab.
MAC Address (Hardware Phones only)	Media Access Control (MAC) address that identifies Cisco Unified IP phones (hardware phones only). Make sure that the value comprises 12 hexadecimal characters. For information on how to access the MAC address for other hardware phones, refer to the Cisco Unified IP Phone administration guide for the version of Cisco Unified Communications Manager that supports your phone model.

Step 9 Click **Save.**

The page refreshes and displays the Association Information section.

Step 10 Click the **Line [1] - Add a New DN** link.

The Directory Number Configuration page appears.

Step 11 Enter a directory number for the line.

Step 12 Make other configuration selections as necessary.

Step 13 Click **Save.**

Task 7: Configure Your Phone to Connect to the Cisco Unified Communications Server

Configure the phones used in your network to connect to the Cisco Unified Communications Server.

Note These instructions are specific to the IP Communicator. If you have a different phone you want to use for testing purposes, see the documentation for that phone.

To configure your phones, follow these steps:

Procedure

Step 1 Start Cisco IP Communicator.

Step 2 Select **Preferences** from the Menu drop-down list.

Step 3 Click the **Network** tab.

Step 4 Select **Use these TFTP servers**.

Step 5 Enter the Cisco Unified Communications Manager IP address in the **TFTP Server 1** field.

Step 6 Click **OK**.

Step 7 The phone registers to the Cisco Unified Communications Manager.

Note If you choose the wrong MAC address or the wrong phone type when creating the phone in Task 9: Install, Configure, and Test Sample Applications, page A-11, the message "Error DB Config" appears on the phone screen.

Task 8: Configure the SIP Provider Plugin

Tor apply configurations to a plugin, follow these steps:

Procedure

Step 1 Log in to the Cisco Unified Application Environment Administration.

Step 2 Choose **Plugins > List Plugins**. The List Plugins page appears.

Step 3 Select SIP Provider. The SIP Provider page appears.

Step 4 Enter the value for key fields as shown in Table A-8.

Table A-8 SIP Provider Plugin

Field	Description/Recommendation
DefaultOutboundFromNumber	Default From number for outbound call.
SIPTrunkIP	SIP Trunk IP address for outbound call (should match the IP used for SIP Trunk in Communications Manager).

Step 5 Click **Done**.

Task 9: Install, Configure, and Test Sample Applications

Install, configure, and test the following sample applications:

- MakeCall Sample Application, page A-12
- AnswerCall Sample Application, page A-14
- JTAPIConnect Sample Application, page A-17

Before You Begin

1. Download the sample applications to your system:
   a. On the Cisco Unified Application Server, navigate to the directory **C:\Program Files\Cisco Systems\Unified Application Environment\Tools\Apps**.
   b. Locate the sample application .mca files.
   c. Store the files.

      If you are logged in remotely to the Cisco Unified Application Environment Administration, store these files on your local machine.

      If you are logged in directly to the Cisco Unified Application Environment Administration, copy them to the desktop.
2. Install the MakeCall, AnswerCall, and JTAPIConnect applications. See Installing an Application, page 5-5.

MakeCall Sample Application

This section contains these topics:

Overview

The MakeCall sample application tests outbound dialing from the Cisco Unified Application Server to Cisco Unified Communications Manager as follows:

1. Uses a configured number to place an outbound call to a specified directory number (DN).
2. Plays ‘goodbye’ three times.
3. Hangs up on the called party.

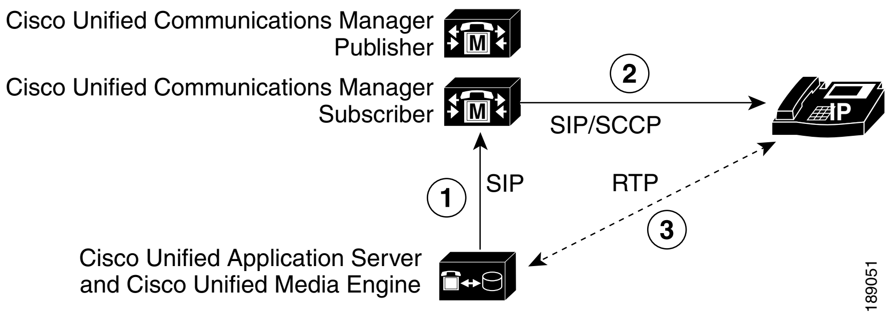
A successful outbound call indicates that the Cisco Unified Communications Manager cluster interprets the call as originating from the SIP trunk that represents the Cisco Unified Application Server.

Making a Call to an Internal IP Phone

Figure 1-1 shows the call flow in which the MakeCall application makes a call to an internal IP phone.

1. The Cisco Unified Application Server makes an SIP call to Cisco Unified Communications Manager.
2. Cisco Unified Communications Manager makes a call using SIP or SCCP to the IP phone as a result of the call from the Cisco Unified Application Server.
3. When the call is answered by the Cisco Unified Application Server, RTP streams are established between the IP phone and the Cisco Unified Media Engine.

Figure 1-1 MakeCall Application IP Phone Call Flow

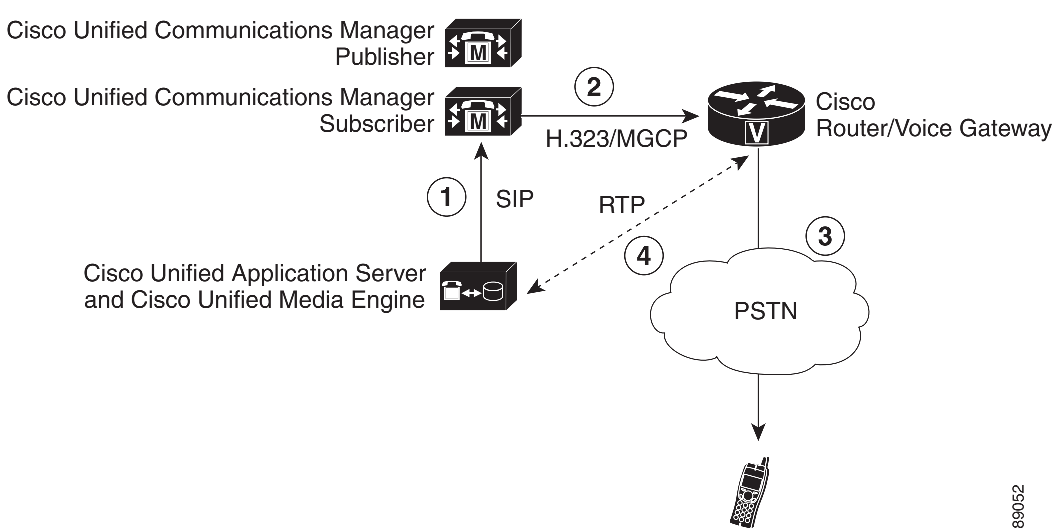


Making a Call to a PSTN Phone

Figure 1-2 shows the call flow in which the MakeCall application makes a call to a phone on the Public Switched Telephone Network (PSTN).

1. The Cisco Unified Application Server makes a SIP call to Cisco Unified Communications Manager.
2. Cisco Unified Communications Manager makes a call using H.323, MGCP, or SCCP to the gateway as a result of the call from the application server.
3. The Cisco Voice Gateway makes a call to the PSTN as a result of the call from Cisco Unified Communications Manager.
4. When the call is answered by the phone on the PSTN, RTP streams are established between the Cisco Voice Gateway and the Cisco Unified Media Engine.

Figure 1-2 MakeCall Application PSTN Phone Call Flow



Verifying the Trigger Parameter

The MakeCall application incorporates the Handle MakeCall script, which triggers, or initiates, when an HTTP request is received over port 8000 on the application server. Because multiple HTTP-triggered scripts can be installed on the application server, you must verify that the Handle MakeCall script uses a unique trigger parameter.

To verify the trigger parameter for the Handle MakeCall script, follow these steps:

Procedure

Step 1 Log in to the Cisco Unified Application Environment Administration.

Step 2 Choose **Applications > List Triggers**. The List Triggers page appears.

Step 3 Click **MakeCall** to open the MakeCall page.

Step 4 Verify that the URL trigger parameter value is /MakeCall. This means that the Handle MakeCall script will initiate when an HTTP request comes in with the URL **http://<**_Application Server IP_**>:8000/MakeCall**.

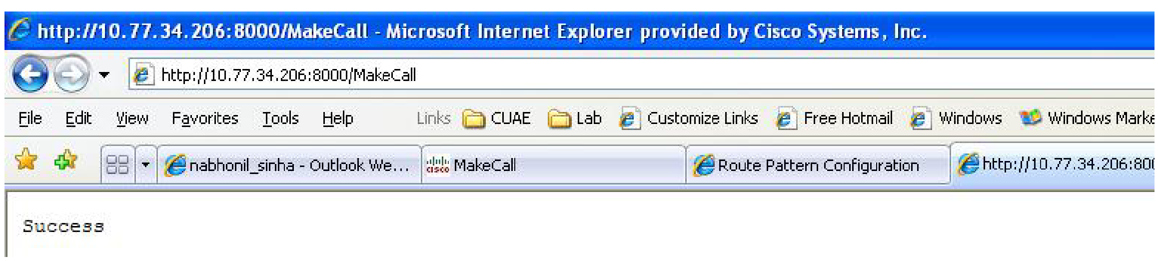
Step 5 Click **Done**.

Testing the MakeCall Application

After installing the MakeCall application and verifying the trigger setting, you can test the application by opening a web browser and entering **http://<**_Application Server IP_**>:8000/MakeCall**.

If the outbound call succeeds, a message is displayed, as shown in Figure 1-3, and you hear ‘goodbye’ three times. This indicates you have successfully integrated outbound calling using SIP and the Cisco Unified Application Environment.

Figure 1-3 Testing the MakeCall Application



Note If the test does not work, check the server logs for any errors. See Viewing Server Logs, page 8-2.

AnswerCall Sample Application

This section contains these topics:

Overview

The AnswerCall sample application tests inbound calling to the Cisco Unified Application Server as follows:

1. Answers a call routed to the application server.
2. Plays 'goodbye' three times.
3. Hangs up on the caller.

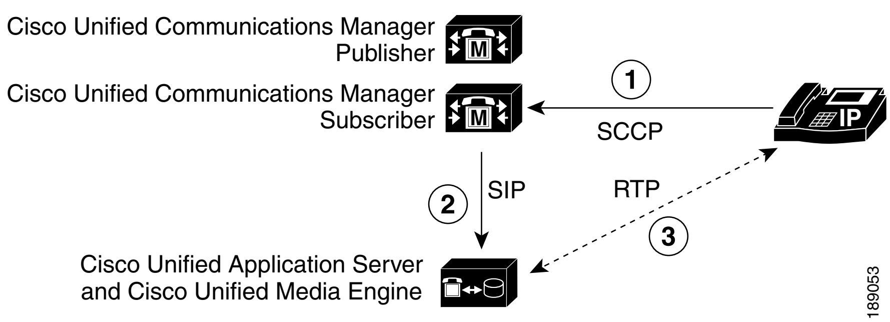
A successful call indicates that the Cisco Unified Application Server is able to receive incoming calls.

Answering a Call from an Internal IP Phone

Figure 1-4 shows the call flow in which the AnswerCall application answers a call from an internal IP phone.

1. A call is made from an IP phone to Cisco Unified Communications Manager.
2. The Cisco Unified Communications Manager makes an SIP call as a result of the call from the IP phone.
3. When the call is answered by the Cisco Unified Application Server, RTP streams are established between the IP phone and the Cisco Unified Media Engine.

Figure 1-4 AnswerCall Application IP Phone Call Flow

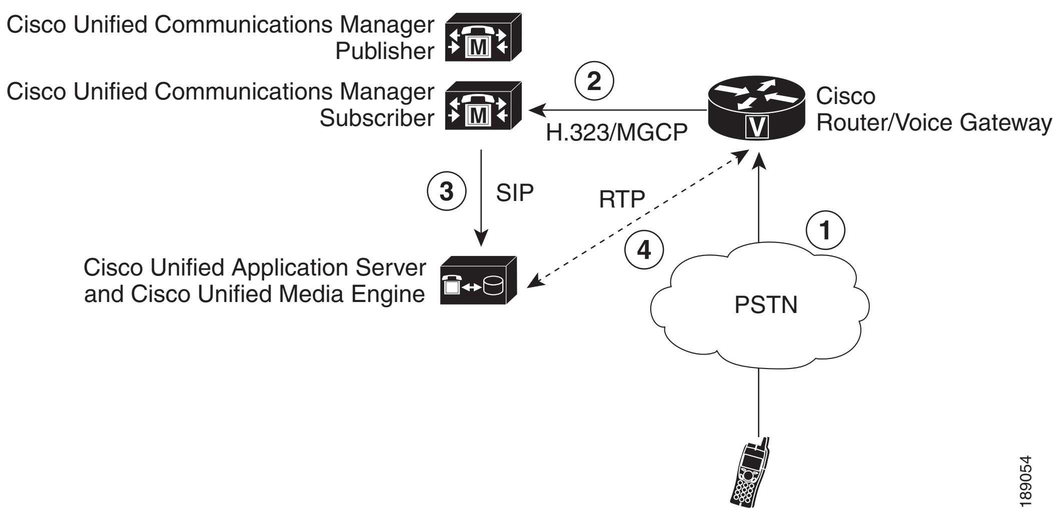


Answering a Call from the PSTN

Figure 1-5 shows the call flow in which the AnswerCall application answers a call from the PSTN.

1. A phone on the PSTN makes a call to an H.323 or MGCP gateway.
2. The Cisco Voice Gateway makes a call to Cisco Unified Communications Manager as a result of the call from the PSTN phone.
3. The Cisco Unified Communications Manager makes a SIP call as a result of the call from the Cisco Voice Gateway.
4. When the call is answered by the application server, RTP streams are established between the Cisco Voice Gateway and the Cisco Unified Media Engine.

Figure 1-5 AnswerCall Application PSTN Call Flow



Defining the Trigger Parameter

The Handle Inbound Call script, which handles calls routed to the application server, does not contain pre-defined trigger parameters. However, because it is a dial-in application (you dial a number to test it), you should define a trigger parameter for the script.

For consistency with the route pattern 5000X, which was defined in Task 9: Install, Configure, and Test Sample Applications, page A-11, define a trigger parameter with the name **"to"** and value "**50000**."

To define the trigger parameter for the Handle Inbound Call script, follow these steps:

Procedure

Step 1 Log in to the Cisco Unified Application Environment Administration.

Step 2 Choose **Applications > List Triggers**. The List Triggers page appears.

Step 3 Click **AnswerCall** to open the AnswerCall page.

Step 4 Enter **To** for the parameter name.

Step 5 Enter **50000** for the value.

Step 6 Click **Add Parameter**.

Step 7 Click **Done**.

Testing the AnswerCall Application

To test AnswerCall application, call 50000 from an IP phone that is configured to dial to the previously-defined route pattern (Figure 1-6). The call should be answered immediately, play goodbye three times, then hang up.

Figure 1-6 Testing the AnswerCall Application

Note If the test does not work, check the server logs for any errors. See Viewing Server Logs, page 8-2.

JTAPIConnect Sample Application

This section contains these topics:

Overview

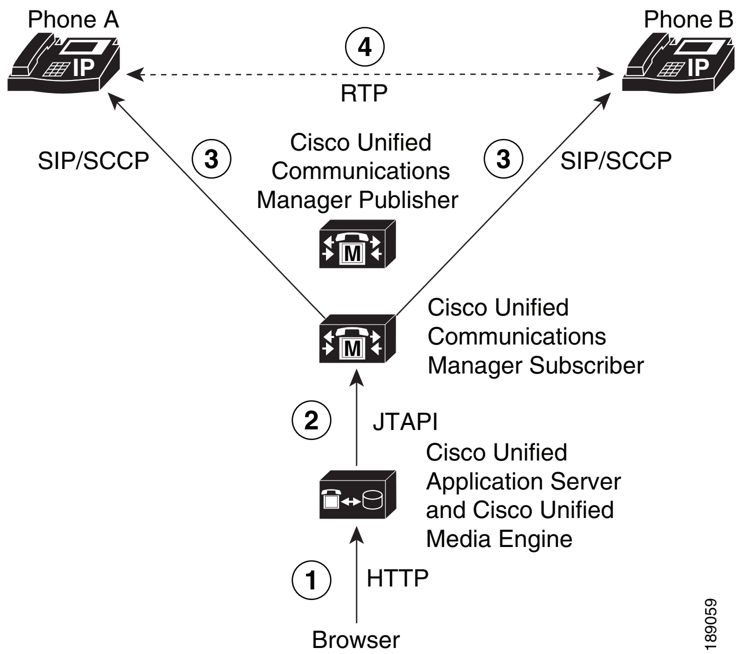
The JTAPIConnect sample application uses JTAPI APIs and triggers to establish a call between two phones as follows:

1. The application is initiated by an HTTP request.
2. Phone A calls phone B.
3. Phone B answers, then hangs up.

Making Calls to Internal IP Phones

Figure 1-7 shows the call flow in which a call is initiated, and phone A calls phone B.

Figure 1-7 JTAPIConnect IP Phone Call Flow



1. An HTTP request invokes the application on the Cisco Unified Application Server.
2. The Cisco Unified Application Server sends JTAPI requests to the CTI Manager on the Cisco Unified Communications Manager.
3. Phone A makes a SIP and/ or SCCP call to Phone B.
4. Phone B answers the call. This establishes an RTP stream between the two phones.

Configuring a Monitored CTI Device Pool

To create a monitored CTI device pool follow these steps:

Procedure

Step 1 Choose **Connections > List Device Pools**.

The List Device Pools page appears.

Step 2 Click **Add**.

The Choose Pool Type page appears.

Step 3 Select **Monitored CTI Device Pool**, then click **Go.**

The Creating Monitored CTI Device Pool page appears.

Step 4 Select the cluster you created in Task 3: Create a SIP Connection to Cisco Unified Communications Manager, page A-3 from the Cluster drop-down list, then click **Go**.

Step 5 Enter the values as described in Table A-9.

Table A-9 Creating a Monitored CTI Device Pool

Field	Description
Name	Pool name.
Primary CTI Manager	First CTI Manager service that the Cisco Unified Application Server will try to connect to. The drop down of available options is auto-populated from any CUCM node defined that has the CTI role checked. (See Task 3: Create a SIP Connection to Cisco Unified Communications Manager, page A-3.)
Secondary CTI Manager	Second CTI Manager service that the Cisco Unified Application Server will try to connect to if the primary is busy or inaccessible. The drop down of available options is auto-populated from any CUCM node defined that has the CTI role checked. (See Task 3: Create a SIP Connection to Cisco Unified Communications Manager, page A-3.)
Username	User name to allow monitoring of all devices configured in the device pool. (This is the user name defined in the Cisco Unified Communications Manager with the this permission: Standard CTI Allow Control of All Devices.)
Password/Verify Password	Password to allow monitoring of all the devices in the device pool. (This is the associated password defined in the Cisco Unified Communications Manager.)

Step 6 Click **Save**.

Step 7 Click the **Devices** tab.

Step 8 Click **Edit**. A new page appears.

Step 9 Under Add One Device, enter the name of one of the test phones, then click **Submit**.

Step 10 Add the second phone as described in Step 9.

Configuring the JTAPI Application

To verify the configuration parameter for the JTAPIConnect application, follow these steps:

Procedure

Step 1 Log in to the Cisco Unified Application Environment Administration

Step 2 Choose **Applications > List Applications** The List Applications page appears.

Step 3 Click **JTAPIConnec**t.

Step 4 Under Extended Configuration:

- In the Device1 field enter the device name of first device from the monitored device pool.
- In the Device1_Line field enter the line number of first device from the monitored device pool.

- In the Device2 field enter the device name of second device from the monitored device pool.
- In the Device2_Line field enter the line number of the second device from monitored device pool.

Step 5 Click **Apply**.

Step 6 Click **Done**.

Verifying the Trigger Parameter

The JTAPIConnect application triggers, or initiates when an HTTP request is received over port 8000 on the Cisco Unified Application Server. Because multiple HTTP-triggered scripts can be installed on the application server, you must verify that the application uses a unique trigger parameter. The default setting is JTAPIConnect.

To verify the trigger parameter for the JTAPIConnect application, follow these steps:

Procedure

Step 1 Log in to the Cisco Unified Application Environment Administration.

Step 2 Choose **Applications > List Triggers**. The List Triggers page appears.

Step 3 Click **JTAPIConnect** to open the JTAPIConnect page.

Step 4 Verify that the URL trigger parameter value is /JTAPIConnect. This means that the JTAPIConnect application will initiate when an HTTP request comes in with the URL **http://<***Application Server IP***>:8000/JTAPIConnect**.

Step 5 Click **Done**.

Testing the JTAPIConnect Application

To test the application by opening a web browser and entering **http://<***Application Server IP***>:8000/JTAPIConnect**.

Both devices (test phones) ring, then auto answer each other. After two seconds, the phones hang up.

Note If the test does not work, check the server logs for any errors. See Viewing Server Logs, page 8-2.