



CCS-UC-1

Crestron Mercury with Cisco[®] Unified Communications Manager 11.0

Configuration Guide

Crestron Electronics, Inc.

The product warranty can be found at www.crestron.com/warranty.

The specific patents that cover Crestron products are listed at patents.crestron.com.

Certain Crestron products contain open source software. For specific information, please visit www.crestron.com/opensource.

Crestron, the Crestron logo, AirMedia, Crestron Toolbox, and Mercury are either trademarks or registered trademarks of Crestron Electronics, Inc. in the United States and/or other countries. Cisco is either a trademark or registered trademark of Cisco Systems, Inc. in the United States and/or other countries. Other trademarks, registered trademarks, and trade names may be used in this document to refer to either the entities claiming the marks and names or their products. Crestron disclaims any proprietary interest in the marks and names of others. Crestron is not responsible for errors in typography or photography.

This document was written by the Technical Publications department at Crestron.
©2017 Crestron Electronics, Inc.

Contents

| | |
|--------------------------------------------------------------------|-----------|
| Introduction | 3 |
| Audience | 3 |
| Topology | 3 |
| Software Requirements | 4 |
| Hardware Requirements | 4 |
| Product Description | 4 |
| Summary | 4 |
| Features Supported | 4 |
| Features Not Supported..... | 5 |
| Mercury Configuration | 5 |
| Setup | 5 |
| Discovering/Accessing the Device..... | 5 |
| Configuring the Device | 6 |
| Network Settings | 8 |
| Configure the SIP Parameters | 8 |
| Cisco UCM Configuration | 10 |
| Configure the User | 10 |
| Configure an SIP Profile | 13 |
| Configure Phone Security Profile | 17 |
| Configure the Crestron Device as a Third-Party SIP Device..... | 18 |
| Configure Media Resource Group and Media Resource Group List | 20 |
| Configure the Duplex Streaming Parameter | 21 |
| Configure Trunks..... | 22 |
| Cisco UCM - PSTN Gateway Trunk Configuration | 22 |
| Cisco UCM - Unity Connection Trunk Configuration..... | 26 |
| Configure Route Patterns | 30 |
| Voicemail Configuration..... | 34 |
| Configure Voicemail Pilot and Voicemail Profile on Cisco UCM | 34 |
| Configuration on Unity Connection: Add New Phone System | 36 |
| Configure a Voicemail User | 40 |

CCS-UC-1: SIP Endpoint with Cisco Unified Communications Manager 11.0

Introduction

This configuration guide describes the necessary procedure to configure a Crestron® Mercury™ device to register to the Cisco® Unified Communications Manager (UCM) as a basic SIP endpoint.

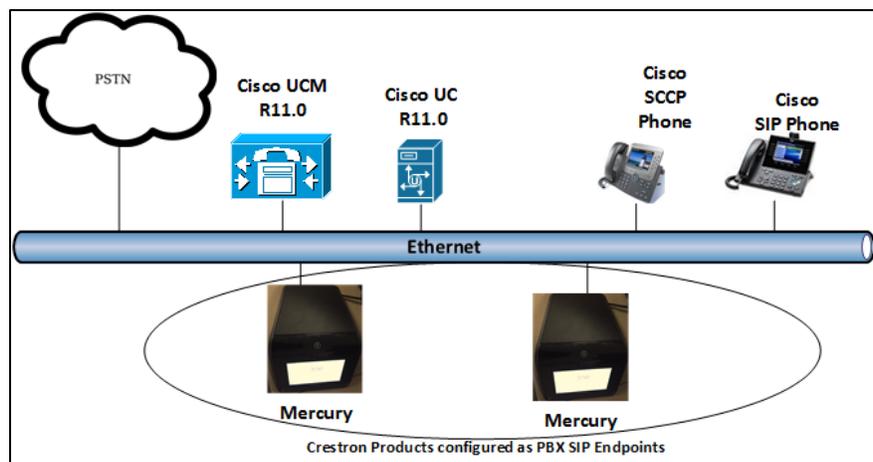
Audience

This document is intended for users attempting to configure and use Crestron Mercury devices as SIP endpoints registering to the Cisco UCM.

Topology

The network topology for the Crestron Mercury endpoint to interop with the Cisco UCM is shown below.

SIP Endpoint Integration with Cisco UCM: Reference Network



The lab network consists of the following components:

- Cisco UCM cluster for voice features
- Cisco SCCP and SIP phones
- Cisco Unity Connection as the voicemail system
- Crestron Mercury as the SIP endpoints

Software Requirements

- Cisco UCM v 11.0.1.20000-2
- Cisco Unity Connection v 11.0.1.20000-2
- Mercury devices v 1.3211.00020

Hardware Requirements

- Cisco UCS-C240-M3S VMWare Host running ESXi 5.5
- Cisco 3845 as PSTN Gateway
- Cisco Phones: models 7960 (SCCP), 8961 (SIP), 8945 (SIP)
- Crestron Mercury devices (2)

Product Description

The Mercury device is a complete solution for conference rooms. It acts as an all-in-one touch screen, speakerphone, and AirMedia® product for conference rooms that integrate microphones and speakers into the user interface at the table.

Crestron Toolbox™ is used to discover and control all Crestron devices on the network.

The Crestron Mercury web interface is used to control the Crestron Mercury devices on the network.

Summary

The Mercury devices were configured on the Cisco UCM as basic SIP endpoints since they support only a single line/extension. The devices were successfully registered to the Cisco UCM with digest authentication.

The sections below describe supported and unsupported features on a Mercury device.

Features Supported

- Registration with digest authentication
- Basic calls with G722, G711u, and G711a codecs
- Caller ID (limited to only calling number)
- DTMF support

- Early media support
- Retrieval of a parked call
- Transferee in a call transfer
- Conference participant
- Member of hunt group
- Member of shared line configuration
- Voicemail access and interaction

Features Not Supported

- Caller ID presentation with name and number display
- Call hold and resume
- Call forwarding on the device (Forwarding can be configured on the PBX for the DN assigned to the endpoint.)
- Call waiting
- Conference
- Attended call transfer
- Early attended call transfer
- Blind call transfer
- Configuration of shared line on device
- Initiating call park
- Message waiting indicator

Mercury Configuration

Setup

The Mercury device requires only one connection from its LAN port. The LAN port needs to be connected to one POE+ port to power it up and to be connected to the network for reachability to the Cisco UCM.

Discovering/Accessing the Device

Crestron Toolbox discovers and accesses Mercury devices on the network.

The Help menu on Crestron Toolbox assists the user through the discovery and configuration procedure.

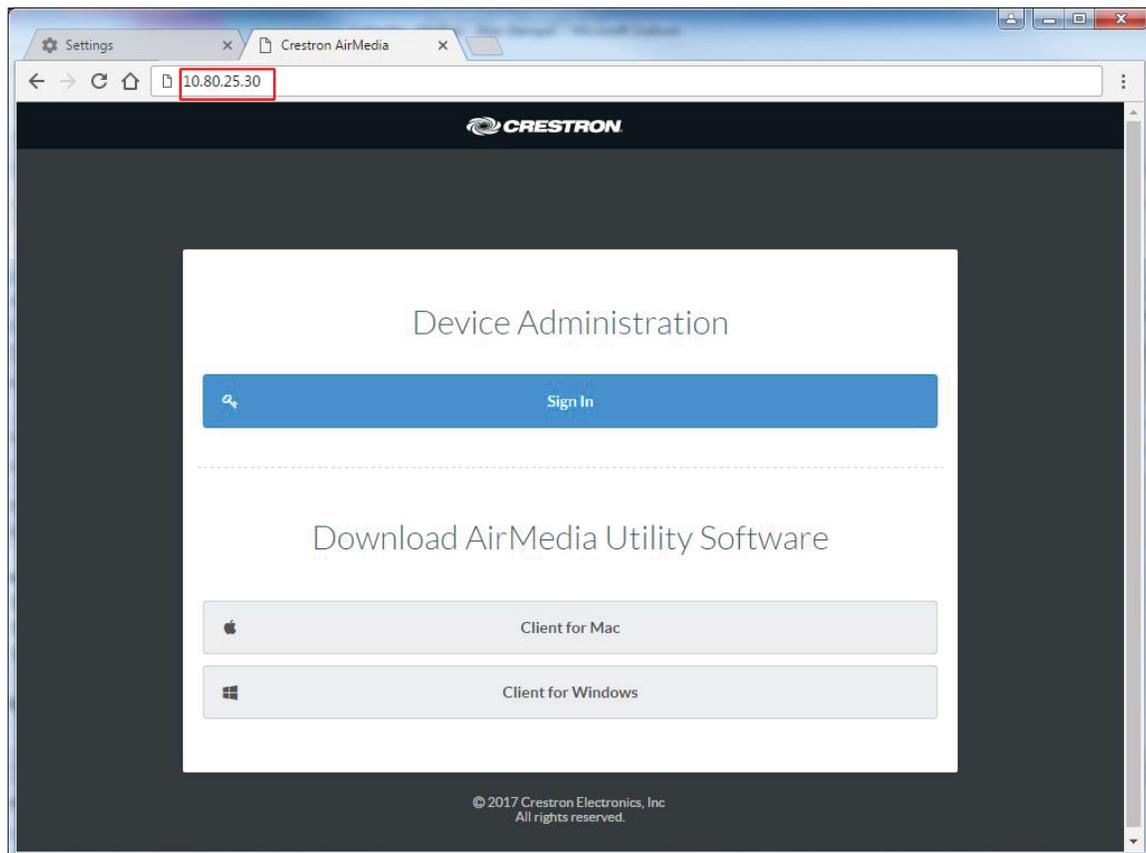
This document will therefore not include details of the same.

Apart from this tool, the device itself provides the IP address that can be used to access and configure the device via the web. (On the device home screen, navigate to **Present a Source > AirMedia**. This specifies the address of the device.)

Configuring the Device

1. Access the web GUI for the device by using an http session with the device's IP address. 10.80.25.30 was used in this example as the device IP. The initial page that displays is shown below.

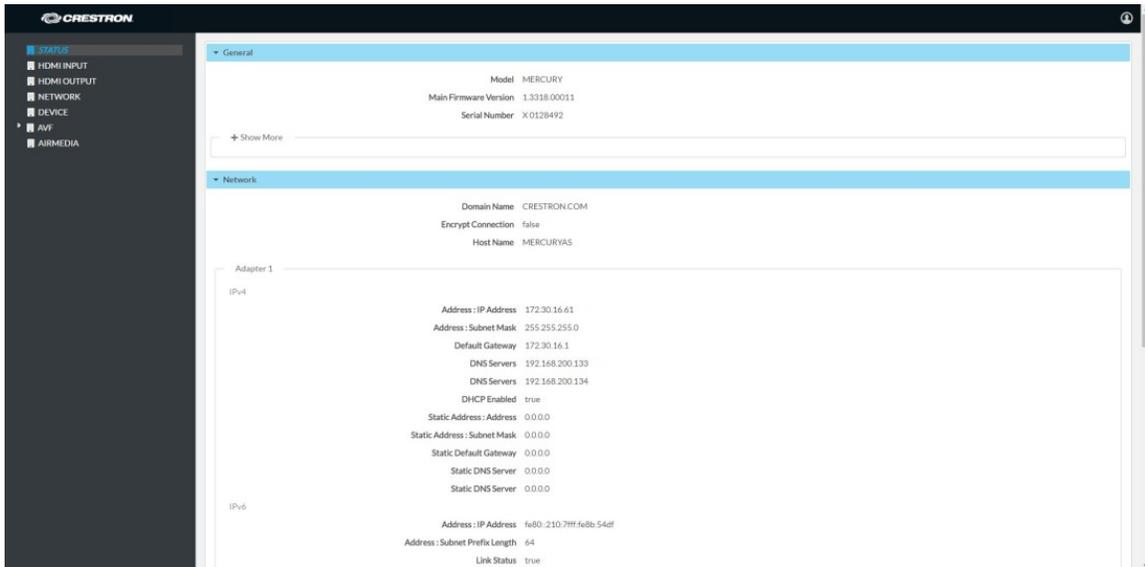
Crestron Mercury Login to Web GUI



2. Click **Device Administration**. For information on device administration, refer to Doc. 7844 at www.crestron.com/manuals.

The **Status** screen that appears displays basic information on the device as shown below.

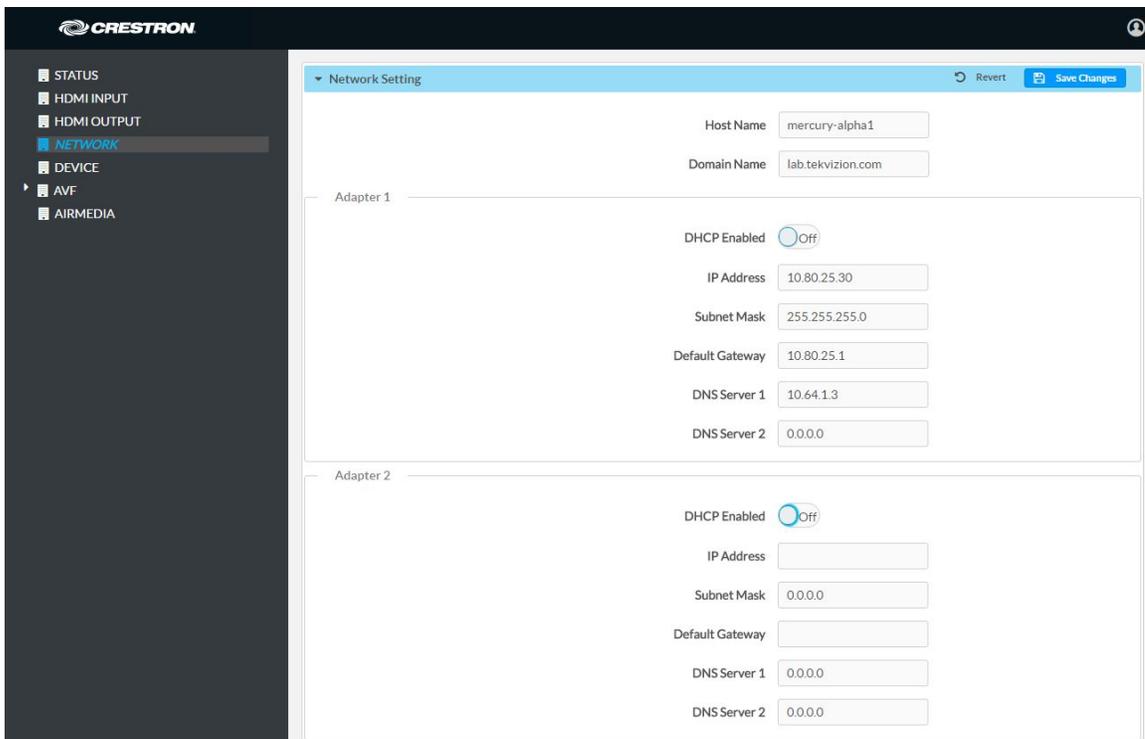
Crestron Mercury: Status Screen



The device can be configured from the **Network Setting** screen.

3. On the web GUI, navigate to **Network**. The **Network Setting** screen is displayed.

Crestron Mercury: Network: Network Setting



Network Settings

Configure the parameters below. Click **Save Changes** when done.

- **Domain Name:** lab.tekvizion.com, used in this example (mostly auto-detected by device when in DHCP mode).
- **DHCP:** Either of the two can be chosen:
 - **Obtain an IP address automatically**
 - **Use the following IP address**

For this example, a static IP was configured.

- **IP address:** 10.80.25.30, used in this example.
- **Subnet Mask:** 255.255.255.0, used in this example.
- **Default Gateway:** 10.80.25.1, used in this example.
- **DNS Servers:** 10.64.1.3, used in this example.

Configure the SIP Parameters

1. On the web GUI, navigate to **Device > SIP Calling**. The **SIP Calling** screen is displayed.

Crestron Mercury: Device: SIP Parameters

The screenshot displays the Crestron Mercury web interface for configuring SIP parameters. The left sidebar shows a navigation menu with options: STATUS, HDMI INPUT, HDMI OUTPUT, NETWORK, DEVICE (highlighted), AVF, and AIRMEDIA. The main content area is titled 'SIP Calling' and includes a 'Revert' button and a 'Save Changes' button. The configuration fields are as follows:

- Enable SIP:** On (checked)
- Transport Type:** UDP
- Server IP Address:** 10.80.25.2
- Port:** 5060
- Server Username:** Mercury_2602
- Server Password:** ****
- Server Realm:** *
- Local Extension:** 2602
- Proxy Server:** NONE
- SIP Server Status:** Online
- Assigned Ethernet Port:** Adapter 1 (selected), Adapter 2
- Enable Server Validation:** Disabled
- Select Trusted Certificate Authorities:** Starfield Services Root Certificat

2. Enable the check box for **Enable SIP**.
3. Configure the **SIP Server IP Address**: Enter the IP address of the Cisco UCM node. *10.80.25.2* was used in this example.
4. Configure the **SIP port**: *5060* was used in this example.
5. Configure the **SIP Server Username**: Enter the end user configured on Cisco UCM for this device. *2102* was used in this example.
6. Configure the **SIP Server Password**: Enter the password as configured on Cisco UCM for this end user.
7. Configure the **SIP Local Extension**: Enter the directory number that was configured for this device on Cisco UCM. *2102* was used in this example.
8. Leave all other fields at their default values.
9. Click **Save Changes**.

Once the device successfully registers with the Cisco UCM, the **SIP Server Status** updates its status to show *Online*.

Cisco UCM Configuration

This section describes the Cisco UCM configuration necessary to integrate the Crestron device as an SIP endpoint.

NOTE: It is assumed that the general installation and basic Cisco UCM configuration has already been administered.

Configure the User

1. Navigate to **User Management > End User**.
2. Click **Add New**. The End User configuration window appears.

Cisco UCM: End User Configuration

 **Cisco Unified CM Administration**
For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

End User Configuration

 Save  Delete  Add New

User Information

User Status: Enabled Local User

User ID*

Password

Confirm Password

Self-Service User ID

PIN

Confirm PIN

Last name*

Middle name

First name

Display name

Title

Directory URI

Telephone Number

Home Number

Mobile Number

Pager Number

Mail ID

Manager User ID

Department

User Locale

Associated PC

Digest Credentials

Confirm Digest Credentials

User Profile

3. Configure **User ID**: Enter a unique end user identification name. Two users were configured for this example for the Mercury devices: *Mercury_2600* and *Mercury_2602*.
4. Configure **Password**: Enter any password. This same password will be entered on the device against SIP Server Password. 123456 was used in this example.
5. Confirm **Password**: Re-enter the same password configured above.
6. Configure the **Last Name**: Enter the end user last name.
7. Configure the **Digest Credentials**: Enter a string of alphanumeric characters.
8. Confirm the **Digest Credentials**: Re-enter the password configured above.

9. Click **Save**. All of the configured users are listed as shown below.

Cisco UCM: End Users Configured for All Mercury Devices

The screenshot displays the Cisco Unified CM Administration web interface. At the top, the navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration For Cisco Unified Communications Solutions", the user role "administrator", and links for "Search Documentation", "About", and "Logout". A secondary navigation bar contains menu items: "System", "Call Routing", "Media Resources", "Advanced Features", "Device", "Application", "User Management", "Bulk Administration", and "Help".

The main content area is titled "Find and List Users". It features a toolbar with "Add New", "Select All", "Clear All", and "Delete Selected" buttons. Below this is a "Status" section indicating "2 records found".

The user list is titled "User (1 - 2 of 2)" and includes a "Rows per Page" dropdown set to 50. A search filter is applied: "Find User where User ID begins with Mer". The search results are as follows:

| <input type="checkbox"/> | User ID | Meeting Number | First Name | Last Name | Department | Directory URI | User Status |
|--------------------------|--------------|----------------|------------|-------------|------------|---------------|--------------------|
| <input type="checkbox"/> | Mercury_2600 | | | Mercury2600 | | | Enabled Local User |
| <input type="checkbox"/> | Mercury_2602 | | | Mercury2602 | | | Enabled Local User |

At the bottom of the list, there are buttons for "Add New", "Select All", "Clear All", and "Delete Selected".

Configure an SIP Profile

For the example, a new SIP Profile **Standard SIP Profile_Test** was configured.

To add a new SIP Profile, perform the following procedure.

1. Navigate to **Device > Device Settings > SIP Profile**.

Cisco UCM: SIP Profile Configuration (1/4)

The screenshot displays the Cisco Unified CM Administration web interface. The top navigation bar includes the Cisco logo and the text "Cisco Unified CM Administration For Cisco Unified Communications Solutions". Below this is a breadcrumb trail: System > Call Routing > Media Resources > Advanced Features > Device > Application > User Management > Bulk Administration. The main heading is "SIP Profile Configuration". A toolbar contains icons for Save, Delete, Copy, Reset, Apply Config, and Add New. The "SIP Profile Information" section is expanded, showing the following configuration details:

| | |
|--------------------------------------------------------------------------|----------------------------------------------------|
| Name* | Standard SIP Profile_Test |
| Description | Default SIP Profile |
| Default MTP Telephony Event Payload Type* | 101 |
| Early Offer for G.Clear Calls* | Disabled |
| User-Agent and Server header information* | Send Unified CM Version Information as User-Agent |
| Version in User Agent and Server Header* | Major And Minor |
| Dial String Interpretation* | Phone number consists of characters 0-9, *, #, and |
| Confidential Access Level Headers* | Disabled |
| <input type="checkbox"/> Redirect by Application | |
| <input type="checkbox"/> Disable Early Media on 180 | |
| <input type="checkbox"/> Outgoing T.38 INVITE include audio mline | |
| <input type="checkbox"/> Use Fully Qualified Domain Name in SIP Requests | |
| <input type="checkbox"/> Assured Services SIP conformance | |

The "SDP Information" section is also expanded, showing the following configuration details:

| | |
|----------------------------------------------------------------------------------|---------------------------------|
| SDP Session-level Bandwidth Modifier for Early Offer and Re-invites* | TIAS and AS |
| SDP Transparency Profile | Pass all unknown SDP attributes |
| Accept Audio Codec Preferences in Received Offer* | Default |
| <input type="checkbox"/> Require SDP Inactive Exchange for Mid-Call Media Change | |
| <input type="checkbox"/> Allow RR/RS bandwidth modifier (RFC 3556) | |

Cisco UCM: SIP Profile Configuration (2/4)

| Parameters used in Phone | |
|----------------------------------------------|------------------------------------------------------------------------------------------------------------------------------------------|
| Timer Invite Expires (seconds)* | <input type="text" value="180"/> |
| Timer Register Delta (seconds)* | <input type="text" value="5"/> |
| Timer Register Expires (seconds)* | <input type="text" value="3600"/> |
| Timer T1 (msec)* | <input type="text" value="500"/> |
| Timer T2 (msec)* | <input type="text" value="4000"/> |
| Retry INVITE* | <input type="text" value="6"/> |
| Retry Non-INVITE* | <input type="text" value="10"/> |
| Media Port Ranges | <input checked="" type="radio"/> Common Port Range for Audio and Video <input type="radio"/> Separate Port Ranges for Audio and Video |
| Start Media Port* | <input type="text" value="16384"/> |
| Stop Media Port* | <input type="text" value="32766"/> |
| DSCP for Audio Calls | <input type="text" value="Use System Default"/> ▼ |
| DSCP for Video Calls | <input type="text" value="Use System Default"/> ▼ |
| DSCP for Audio Portion of Video Calls | <input type="text" value="Use System Default"/> ▼ |
| DSCP for TelePresence Calls | <input type="text" value="Use System Default"/> ▼ |
| DSCP for Audio Portion of TelePresence Calls | <input type="text" value="Use System Default"/> ▼ |
| Call Pickup URI* | <input type="text" value="x-cisco-serviceuri-pickup"/> |
| Call Pickup Group Other URI* | <input type="text" value="x-cisco-serviceuri-opickup"/> |
| Call Pickup Group URI* | <input type="text" value="x-cisco-serviceuri-gpickup"/> |
| Meet Me Service URI* | <input type="text" value="x-cisco-serviceuri-meetme"/> |
| User Info* | <input type="text" value="None"/> ▼ |
| DTMF DB Level* | <input type="text" value="Nominal"/> ▼ |

Cisco UCM: SIP Profile Configuration (3/4)

| Call Hold Ring Back* | Off | | | | | | |
|--------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|-----------------------------|-----------------|----------------|-----------------|---|--|--|
| Anonymous Call Block* | Off | | | | | | |
| Caller ID Blocking* | Off | | | | | | |
| Do Not Disturb Control* | User | | | | | | |
| Telnet Level for 7940 and 7960* | Disabled | | | | | | |
| Resource Priority Namespace | < None > | | | | | | |
| Timer Keep Alive Expires (seconds)* | 120 | | | | | | |
| Timer Subscribe Expires (seconds)* | 120 | | | | | | |
| Timer Subscribe Delta (seconds)* | 5 | | | | | | |
| Maximum Redirections* | 70 | | | | | | |
| Off Hook To First Digit Timer (milliseconds)* | 15000 | | | | | | |
| Call Forward URI* | x-cisco-serviceuri-cfwdall | | | | | | |
| Speed Dial (Abbreviated Dial) URI* | x-cisco-serviceuri-abbrdial | | | | | | |
| <input checked="" type="checkbox"/> Conference Join Enabled <input type="checkbox"/> RFC 2543 Hold <input checked="" type="checkbox"/> Semi Attended Transfer <input type="checkbox"/> Enable VAD <input type="checkbox"/> Stutter Message Waiting <input type="checkbox"/> MLPP User Authorization | | | | | | | |
| Normalization Script | | | | | | | |
| Normalization Script | < None > | | | | | | |
| <input type="checkbox"/> Enable Trace | | | | | | | |
| <table border="1"> <thead> <tr> <th></th> <th>Parameter Name</th> <th>Parameter Value</th> </tr> </thead> <tbody> <tr> <td>1</td> <td></td> <td></td> </tr> </tbody> </table> | | | Parameter Name | Parameter Value | 1 | | |
| | Parameter Name | Parameter Value | | | | | |
| 1 | | | | | | | |

Cisco UCM: SIP Profile Configuration (4/4)

| | |
|-------------------------------------------------------------------------------------------------------------------------------------|----------------------------------------------------|
| Incoming Requests FROM URI Settings | |
| Caller ID DN | <input type="text"/> |
| Caller Name | <input type="text"/> |
| Trunk Specific Configuration | |
| Reroute Incoming Request to new Trunk based on* | Never <input type="text"/> |
| Resource Priority Namespace List | < None > <input type="text"/> |
| SIP Rel1XX Options* | Disabled <input type="text"/> |
| Video Call Traffic Class* | Mixed <input type="text"/> |
| Calling Line Identification Presentation* | Default <input type="text"/> |
| Session Refresh Method* | Invite <input type="text"/> |
| Early Offer support for voice and video calls* | Best Effort (no MTP inserted) <input type="text"/> |
| <input type="checkbox"/> Enable ANAT | |
| <input type="checkbox"/> Deliver Conference Bridge Identifier | |
| <input type="checkbox"/> Allow Passthrough of Configured Line Device Caller Information | |
| <input type="checkbox"/> Reject Anonymous Incoming Calls | |
| <input type="checkbox"/> Reject Anonymous Outgoing Calls | |
| <input type="checkbox"/> Send ILS Learned Destination Route String | |
| SIP OPTIONS Ping | |
| <input checked="" type="checkbox"/> Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)" | |
| Ping Interval for In-service and Partially In-service Trunks (seconds)* | 60 <input type="text"/> |
| Ping Interval for Out-of-service Trunks (seconds)* | 120 <input type="text"/> |
| Ping Retry Timer (milliseconds)* | 500 <input type="text"/> |
| Ping Retry Count* | 6 <input type="text"/> |
| SDP Information | |
| <input type="checkbox"/> Send send-receive SDP in mid-call INVITE | |
| <input type="checkbox"/> Allow Presentation Sharing using BFCP | |
| <input type="checkbox"/> Allow iX Application Media | |
| <input type="checkbox"/> Allow multiple codecs in answer SDP | |

2. On the screen that appears, click **Add New** and configure the SIP Profile as below.
 - a. Assign a Name: **Standard SIP Profile_Test**, used in the example.
 - b. Configure **Early offer support for voice and video calls *** as Best Effort (no MTP inserted).
 - c. Retain all other default config.
3. Then click **Save** and then **Apply Config**.

Configure Phone Security Profile

1. Navigate to **System > Security > Phone Security Profile**.

Cisco UCM: Phone Security Profile

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Phone Security Profile Configuration

Save Delete Copy Reset Apply Config Add New

Status
Status: Ready

Phone Security Profile Information
Product Type: Third-party SIP Device (Basic)
Device Protocol: SIP
Name*: Crestron
Description: Phone security Profile for Crestron Devices
Nonce Validity Time*: 600
Transport Type*: TCP+UDP
 Enable Digest Authentication

Parameters used in Phone
SIP Phone Port*: 5060

Save Delete Copy Reset Apply Config Add New

2. Click **Add New**.
3. Configure a **Name**: *Crestron*, used in this example.
4. Configure **Transport Type**: TCP+UDP.
5. Check the **Enable Digest Authentication** check box.
6. Click **Save**.

Configure the Crestron Device as a Third-Party SIP Device

1. Navigate to Device > Phone.
2. Click Add New.

Cisco UCM: Add Crestron Device as Third-Party SIP Device (1/2)

Phone Configuration

 Save

Phone Type

Product Type: Third-party SIP Device (Basic)
Device Protocol: SIP

Device Information

 Device is not trusted

| | |
|------------------------------------------|---------------------------------------------------------------------------------------------|
| MAC Address* | 00107F0522CC |
| Description | SEP00107F0522CC |
| Device Pool* | Default View Details |
| Common Device Configuration | < None > View Details |
| Phone Button Template* | Third-party SIP Device (Basic) |
| Common Phone Profile* | Standard Common Phone Profile View Details |
| Calling Search Space | < None > |
| AAR Calling Search Space | < None > |
| Media Resource Group List | < None > |
| Location* | Hub_None |
| AAR Group | < None > |
| Device Mobility Mode* | Default |
| Owner | <input checked="" type="radio"/> User <input type="radio"/> Anonymous (Public/Shared Space) |
| Owner User ID* | Mercury_2600 |
| Use Trusted Relay Point* | Default |
| Always Use Prime Line* | Default |
| Always Use Prime Line for Voice Message* | Default |
| Geolocation | < None > |

Ignore Presentation Indicators (internal calls only)
 Logged Into Hunt Group
 Remote Device

Cisco UCM: Add Crestron Device as Third-Party SIP Device (2/2)

| | |
|----------------------------------------------------------------------------------------------------------------------------|---------------------------|
| Number Presentation Transformation | |
| Caller ID For Calls From This Phone | |
| Calling Party Transformation CSS | < None > |
| <input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS (Caller ID For Calls From This Phone) | |
| Remote Number | |
| Calling Party Transformation CSS | < None > |
| <input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS (Device Mobility Related Information) | |
| Protocol Specific Information | |
| BLF Presence Group* | Standard Presence group |
| MTP Preferred Originating Codec* | 711ulaw |
| Device Security Profile* | Crestron |
| Rerouting Calling Search Space | < None > |
| SUBSCRIBE Calling Search Space | < None > |
| SIP Profile* | Standard SIP Profile_Test |
| Digest User | Mercury_2600 |
| <input type="checkbox"/> Media Termination Point Required | |
| <input type="checkbox"/> Unattended Port | |
| <input type="checkbox"/> Require DTMF Reception | |
| View Details | |
| MLPP and Confidential Access Level Information | |
| MLPP Domain | < None > |
| Confidential Access Mode | < None > |
| Confidential Access Level | < None > |
| <input type="button" value="Save"/> | |

3. Select **Phone Type** as **Third-party SIP Device (Basic)**.
4. Click **Next**.
5. Configure **MAC Address**: Enter the MAC Address of the Mercury device.
6. Select **Device Pool** as **Default**.
7. Select **Phone Button Template** as **Third-party SIP Device (Basic)**.
8. Select **Owner User ID**: select the End User configured earlier from the drop-down. In this example, *Mercury_2600* was selected for the first Mercury device and *Mercury_2602* for the second Mercury device.
9. Select **Device Security Profile** as configured earlier from the drop-down. *Crestron* was used in this example.

10. Select **SIP Profile** as configured earlier from the drop-down menu. *Standard SIP Profile_Test* was used in this example.
11. Select **Digest User ID**: select the End User configured earlier from the drop-down. In this example, *Mercury_2600* was selected for the first Mercury device and *Mercury_2602* for the second Mercury device.
12. Click **Save**.
13. Add a **DN** to this phone. *2600* was configured for one of the Mercury devices in this example. *DN 2602* was added to the other Mercury device.

Configure Media Resource Group and Media Resource Group List

A media resource group is required to include Music on Hold servers Conference Bridges and Media Termination Points that may be required to test the Cisco UCM or Service Provider features.

To configure the Media Resource Group (MRG), perform the following procedure.

1. Select **Media Resources > Media Resource Group**.
2. Click **Add New**.

Cisco UCM: Media Resource Group Configuration

The screenshot displays the Cisco Unified CM Administration interface for configuring a Media Resource Group (MRG). The page title is "Media Resource Group Configuration". The breadcrumb trail shows the path: System > Call Routing > Media Resources > Advanced Features > Device > Application > User Management > Bulk Administration > Help. The page includes a navigation bar with "Navigation" and "Go" buttons, and a user profile section for "administrator".

The configuration page features several sections:

- Status:** Status: Ready
- Media Resource Group Status:** Media Resource Group: MRG (used by 23 devices)
- Media Resource Group Information:**
 - Name*: MRG
 - Description: (empty)
- Devices for this Group:**
 - Available Media Resources**: ANN_3, CFB_3, IVR_2, IVR_3, MOH_3
 - Selected Media Resources*: ANN_2 (ANN), CFB_2 (CFB), MOH_2 (MOH), MTP_2 (MTP)
- Use Multi-cast for MOH Audio (If at least one multi-cast MOH resource is available)

3. Provide a **Name** and select Media Resources from the **Available Media Resources**.

NOTE: These are assumed to have been added earlier and are available for use /registered with this Cisco UCM.)

Perform the following procedure to configure the Media Resource Group List (MRGL).

1. Select **Media Resources > Media Resource Group List**.

Cisco UCM: Media Resource Group List Configuration

The screenshot shows the Cisco Unified CM Administration interface for the Media Resource Group List Configuration. The page title is "Media Resource Group List Configuration" and it includes a "Related Links" section with "Back To Find/List" and "Go". The interface has a top navigation bar with "System", "Call Routing", "Media Resources", "Advanced Features", "Device", "Application", "User Management", "Bulk Administration", and "Help". Below the navigation bar, there are buttons for "Save", "Delete", "Copy", and "Add New". The main configuration area is divided into several sections: "Status" (Ready), "Media Resource Group List Status" (MRGL used by 23 devices), "Media Resource Group List Information" (Name: MRGL), and "Media Resource Groups for this List" (Available Media Resource Groups and Selected Media Resource Groups: MRG). At the bottom, there are buttons for "Save", "Delete", "Copy", and "Add New".

2. Click **Add New**.
3. Provide a **Name** and select the media resource groups from the **Available Media Resource Groups**.

Configure the Duplex Streaming Parameter

1. Navigate to **System > Service Parameters**.
2. Select **Server**: Cisco UCM publisher from the drop-down menu.
3. Select **Service**: Cisco Call Manager (Active).
4. Configure **Duplex Streaming Enabled** to **True**.
This parameter is configured to **True** to enable the device to hear MoH when it is put on hold. When set to false, the device user hears silence when the call is put on hold.

Configure Trunks

Two trunks were configured for this validation example:

- Between the Cisco UCM and the PSTN Gateway for calls to the PSTN
- Between the Cisco UCM and Cisco Unity Connection for voicemail

Cisco UCM - PSTN Gateway Trunk Configuration

To create a new trunk, perform the following procedure.

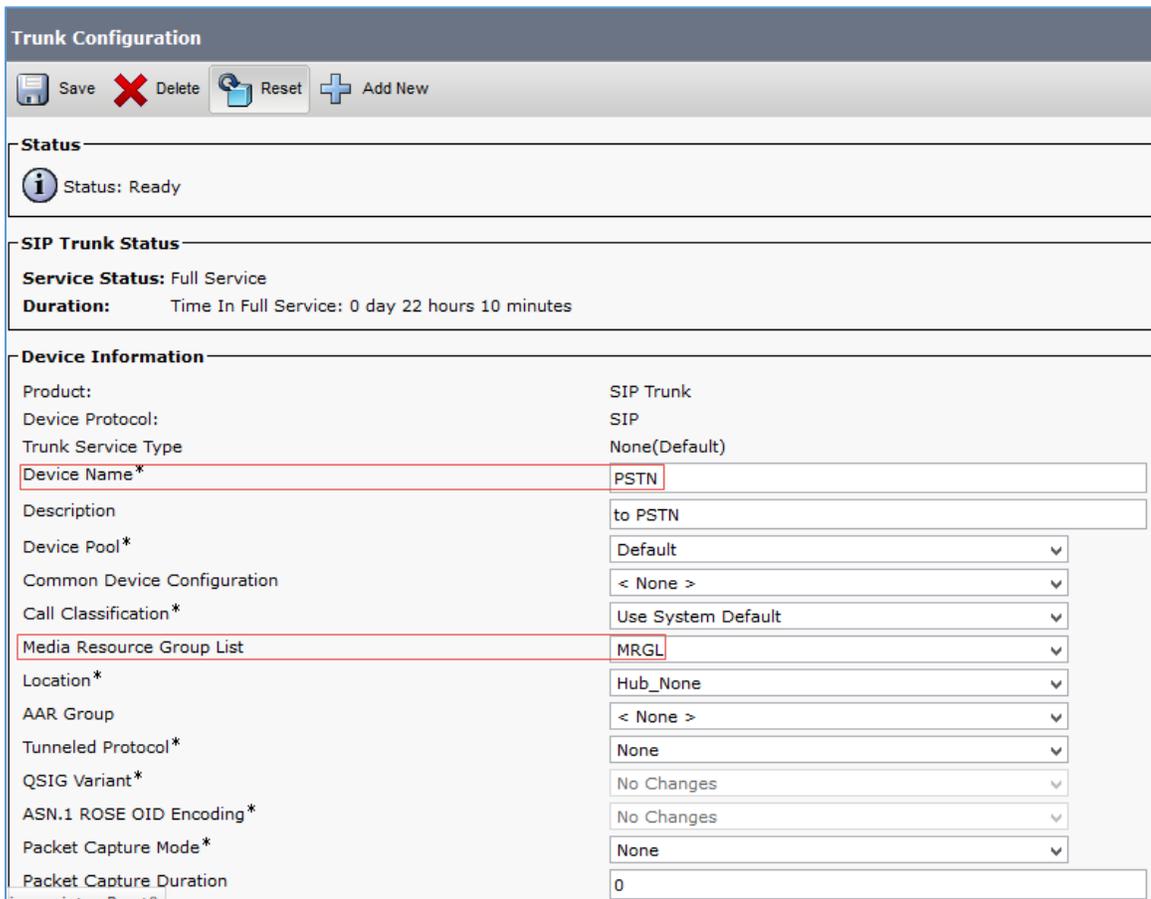
1. From the **Device** menu drop-down list, select **Trunk**.
2. Click **Add New**.

Cisco UCM: Add New Trunk

The screenshot shows the Cisco Unified CM Administration interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration", and the user role "administrator". Below the navigation bar is a menu with options: System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The main content area is titled "Trunk Configuration" and includes a "Next" button with a green arrow. Below this is a "Status" section showing "Status: Ready". The "Trunk Information" section contains three dropdown menus: "Trunk Type*" set to "SIP Trunk", "Device Protocol*" set to "SIP", and "Trunk Service Type*" set to "None(Default)". A "Next" button is located at the bottom left of the form.

3. Select Trunk Type as **SIP Trunk**, Device Protocol as **SIP**, and Trunk Service Type as **None (Default)**.
4. Click **Next**.

Cisco UCM: Configure Cisco UCM-PSTN Trunk Parameters (1/5)



Trunk Configuration

Save Delete Reset Add New

Status
Status: Ready

SIP Trunk Status
Service Status: Full Service
Duration: Time In Full Service: 0 day 22 hours 10 minutes

Device Information

| | |
|-----------------------------|--------------------|
| Product: | SIP Trunk |
| Device Protocol: | SIP |
| Trunk Service Type | None(Default) |
| Device Name* | PSTN |
| Description | to PSTN |
| Device Pool* | Default |
| Common Device Configuration | < None > |
| Call Classification* | Use System Default |
| Media Resource Group List | MRGL |
| Location* | Hub_None |
| AAR Group | < None > |
| Tunneled Protocol* | None |
| QSIG Variant* | No Changes |
| ASN.1 ROSE OID Encoding* | No Changes |
| Packet Capture Mode* | None |
| Packet Capture Duration | 0 |

5. In the **Device Name** field, enter a unique SIP Trunk name, and, as an option, provide a description. *PSTN* was used in this example.
6. From the **Device Pool** drop-down list, select a device pool. *Default* was used in this example.
7. From the Media Resource Group List, select **MRGL** from the drop-down menu.
8. Ensure that the **Media Termination Point Required** is unchecked.

Cisco UCM: Configure Cisco UCM-PSTN Trunk Parameters (2/5)

| | |
|-----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|------------------------------|
| <input type="checkbox"/> Media Termination Point Required | |
| <input checked="" type="checkbox"/> Retry Video Call as Audio | |
| <input type="checkbox"/> Path Replacement Support | |
| <input type="checkbox"/> Transmit UTF-8 for Calling Party Name | |
| <input type="checkbox"/> Transmit UTF-8 Names in QSIG APDU | |
| <input type="checkbox"/> Unattended Port | |
| <input type="checkbox"/> SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information. | |
| Consider Traffic on This Trunk Secure* | When using both sRTP and TLS |
| Route Class Signaling Enabled* | Default |
| Use Trusted Relay Point* | Default |
| <input checked="" type="checkbox"/> PSTN Access | |
| <input checked="" type="checkbox"/> Run On All Active Unified CM Nodes | |
| Intercompany Media Engine (IME) | |
| E.164 Transformation Profile | < None > |
| MLPP and Confidential Access Level Information | |
| MLPP Domain | < None > |
| Confidential Access Mode | < None > |
| Confidential Access Level | < None > |

9. Select the **Redirecting Diversion Header Delivery – Inbound** check box.

Cisco UCM: Configure Cisco UCM-PSTN Trunk Parameters (3/5)

| Call Routing Information | | | | |
|-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|----------|-------------------------|----------------------|-------------------------------------|
| <input checked="" type="checkbox"/> Remote-Party-Id | | | | |
| <input checked="" type="checkbox"/> Asserted-Identity | | | | |
| Asserted-Type* | Default | | | |
| SIP Privacy* | Default | | | |
| Inbound Calls | | | | |
| Significant Digits* | All | | | |
| Connected Line ID Presentation* | Default | | | |
| Connected Name Presentation* | Default | | | |
| Calling Search Space | < None > | | | |
| AAR Calling Search Space | < None > | | | |
| Prefix DN | | | | |
| <input checked="" type="checkbox"/> Redirecting Diversion Header Delivery - Inbound | | | | |
| Incoming Calling Party Settings | | | | |
| If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned. | | | | |
| Clear Prefix Settings | | Default Prefix Settings | | |
| Number Type | Prefix | Strip Digits | Calling Search Space | Use Device Pool CSS |
| Incoming Number | Default | 0 | < None > | <input checked="" type="checkbox"/> |

10. Select the **Redirecting Diversion Header Delivery – Outbound** check box.

Cisco UCM: Configure Cisco UCM-PSTN Trunk Parameters (4/5)

| | |
|------------------------------------------------------------------------------------------|------------------------------------|
| Connected Party Settings | |
| Connected Party Transformation CSS | < None > |
| <input checked="" type="checkbox"/> Use Device Pool Connected Party Transformation CSS | |
| Outbound Calls | |
| Called Party Transformation CSS | < None > |
| <input checked="" type="checkbox"/> Use Device Pool Called Party Transformation CSS | |
| Calling Party Transformation CSS | < None > |
| <input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS | |
| Calling Party Selection* | Originator |
| Calling Line ID Presentation* | Default |
| Calling Name Presentation* | Default |
| Calling and Connected Party Info Format* | Deliver DN only in connected party |
| <input checked="" type="checkbox"/> Redirecting Diversion Header Delivery - Outbound | |
| Redirecting Party Transformation CSS | < None > |
| <input checked="" type="checkbox"/> Use Device Pool Redirecting Party Transformation CSS | |
| Caller Information | |
| Caller ID DN | |
| Caller Name | |

11. Configure the SIP Information as described in the following procedure.

Cisco UCM: Configure Cisco UCM-PSTN Trunk Parameters (5/5)

| SIP Information | | | | | | | | | |
|--------------------------------------------------------|------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|--------------------------|---------------------|--------------------------|------------------|----|------------|--|------|
| Destination | | | | | | | | | |
| <input type="checkbox"/> Destination Address is an SRV | | | | | | | | | |
| | <table border="1"> <thead> <tr> <th></th> <th>Destination Address</th> <th>Destination Address IPv6</th> <th>Destination Port</th> </tr> </thead> <tbody> <tr> <td>1*</td> <td>10.64.1.72</td> <td></td> <td>5060</td> </tr> </tbody> </table> | | Destination Address | Destination Address IPv6 | Destination Port | 1* | 10.64.1.72 | | 5060 |
| | Destination Address | Destination Address IPv6 | Destination Port | | | | | | |
| 1* | 10.64.1.72 | | 5060 | | | | | | |
| MTP Preferred Originating Codec* | 711ulaw | | | | | | | | |
| BLF Presence Group* | Standard Presence group | | | | | | | | |
| SIP Trunk Security Profile* | Non Secure SIP Trunk Profile_Crestron | | | | | | | | |
| Rerouting Calling Search Space | < None > | | | | | | | | |
| Out-Of-Dialog Refer Calling Search Space | < None > | | | | | | | | |
| SUBSCRIBE Calling Search Space | < None > | | | | | | | | |
| SIP Profile* | Standard SIP Profile_Test View Details | | | | | | | | |
| DTMF Signaling Method* | No Preference | | | | | | | | |
| Normalization Script | | | | | | | | | |
| Normalization Script | < None > | | | | | | | | |
| <input type="checkbox"/> Enable Trace | | | | | | | | | |
| | <table border="1"> <thead> <tr> <th></th> <th>Parameter Name</th> <th>Parameter Value</th> </tr> </thead> <tbody> <tr> <td>1</td> <td></td> <td></td> </tr> </tbody> </table> | | Parameter Name | Parameter Value | 1 | | | | |
| | Parameter Name | Parameter Value | | | | | | | |
| 1 | | | | | | | | | |

- a. Enter the **Destination Address** and port of the PSTN Gateway.
- b. Select the **Non Secure SIP Trunk Profile_Crestron** as the **SIP Trunk Security Profile**.
- c. Select the configured **Standard SIP Profile_Test** SIP Profile.

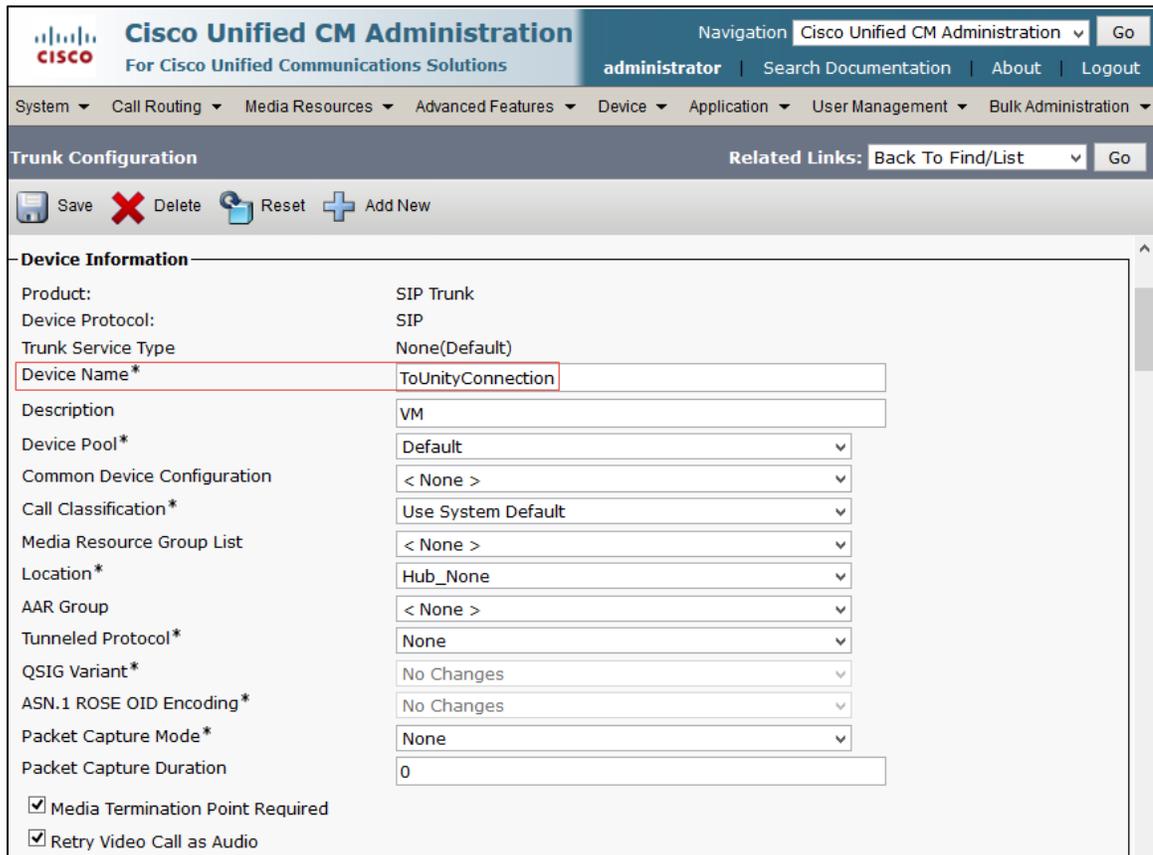
12. Click **Save**.

Cisco UCM - Unity Connection Trunk Configuration

Similar to the above trunk configuration, configure a new trunk from Cisco UCM to the Unity Connection Server.

Below are screenshots of the trunk parameters.

Cisco UCM: Trunk to Voicemail System - Unity Connection (1/6)



The screenshot displays the Cisco Unified CM Administration web interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration", and the user role "administrator". Below the navigation bar, a breadcrumb trail shows the path: System > Call Routing > Media Resources > Advanced Features > Device > Application > User Management > Bulk Administration > Trunk Configuration. The "Trunk Configuration" page has a "Related Links" section with a "Back To Find/List" button. Below this, there are icons for Save, Delete, Reset, and Add New. The main content area is titled "Device Information" and contains the following configuration fields:

| | |
|-----------------------------|--------------------|
| Product: | SIP Trunk |
| Device Protocol: | SIP |
| Trunk Service Type | None(Default) |
| Device Name* | ToUnityConnection |
| Description | VM |
| Device Pool* | Default |
| Common Device Configuration | < None > |
| Call Classification* | Use System Default |
| Media Resource Group List | < None > |
| Location* | Hub_None |
| AAR Group | < None > |
| Tunneled Protocol* | None |
| QSIG Variant* | No Changes |
| ASN.1 ROSE OID Encoding* | No Changes |
| Packet Capture Mode* | None |
| Packet Capture Duration | 0 |

At the bottom of the form, there are two checked checkboxes: "Media Termination Point Required" and "Retry Video Call as Audio".

Cisco UCM: Trunk to Voicemail System - Unity Connection (2/6)

| | |
|-----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|------------------------------|
| <input type="checkbox"/> Path Replacement Support | |
| <input type="checkbox"/> Transmit UTF-8 for Calling Party Name | |
| <input type="checkbox"/> Transmit UTF-8 Names in QSIG APDU | |
| <input type="checkbox"/> Unattended Port | |
| <input type="checkbox"/> SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information. | |
| Consider Traffic on This Trunk Secure* | When using both sRTP and TLS |
| Route Class Signaling Enabled* | Default |
| Use Trusted Relay Point* | Default |
| <input type="checkbox"/> PSTN Access | |
| <input checked="" type="checkbox"/> Run On All Active Unified CM Nodes | |

Intercompany Media Engine (IME)

E.164 Transformation Profile < None >

MLPP and Confidential Access Level Information

MLPP Domain < None >

Confidential Access Mode < None >

Confidential Access Level < None >

Cisco UCM: Trunk to Voicemail System - Unity Connection (3/6)

Call Routing Information

Remote-Party-Id

Asserted-Identity

Asserted-Type* Default

SIP Privacy* Default

Inbound Calls

Significant Digits* All

Connected Line ID Presentation* Default

Connected Name Presentation* Default

Calling Search Space < None >

AAR Calling Search Space < None >

Prefix DN

Redirecting Diversion Header Delivery - Inbound

Incoming Calling Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

| Number Type | Prefix | Strip Digits | Calling Search Space | Use Device Pool CSS |
|-----------------|---------|--------------|----------------------|-------------------------------------|
| Incoming Number | Default | 0 | < None > | <input checked="" type="checkbox"/> |

Cisco UCM: Trunk to Voicemail System - Unity Connection (4/6)

Incoming Called Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

| Number Type | Prefix | Strip Digits | Calling Search Space | Use Device Pool CSS |
|-----------------|---------|--------------|----------------------|-------------------------------------|
| Incoming Number | Default | 0 | < None > | <input checked="" type="checkbox"/> |

Connected Party Settings

Connected Party Transformation CSS

Use Device Pool Connected Party Transformation CSS

Outbound Calls

Called Party Transformation CSS

Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS

Use Device Pool Calling Party Transformation CSS

Calling Party Selection*

Calling Line ID Presentation*

Calling Name Presentation*

Calling and Connected Party Info Format*

Redirecting Diversion Header Delivery - Outbound

Cisco UCM: Trunk to Voicemail System - Unity Connection (5/6)

Use Device Pool Redirecting Party Transformation CSS

Caller Information

Caller ID DN

Caller Name

Maintain Original Caller ID DN and Caller Name in Identity Headers

SIP Information

Destination

Destination Address is an SRV

| | Destination Address | Destination Address IPv6 | Destination Port |
|-----|---------------------|--------------------------|------------------|
| 1 * | 10.80.25.5 | | 5060 |

MTP Preferred Originating Codec*

BLF Presence Group*

SIP Trunk Security Profile*

Rerouting Calling Search Space

Out-Of-Dialog Refer Calling Search Space

SUBSCRIBE Calling Search Space

SIP Profile* [View Details](#)

DTMF Signaling Method*

Cisco UCM: Trunk to Voicemail System - Unity Connection (6/6)

Normalization Script

Normalization Script

Enable Trace

| | Parameter Name | Parameter Value | |
|---|----------------------|----------------------|-------------------------------------------------------------------|
| 1 | <input type="text"/> | <input type="text"/> | <input type="button" value="+"/> <input type="button" value="-"/> |

Recording Information

None

This trunk connects to a recording-enabled gateway

This trunk connects to other clusters with recording-enabled gateways

Geolocation Configuration

Geolocation

Geolocation Filter

Send Geolocation Information

Configure Route Patterns

Route patterns were configured for the following:

- To route calls from the Cisco UCM to the PSTN
- To restrict Caller ID on outgoing calls
- To access the voicemail

To configure route patterns, perform the following procedure.

1. Navigate to **Call Routing > Route/Hunt > Route Pattern**.
2. Click **Add New**.
3. Enter the details desired and then Click **Save**.

The route pattern 9.@ was configured to enable outbound dialing from Cisco UCM to PSTN using the access code as “9”. The screenshot below shows the configuration.

Cisco UCM: Route Pattern: Outbound Dialing Using Access Code 9 (1/2)

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Route Pattern Configuration

Save
 Delete
 Copy
 Add New

Status

Status: Ready

Pattern Definition

| | | |
|------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|---------------------------------------------------------------------------------------------------------------------------------------------------------------------|------------------------|
| Route Pattern* | 9.@ | |
| Route Partition | < None > | |
| Description | | |
| Numbering Plan* | NANP | |
| Route Filter | < None > | |
| MLPP Precedence* | Default | |
| <input type="checkbox"/> Apply Call Blocking Percentage | | |
| Resource Priority Namespace Network Domain | < None > | |
| Route Class* | Default | |
| Gateway/Route List* | PSTN | (Edit) |
| Route Option | <input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern No Error | |
| Call Classification* | OffNet | |
| External Call Control Profile | < None > | |
| <input type="checkbox"/> Allow Device Override <input checked="" type="checkbox"/> Provide Outside Dial Tone <input type="checkbox"/> Allow Overlap Sending <input type="checkbox"/> Urgent Priority | | |
| <input type="checkbox"/> Require Forced Authorization Code | | |
| Authorization Level* | 0 | |
| <input type="checkbox"/> Require Client Matter Code | | |

Calling Party Transformations

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation* Default

Calling Name Presentation* Default

Calling Party Number Type* Cisco CallManager

Calling Party Numbering Plan* Cisco CallManager

Cisco UCM: Route Pattern: Outbound Dialing Using Access Code 9 (2/2)

| | | |
|------------------------------------------------------------------------------------------------------------------------------------------------------|------------------------|-------------------------|
| Connected Party Transformations | | |
| Connected Line ID Presentation * | Default | |
| Connected Name Presentation * | Default | |
| Called Party Transformations | | |
| Discard Digits | PreDot | |
| Called Party Transform Mask | | |
| Prefix Digits (Outgoing Calls) | | |
| Called Party Number Type * | Cisco CallManager | |
| Called Party Numbering Plan * | Cisco CallManager | |
| ISDN Network-Specific Facilities Information Element | | |
| Network Service Protocol | -- Not Selected -- | |
| Carrier Identification Code | | |
| Network Service | Service Parameter Name | Service Parameter Value |
| -- Not Selected -- | < Not Exist > | |
| <input type="button" value="Save"/> <input type="button" value="Delete"/> <input type="button" value="Copy"/> <input type="button" value="Add New"/> | | |

The route pattern 67.@ was configured to restrict Caller ID on outbound calls. The screenshots below show the configuration.

Cisco UCM: Route Pattern: Restrict Caller ID (1/2)

| | | |
|------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|----------------------------------------------------------------------------------------------------------|---------------------------------------------------------------------|
| Cisco Unified CM Administration For Cisco Unified Communications Solutions | | Navigation Cisco Unified administrator Search Documents |
| System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾ | | |
| Route Pattern Configuration | | Related Links: |
| <input type="button" value="Save"/> <input type="button" value="Delete"/> <input type="button" value="Copy"/> <input type="button" value="Add New"/> | | |
| Pattern Definition | | |
| Route Pattern * | 67.@ | |
| Route Partition | < None > | |
| Description | CLIR | |
| Numbering Plan * | NANP | |
| Route Filter | < None > | |
| MLPP Precedence * | Default | |
| <input type="checkbox"/> Apply Call Blocking Percentage | | |
| Resource Priority Namespace Network Domain | < None > | |
| Route Class * | Default | |
| Gateway/Route List * | PSTN (Edit) | |
| Route Option | <input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern No Error | |
| Call Classification * | OffNet | |
| External Call Control Profile | < None > | |
| <input type="checkbox"/> Allow Device Override <input checked="" type="checkbox"/> Provide Outside Dial Tone <input type="checkbox"/> Allow Overlap Sending <input type="checkbox"/> Urgent Priority | | |
| <input type="checkbox"/> Require Forced Authorization Code | | |
| Authorization Level * | 0 | |
| <input type="checkbox"/> Require Client Matter Code | | |

Cisco UCM: Route Pattern: Restrict Caller ID (2/2)

| | | |
|-------------------------------------------------------------------------|------------------------|-------------------------|
| Calling Party Transformations | | |
| <input type="checkbox"/> Use Calling Party's External Phone Number Mask | | |
| Calling Party Transform Mask | <input type="text"/> | |
| Prefix Digits (Outgoing Calls) | <input type="text"/> | |
| Calling Line ID Presentation* | Restricted | |
| Calling Name Presentation* | Restricted | |
| Calling Party Number Type* | Cisco CallManager | |
| Calling Party Numbering Plan* | Cisco CallManager | |
| Connected Party Transformations | | |
| Connected Line ID Presentation* | Default | |
| Connected Name Presentation* | Default | |
| Called Party Transformations | | |
| Discard Digits | PreDot | |
| Called Party Transform Mask | <input type="text"/> | |
| Prefix Digits (Outgoing Calls) | <input type="text"/> | |
| Called Party Number Type* | Cisco CallManager | |
| Called Party Numbering Plan* | Cisco CallManager | |
| ISDN Network-Specific Facilities Information Element | | |
| Network Service Protocol | -- Not Selected -- | |
| Carrier Identification Code | <input type="text"/> | |
| Network Service | Service Parameter Name | Service Parameter Value |
| -- Not Selected -- | < Not Exist > | |

The route pattern 2900 was configured to route the voicemail pilot number (2900) to the Unity Connection server as shown in the following screenshots.

Cisco UCM: Route Pattern: Voicemail Pilot Number (1/2)

| | | |
|---------------------------------------------------------|-----------------------------------------------------|---------------------------------------------------------------------|
| Route Pattern Configuration | | Related Links: Back To Find/List Go |
| Save | Delete | Copy Add New |
| Pattern Definition | | |
| Route Pattern* | 2900 | |
| Route Partition | < None > | |
| Description | <input type="text"/> | |
| Numbering Plan | -- Not Selected -- | |
| Route Filter | < None > | |
| MLPP Precedence* | Default | |
| <input type="checkbox"/> Apply Call Blocking Percentage | | |
| Resource Priority Namespace Network Domain | < None > | |
| Route Class* | Default | |
| Gateway/Route List* | ToUnityConnection (Edit) | |
| Route Option | <input checked="" type="radio"/> Route this pattern | |

Cisco UCM: Route Pattern: Voicemail Pilot Number (2/2)

| | | |
|-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|---------------------------------------------------------------------------------|-------------------------------------------|
| <input type="radio"/> Block this pattern No Error | | |
| Call Classification* | OnNet | |
| External Call Control Profile | < None > | |
| <input type="checkbox"/> Allow Device Override <input type="checkbox"/> Provide Outside Dial Tone <input type="checkbox"/> Allow Overlap Sending <input type="checkbox"/> Urgent Priority | | |
| <input type="checkbox"/> Require Forced Authorization Code | | |
| Authorization Level* | <input style="width: 100%;" type="text" value="0"/> | |
| <input type="checkbox"/> Require Client Matter Code | | |
| Calling Party Transformations | | |
| <input type="checkbox"/> Use Calling Party's External Phone Number Mask | | |
| Calling Party Transform Mask | <input style="width: 100%;" type="text"/> | |
| Prefix Digits (Outgoing Calls) | <input style="width: 100%;" type="text"/> | |
| Calling Line ID Presentation* | Default | |
| Calling Name Presentation* | Default | |
| Calling Party Number Type* | Cisco CallManager | |
| Calling Party Numbering Plan* | Cisco CallManager | |
| Connected Party Transformations | | |
| Connected Line ID Presentation* | Default | |
| Connected Name Presentation* | Default | |
| Called Party Transformations | | |
| Discard Digits | < None > | |
| Called Party Transform Mask | <input style="width: 100%;" type="text"/> | |
| Prefix Digits (Outgoing Calls) | <input style="width: 100%;" type="text"/> | |
| Called Party Number Type* | Cisco CallManager | |
| Called Party Numbering Plan* | Cisco CallManager | |
| ISDN Network-Specific Facilities Information Element | | |
| Network Service Protocol | -- Not Selected -- | |
| Carrier Identification Code | <input style="width: 100%;" type="text"/> | |
| Network Service | Service Parameter Name | Service Parameter Value |
| -- Not Selected -- | < Not Exist > | <input style="width: 100%;" type="text"/> |
| <input type="button" value="Save"/> <input type="button" value="Delete"/> <input type="button" value="Copy"/> <input type="button" value="Add New"/> | | |

Voicemail Configuration

A Cisco UCM - Cisco Unity Connection SIP integration was performed to test voicemail scenarios. Below is the configuration on Cisco UCM and Unity Connection.

Configure Voicemail Pilot and Voicemail Profile on Cisco UCM

1. Navigate to **Advanced Features > Voicemail > Voicemail Pilot**.
2. Add a new pilot number. 2900 was used in this example.

3. Check the **Make this the default Voice Mail Pilot for the System** check box.

Cisco UCM: Add Voicemail Pilot Number

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Voice Mail Pilot Configuration

Save Delete Add New

Status
Status: Ready

Voice Mail Pilot Information

Voice Mail Pilot Number: 2900

Calling Search Space: < None >

Description:

Make this the default Voice Mail Pilot for the system

Save Delete Add New

4. Configure a **Voicemail Profile** with this pilot number as shown below.
5. Check the **Make this the default Voice Mail Pilot for the System** check box.

Cisco UCM: Voicemail Profile

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Voice Mail Profile Configuration

Save

Status
Status: Ready

Voice Mail Profile Information

Voice Mail Profile Name*: UnityConnection

Description:

Voice Mail Pilot**: 2900/< None >

Voice Mail Box Mask:

Make this the default Voice Mail Profile for the System

Save

Configuration on Unity Connection: Add New Phone System

To configure a new phone system after logging into Unity Connection, follow this procedure.

1. Navigate to **Telephony Integrations > Phone System**.
2. Click **Add New**.

Cisco Unity Connection: Phone System

Cisco Unity Connection Administration
For Cisco Unified Communications Solutions

Phone System Basics (CUCM11.0)

Phone System Edit Refresh Help

Save Delete Previous Next

Phone System

Phone System Name* CUCM11.0

Default TRAP Phone System

Message Waiting Indicators

Send Message Counts

Use Same Port for Enabling and Disabling MWIs

Force All MWIs Off for this Phone System

Run Synchronize All MWIs on This Phone System

Call Loop Detection by Using DTMF

Enable for Supervised Transfers

Enable for Forwarded Message Notification Calls (by Using DTMF)

DTMF Tone To Use A

Guard Time 2500 milliseconds

Call Loop Detection by Using Extension

Enable for Forwarded Message Notification Calls (by Using Extension)

Phone View Settings

Enable Phone View

CTI Phone Access Username

CTI Phone Access Password

Outgoing Call Restrictions

Enable outgoing calls

Disable all outgoing calls immediately

Disable all outgoing calls between

Beginning Time: 12 00 AM

Ending Time: 12 00 AM

Save Delete Previous Next

Fields marked with an asterisk (*) are required.

3. Configure the **Phone System Name**. CUCM11.0 was used in this example.
4. Click **Save**.
5. Add a new **Port group** as shown in the screenshot below.

Cisco Unity Connection: Add New Port Group

The screenshot displays the 'New Port Group' configuration page in the Cisco Unity Connection Administration interface. The page is titled 'New Port Group' and includes a navigation menu on the left with categories like 'Users', 'Class of Service', 'Templates', 'Contacts', 'Distribution Lists', 'Call Management', 'Message Storage', 'Networking', 'Unified Messaging', 'Video', 'Dial Plan', 'System Settings', and 'Telephony Integrations'. The main content area contains the following fields and options:

- Phone System:** A dropdown menu set to 'CUCM11.0'.
- Create From:** Radio buttons for 'Port Group Type' (selected) and 'Port Group'. The 'Port Group Type' dropdown is set to 'SIP'.
- Port Group Description:**
 - Display Name*:** Text input field containing 'CUCM11.0-1'.
 - Authenticate with SIP Server**
 - Authentication Username:** Text input field.
 - Authentication Password:** Text input field.
 - Contact Line Name:** Text input field.
 - SIP Security Profile:** Dropdown menu set to '5060'.
 - SIP Transport Protocol:** Dropdown menu set to 'TCP'.
- Primary Server Settings:**
 - IPv4 Address or Host Name:** Text input field containing '10.80.25.2'.
 - IPv6 Address or Host Name:** Text input field.
 - Port:** Text input field containing '5060'.

At the bottom of the form, there is a 'Save' button and a note: 'Fields marked with an asterisk (*) are required.'

- a. On the **Phone System Basics** page, in the **Related Links** drop-down box, select **Add Port Group** and select **Go**.
 - b. **Phone System:** Select the one created earlier. CUCM11.0 was used in this example.
 - c. **Create From:** Select **Port Group Type** and select **SIP** from the drop-down menu.
 - d. **IPv4 Address or Host Name:** Enter the IP address (or host name) of the primary Cisco UCM server that is being integrated with Cisco Unity Connection.
 - e. Click **Save**.
6. On the Port Group Basics page, in the Related Links drop-down box, select **Add Ports**, and select **Go**.

Cisco Unity Connection: Port Group Added: Related Links to Add Port

Cisco Unity Connection Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unity Connection Administration | administrator | Search Documentation | About | Sign Out

Cisco Unity Connection

- Users
- Class of Service
- Templates
- Contacts
- Distribution Lists
- Call Management
- Message Storage
- Networking
- Unified Messaging
- Video
- Dial Plan
- System Settings
- Telephony Integrations
 - Phone System
 - Port Group
 - Port
 - Speech Connect Port
 - Trunk
 - Security
- Tools

Port Group Basics (CUCM11.0-1)

Search Port Groups | Port Group Basics (CUCM11.0-1)
Related Links: Add Ports

Port Group Edit Refresh Help

Save Delete Previous Next

Status

⚠ The phone system cannot take calls if it has no ports. Use the Related Links to add ports.

Port Group

Display Name* CUCM11.0-1

Integration Method SIP

Reset Status Reset Not Required Reset

Session Initiation Protocol (SIP) Settings

Register with SIP Server

Authenticate with SIP Server

Authentication Username

Authentication Password

Contact Line Name

SIP Security Profile 5060

SIP Transport Protocol TCP

Advertised Codec Settings

Change Advertising

| Display Name | Packet Size |
|--------------|-------------|
| G.711 mu-law | 20 |
| G.729 | 20 |

Change Advertising

Message Waiting Indicator Settings

Enable Message Waiting Indicators

Delay between Requests 0 milliseconds

Maximum Concurrent Requests 0

Retries After Successful Attempt 0

Retry Interval After Successful Attempt 5 milliseconds

Save Delete Previous Next

- On the **New Port** page, configure the settings as shown below and select **Save**.

Cisco Unity Connection: Add New Port

The screenshot shows the Cisco Unity Connection Administration web interface. The page title is "New Port". The navigation menu on the left includes: Users, Class of Service, Templates, Contacts, Distribution Lists, Call Management, Message Storage, Networking, Unified Messaging, Video, Dial Plan, System Settings, and Telephone Integrations. Under Telephone Integrations, the following options are listed: Phone System, Port Group, Port, Speech Connect Port, Trunk, Security, and Tools. The "New Phone System Port" section is highlighted with a red box and contains the following configuration options:

- Enabled
- Number of Ports:
- Phone System:
- Port Group:
- Server:

The "Port Behavior" section includes the following options:

- Answer Calls
- Perform Message Notification
- Send MWI Requests (may also be disabled by the port group)
- Allow TRAP Connections

The "Status" section contains two warning messages:

- ⚠ Because it has no port groups, PhoneSystem is not listed in the Phone system field.
- ⚠ Because it has no port groups, test is not listed in the Phone system field.

8. Add the Cisco UCM subscriber IP to the list of AXL servers for this phone system.

Cisco Unity Connection: Edit AXL Servers

The screenshot shows the Cisco Unity Connection Administration interface. The left sidebar contains a navigation menu with 'Telephony Integrations' expanded to 'Phone System'. The main content area is titled 'Edit AXL Servers' and includes a 'Save' button, a table of AXL Servers, and an 'AXL Server Settings' section.

| <input type="checkbox"/> | Order | IP Address | Port | |
|--------------------------|-------|------------|------|------|
| <input type="checkbox"/> | 0 | 10.80.25.2 | 5060 | Test |
| <input type="checkbox"/> | 1 | 10.80.25.3 | 5060 | Test |

AXL Server Settings

Username: administrator
Password:
Cisco Unified Communications Manager Version: 5.0 or Greater (SSL) v

- a. Navigate to **Telephony Integrations > Phone System > CUCM11.0**.
 - b. On the Phone System Basics, click **Edit > Cisco Unified Communications Manager AXL Servers**.
 - c. Click **Add New** or in the second row, configure the Cisco UCM Subscriber IP and port. 10.80.25.3 and 5060 was used in this example.
9. Click **Save**.

Configure a Voicemail User

To configure a new user that would have a voicemail box, after logging into Unity Connection, perform the following procedure.

1. Navigate to **Users > Users**.
2. Click **Add New**.

Cisco Unity Connection: Add User

The screenshot shows the Cisco Unity Connection Administration interface. The left sidebar contains a navigation tree with 'Users' highlighted. The main content area is titled 'New User' and contains the following configuration options:

- User Type:** User With Mailbox
- Based on Template*:** voicemailusertemplate
- Name:** Alias* (Crestron_Mercury)
- Mailbox Store:** Unity Messaging Database -1
- Phone:** Extension* (2600)

A 'Save' button is located at the top left of the form area.

3. Configure a **Based on Template** from the drop-down menu. *voicemailusertemplate* was used in this example.
4. Configure an **Alias**. *Crestron_Mercury* was used in this example.
5. Configure an **Extension** for the user. *2600* was used in this example.
6. Click **Save**.
7. On the screen that follows, configure the **Phone System**.

Cisco Unity Connection: Assign Phone System to User

The screenshot displays the Cisco Unity Connection Administration interface. The left sidebar shows a navigation tree with categories like Users, Class of Service, Templates, Contacts, Distribution Lists, Call Management, Message Storage, Networking, and Unified Messaging. The main content area is titled 'Edit User Basics (Crestron_Mercury)'. It includes a header with 'User Edit Refresh Help' and buttons for 'Save', 'Delete', 'Previous', and 'Next'. The form fields are as follows:

| Name | |
|--------------|----------------------------------------------|
| Alias* | Crestron_Mercury |
| First Name | |
| Last Name | |
| Display Name | Crestron_Mercury |
| SMTP Address | crestron_mercury@clus35cuc.lab.tekvizion.com |
| Initials | |
| Title | |
| Employee ID | |

LDAP Integration Status

Integrate with LDAP Directory
 Do Not Integrate with LDAP Directory

Phone

| | |
|----------------------------------------|------------------------|
| Extension* | 2600 |
| Cross-Server Transfer Extension or URI | |
| Outgoing Fax Number | |
| Outgoing Fax Server | --- Not Selected --- |
| Partition | clus35cuc Partition |
| Search Scope | clus35cuc Search Space |
| Phone System | CUCM11.0 |
| Class of Service | Voice Mail User COS |
| Active Schedule | Weekdays |

Set for Self-enrollment at Next Sign-In

- Select the Phone System configured earlier from the drop-down menu. **CUCM11.0** was used in this example.
- Click **Save**.

This page is intentionally left blank.

Crestron Electronics, Inc.
15 Volvo Drive Rockleigh, NJ 07647
Tel: 888.CRESTRON
Fax: 201.767.7576
www.crestron.com



Configuration Guide – DOC. 7981A
(2048641)
04.17
Specifications subject to
change without notice.