



# **CCS-UC-1**

## SIP Endpoint with Cisco<sup>®</sup> Unified Communications Manager 10.5

Configuration Guide

Crestron Electronics, Inc.

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# CCS-UC-1: SIP Endpoint with Cisco Unified Communications Manager 10.5

## Introduction

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

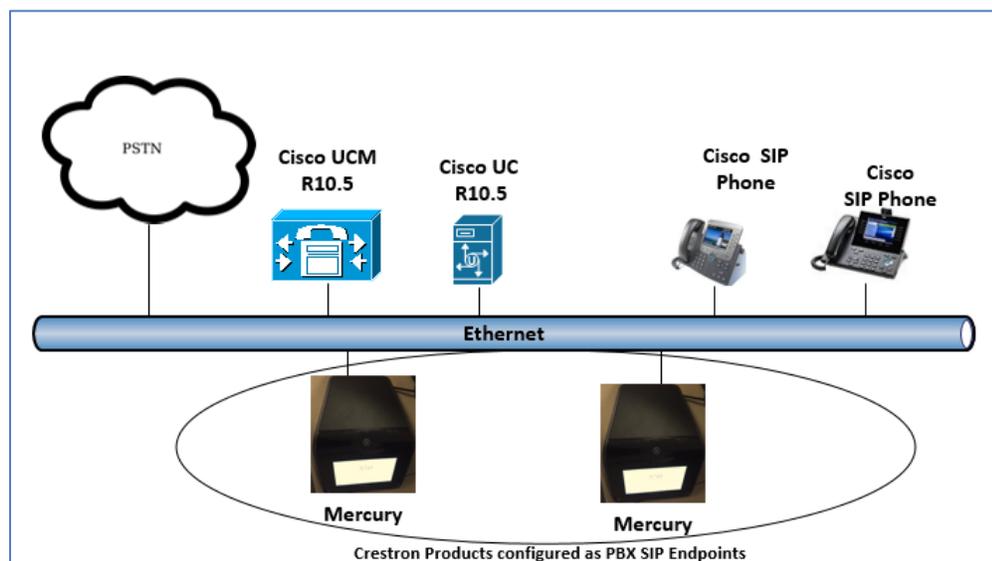
## Audience

This document is intended for users attempting to configure and use the Crestron Mercury devices as SIP endpoints registering to the Cisco Unified Communications Manager (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 SIP phones
- Crestron Mercury devices as SIP endpoints

## Software Requirements

- Cisco Unified Communication Manager v 10.5.2.13900-12
- Cisco Unity Connection v 10.5.2.13900-12
- Crestron Mercury device v 1.3353.00029

## Hardware Requirements

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

## Product Description

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

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

### *Features Supported*

- Registration with digest authentication
- Basic Calls with G729, 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
- Voice mail 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
- Initiating attended call transfer
- Initiating semi-attended call transfer
- Initiating blind call transfer
- Configuration of shared line on device
- Initiating call park
- Do Not Disturb (DND)

### **Known Issues and Limitations**

- The Crestron Mercury device does not allow calls to be placed or accepted as long as the SIP server is unreachable and the device is un-registered. This causes an established active call to be dropped in a scenario where there is a PBX network outage due to the device losing its registration status. This issue is tracked via Crestron's Bugzilla™ software Defect: 128016.
- Caller ID is not supported on the Crestron Mercury device. Currently, only the calling party number is displayed as the caller ID. This issue is tracked via Crestron's Bugzilla software Defect: 119006.
- The active call timer on the Crestron Mercury device does not reflect the correct call duration. The active call duration includes the time for which the unit was being alerted also. This issue is tracked via Crestron's Bugzilla software Defect: 124001.
- The first ringback heard on Crestron Mercury device is stuttered. It resembles a mix of local and remote ringback. This issue is tracked via Crestron's Bugzilla software Defect: 122421.
- On the Crestron Mercury web user interface, there is currently no notification provided to the user when certain mandatory configurations are missing. This issue is tracked via Crestron's Bugzilla software Defect: 125193.
- On the Crestron Mercury web user interface, a configuration of DHCP OFF on the Network configuration page mandates configuration of both the adapters. The user is unable to save changes unless both the adapters are configured and is notified of an invalid IP against the default of 0.0.0.0 for an unused adapter. This issue is tracked via Crestron's Bugzilla software Defect: 126236.

- On the Crestron Mercury device, for certain called numbers that cannot be reached or are invalid, the user only hears a reorder tone and does not have the option to disconnect the call except by pressing the call button again. This issue is tracked via Crestron’s Bugzilla software Defect: 122633.

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## Crestron Mercury Configuration

### Setup

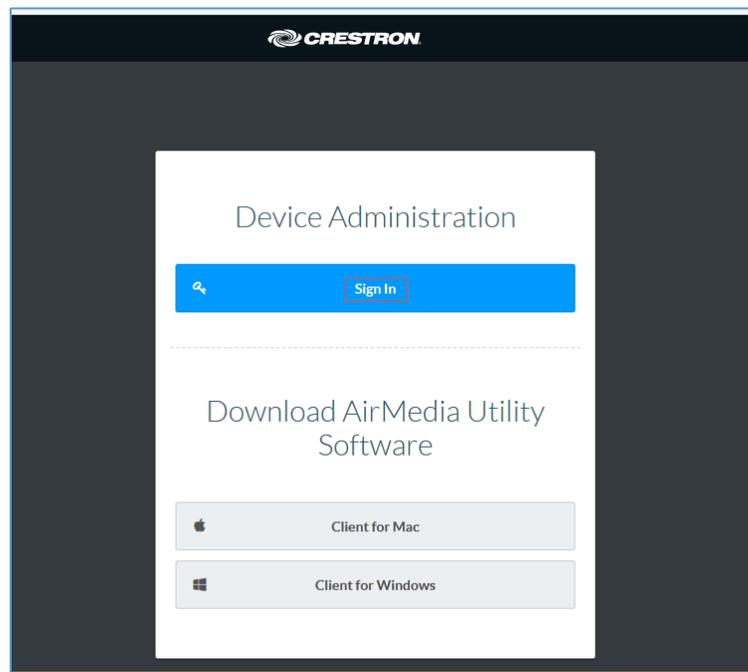
The LAN port of the Crestron Mercury device needs to be connected to one PoE+ port to power it up and network for connectivity with the Cisco UCM. The PoE+ switch that is used should have the LLDP functionality enabled for the device to power up and be completely functional. By default, the “poeplus” configuration is set to Off on the device.

### Configuring the device

To configure the Crestron Mercury device, follow this procedure:

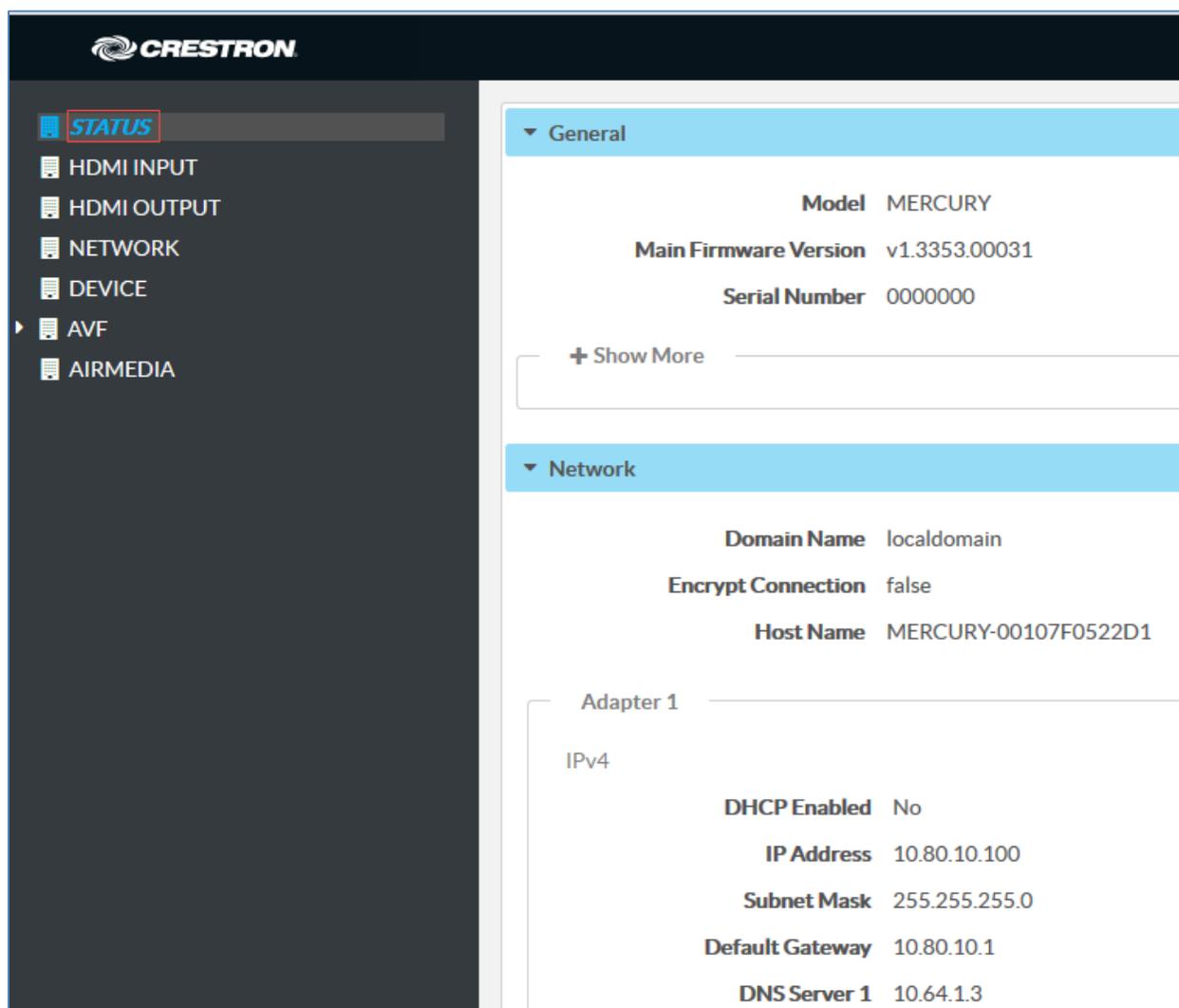
1. Access the web GUI for the device by using an http session with the device’s IP address. 10.80.10.100 was used in this example as the device IP. The initial page that displays is shown below.

*Crestron Mercury Configuration: Login to Web GUI*



2. Click **Sign In** and log in to the device. For information on device administration, refer to the CCS-UC-1 Supplemental Guide (Doc. 7844) at [www.crestron.com/manuals](http://www.crestron.com/manuals).

The Status screen that appears displays basic information on the device.



**CRESTRON**

**STATUS**

- HDMI INPUT
- HDMI OUTPUT
- NETWORK
- DEVICE
- ▶ AVF
- AIRMEDIA

▼ General

**Model** MERCURY

**Main Firmware Version** v1.3353.00031

**Serial Number** 0000000

+ Show More

▼ Network

**Domain Name** localdomain

**Encrypt Connection** false

**Host Name** MERCURY-00107F0522D1

Adapter 1

IPv4

**DHCP Enabled** No

**IP Address** 10.80.10.100

**Subnet Mask** 255.255.255.0

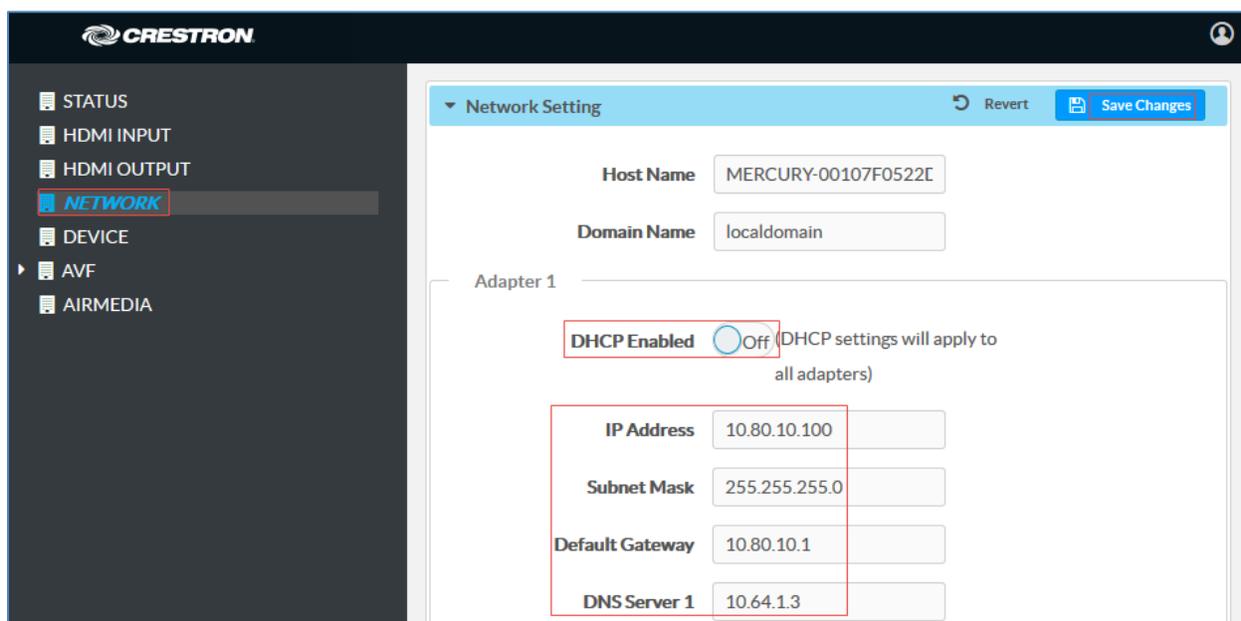
**Default Gateway** 10.80.10.1

**DNS Server 1** 10.64.1.3

The device can be configured from the **Network** page.

3. On the web GUI, navigate to **Network**.

*Crestron Mercury Configuration: Network Setting: DHCP Off: Static IP Configured*



4. Enter the following parameters in the **Adapter 1** section to configure the Crestron Mercury device.
    - **DHCP:** Either of the two can be chosen:
      - Obtain an IP address automatically
      - Use the following IP address
- For the test, a static IP was configured.
- **IP address:** *10.80.10.100* was used in this example.
  - **Subnet Mask:** *255.255.255.0* was used in this example.
  - **Default Gateway:** *10.80.10.1* was used in this example.
  - **DNS Server 1:** *10.64.1.3* was used in this example.
5. Click **Save Changes**.

## Configuring the SIP Parameters

1. On the web GUI, navigate to **Device > SIP Calling**

### *Crestron Mercury Configuration: SIP Calling Parameters*

The screenshot displays the Crestron Mercury Configuration web GUI. On the left is a dark sidebar with a menu: STATUS, HDMI INPUT, HDMI OUTPUT, NETWORK, **DEVICE** (highlighted), AVF, and AIRMEDIA. The main content area is titled 'SIP Calling' and features a 'Revert' button and a 'Save Changes' button. Below this, there is a section for 'Enable SIP' with a toggle switch set to 'On'. A red box highlights the 'Enable SIP' toggle and the 'Transport Type' dropdown menu (set to UDP). Another red box highlights the 'Server IP Address' (10.80.10.2), 'Port' (5060), 'Server Username' (Mercury\_2600), and 'Server Password' (masked with dots) fields. Below these, the 'Server Realm' is set to '\*'. A third red box highlights the 'Local Extension' field (2600). The 'Proxy Server' is set to 'NONE'. At the bottom, the 'SIP Server Status' is shown as 'Online'.

2. Enable the check box for **Enable SIP**.
3. Configure the **Server IP Address**: Enter the IP Address of the primary Cisco UCM node. *10.80.10.2* was used in this example.
4. Configure the **Port**: *5060* was used in this example.
5. Configure the **Server Username**: Enter the end user configured on Cisco UCM for this device. *Mercury\_2600* was used in this example.
6. Configure the **Server Password**: Enter the password as configured on Cisco UCM for this end user.
7. Configure the **Local Extension**: Enter the directory number that was configured for this device on Cisco UCM. *2600* 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 Mercury device as a basic SIP endpoint.

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

## Configure the End User

To configure the end user, follow this procedure:

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 **X** Delete **+** Add New

#### User Information

User Status: Enabled Local User

User ID\*: Mercury\_2600

Password: [Masked] **Edit Credential**

Confirm Password: [Masked]

Self-Service User ID: [Empty]

PIN: [Masked] **Edit Credential**

Confirm PIN: [Masked]

Last name\*: Mercury2600

Middle name: [Empty]

First name: [Empty]

Display name: [Empty]

Title: [Empty]

Directory URI: [Empty]

Telephone Number: [Empty]

Home Number: [Empty]

Mobile Number: [Empty]

Pager Number: [Empty]

Mail ID: [Empty]

Manager User ID: [Empty]

Department: [Empty]

User Locale: < None > ▾

Associated PC: [Empty]

Digest Credentials: [Masked]

Confirm Digest Credentials: [Masked]

User Profile: Use System Default( "Standard (Factory Default) U:' [View Details](#)

3. Configure **User ID**: Enter a unique end user identification name. Two users were configured for this example for the Crestron Mercury devices: *Mercury\_2600* and *Mercury\_2602*.
4. Configure **Password**: Enter any password. This same password will be entered on the device for the SIP Server Password. *123456* was used in this example.
5. Confirm **Password**: Re-enter the same password entered above.
6. Configure the **Last Name**: Enter the end user's last name.
7. Configure the **Digest Credentials**: Enter a string of alphanumeric characters.
8. Confirm the **Digest Credentials**: Re-enter the digest credentials configured above.
9. Click **Save**. All of the configured users are listed .

*Cisco UCM: End Users configured for all Mercury devices*

The screenshot shows the Cisco Unified CM Administration interface. At the top, there is a navigation bar with the Cisco logo and the text "Cisco Unified CM Administration For Cisco Unified Communications Solutions". The user is logged in as "administrator". Below the navigation bar, there are several tabs: System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The "User Management" tab is selected, and the "Find and List Users" section is active. In this section, there are buttons for "Add New", "Select All", "Clear All", and "Delete Selected". Below these buttons, a status box indicates "2 records found". A table titled "User (1 - 2 of 2)" displays the following data:

	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 table, there are buttons for "Add New", "Select All", "Clear All", and "Delete Selected".

## Configure Region for G729

To test the device capabilities with G729, a separate region with the G.729 codec as preference needs to be configured. This new region needs to be assigned to the default device pool.

To configure a new region, perform the following procedure.

1. Navigate to **System > Region Information > Region**.
2. Click **Add New**.

### Cisco UCM: Region Configuration

The screenshot displays the Cisco Unified CM Administration interface for configuring a region. The top navigation bar includes 'System', 'Call Routing', 'Media Resources', 'Advanced Features', 'Device', 'Application', 'User Management', 'Bulk Administration', and 'Help'. The main content area is titled 'Region Configuration' and includes a 'Related Links' section with 'Back To Find/List'. Below this are icons for 'Save', 'Delete', 'Reset', 'Apply Config', and 'Add New'. The 'Region Information' section shows the 'Name' field set to 'G729'. The 'Region Relationships' table is as follows:

Region	Audio Codec Preference List	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls	Maximum Session Bit Rate for Immersive Video Calls
G729	Factory Default lossy	8 kbps (G.729)	Use System Default (384 kbps)	Use System Default (2000000000 kbps)
NOTE: Regions not displayed	Use System Default	Use System Default	Use System Default	Use System Default

The 'Modify Relationship to other Regions' section shows a list of regions with 'G729' selected. Below the list are four columns of radio button options for setting relationships: 'Keep Current Setting', 'Use System Default', 'None', and 'Keep Current Setting'.

3. Configure a **Name**: G729 was used in this example.
4. Click **Save**.
5. On the screen that follows, select the newly added region in the lower pane and select the **Maximum Audio bit Rate** from the drop-down menu as *8kbps (G.729)*.
6. Click **Save**.

## Configure a SIP Profile

For the example, a new SIP profile, **Standard SIP Profile\_Crestron**, was configured and assigned to the trunk used for PSTN calls.

To add a new SIP profile, follow this procedure.

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

### *Cisco UCM: SIP Profile Configuration (1/4)*

The screenshot shows the Cisco Unified CM Administration interface for configuring a SIP profile. The page title is "SIP Profile Configuration" and it includes a navigation menu at the top. The main content area is divided into two sections: "SIP Profile Information" and "SDP Information".

**SIP Profile Information**

- Name\*: Standard SIP Profile\_Crestron
- Description: Crestron-SIPProfile
- 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-Agen
- Version in User Agent and Server Header\*: Major And Minor
- Dial String Interpretation\*: Phone number consists of characters 0-9, \*, #, anc
- Confidential Access Level Headers\*: Disabled

**SDP Information**

- 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

There are also several checkboxes for various options:

- Redirect by Application
- Disable Early Media on 180
- Outgoing T.38 INVITE include audio mline
- Use Fully Qualified Domain Name in SIP Requests
- Assured Services SIP conformance
- Require SDP Inactive Exchange for Mid-Call Media Change
- Allow RR/RS bandwidth modifier (RFC 3556)

2. On the screen that appears, click **Add New** and configure the SIP Profile.
  - a. Assign a **Name**. *Standard SIP Profile\_Crestron* was used for this example.

Cisco UCM: SIP Profile Configuration (2/4)

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)	

---

**Parameters used in Phone**

Timer Invite Expires (seconds)*	180
Timer Register Delta (seconds)*	5
Timer Register Expires (seconds)*	3600
Timer T1 (msec)*	500
Timer T2 (msec)*	4000
Retry INVITE*	6
Retry Non-INVITE*	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*	16384
Stop Media Port*	32766
DSCP for Audio Calls	Use System Default
DSCP for Video Calls	Use System Default
DSCP for Audio Portion of Video Calls	Use System Default
DSCP for TelePresence Calls	Use System Default
DSCP for Audio Portion of TelePresence Calls	Use System Default
Call Pickup URI*	x-cisco-serviceuri-pickup

Cisco UCM: SIP Profile Configuration (3/4)

Call Pickup Group Other URI*	x-cisco-serviceuri-opickup
Call Pickup Group URI*	x-cisco-serviceuri-gpickup
Meet Me Service URI*	x-cisco-serviceuri-meetme
User Info*	None ▾
DTMF DB Level*	Nominal ▾
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	

## 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
RSVP Over SIP*	Local RSVP
Resource Priority Namespace List	< None >
<input type="checkbox"/> Fall back to local RSVP	
SIP Rel1XX Options*	Disabled
Video Call Traffic Class*	Mixed
Calling Line Identification Presentation*	Default
Session Refresh Method*	Invite
Early Offer support for voice and video calls*	Best Effort (no MTP inserted)
<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
Ping Interval for Out-of-service Trunks (seconds)*	120
Ping Retry Timer (milliseconds)*	500
Ping Retry Count*	6

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	

Save Delete Copy Reset Apply Config Add New

- b. Configure **Early Offer support for voice and video calls** as *Best Effort (no MTP inserted)*.
3. Retain all other default configurations.
4. Click **Save**, and then click **Apply Config**.

## Configure Phone Security Profile

For the example, a phone security profile was configured for the Crestron Mercury device with digest authentication enabled.

To configure the common Phone Security Profile, perform the following procedure.

1. Navigate to **System > Security > Phone Security Profile**.
2. Click **Add New**.

*Cisco UCM: Phone Security Profile Configuration for Crestron Mercury*

**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

3. Configure a **Name**: *Crestron* was used in this example.
4. Configure **Transport Type**: *TCP+UDP*.
5. Check the **Enable Digest Authentication** check box.
6. Configure **SIP Phone Port**: *5060*.
7. Click **Save**.

## Configure the Crestron Mercury Device as a Third-party SIP Device

To configure the Crestron Mercury device as a third-party SIP device, follow this procedure:

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

*Cisco UCM: Phone Configuration: Add Crestron Mercury Device as Third-party SIP Device (1/2)*

The screenshot displays the Cisco Unified CM Administration interface for configuring a phone. The top navigation bar shows 'Cisco Unified CM Administration' and 'administrator'. The main menu includes 'System', 'Call Routing', 'Media Resources', 'Advanced Features', 'Device', 'Application', 'User Management', 'Bulk Administration', and 'Help'. The 'Phone Configuration' section is active, with a toolbar containing 'Save', 'Delete', 'Copy', 'Reset', 'Apply Config', and 'Add New'. The configuration is divided into four main sections:

- Association:** Shows two lines: 'Line [1] - 2600 (no partition)' and 'Line [2] - Add a new DN'. A 'Modify Button Items' button is present.
- Phone Type:** 'Product Type' is set to 'Third-party SIP Device (Basic)' and 'Device Protocol' is 'SIP'.
- Real-time Device Status:** 'Registration' is 'Registered with Cisco Unified Communications Manager clus20pub' and 'IPv4 Address' is '10.80.10.100'. 'Active Load ID' and 'Download Status' are both 'None'.
- Device Information:** 'Device is Active' is checked. 'Device is not trusted' is indicated with a warning icon. 'MAC Address\*' is '00107F8B67B8' and 'Description' is 'SEP00107F8B67B8'. 'Device Pool\*' is 'Default'. 'Common Device Configuration' is '< None >'. 'Phone Button Template\*' is 'Third-party SIP Device (Basic)'. 'Common Phone Profile\*' is 'Standard Common Phone Profile'. 'Calling Search Space' and 'AAR Calling Search Space' are both '< None >'. 'Media Resource Group List' is 'MRGL'. 'Location\*' is 'Hub\_None'. 'AAR Group' is '< None >'. 'Device Mobility Mode\*' is 'Default'. 'Owner' is 'User'. 'Owner User ID\*' is 'Mercury\_2600'.

3. Select **Product Type** as **Third-party SIP Device (Basic)**.
4. Click **Next**.
5. Configure **MAC Address**: Enter MAC address of the 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 menu. *Mercury\_2600* was selected in this example for the first Crestron Mercury device, and *Mercury\_2602* for the second Crestron Mercury device.

Cisco UCM: Phone Configuration: Add Crestron Mercury Device as Third-party SIP Device (2/2)

Use Trusted Relay Point*	Off	▼
Always Use Prime Line*	Default	▼
Always Use Prime Line for Voice Message*	Default	▼
Geolocation	< None >	▼
<input type="checkbox"/> Ignore Presentation Indicators (internal calls only)		
<input checked="" type="checkbox"/> Logged Into Hunt Group		
<input type="checkbox"/> Remote Device		
<b>Number Presentation Transformation</b>		
<b>Caller ID For Calls From This Phone</b>		
Calling Party Transformation CSS	< None >	▼
<input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS (Caller ID For Calls From This Phone)		
<b>Remote Number</b>		
Calling Party Transformation CSS	< None >	▼
<input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS (Device Mobility Related Information)		
<b>Protocol Specific Information</b>		
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_Crestron	▼ <a href="#">View Details</a>
Digest User	Mercury_2600	▼
<input type="checkbox"/> Media Termination Point Required		
<input type="checkbox"/> Unattended Port		
<input type="checkbox"/> Require DTMF Reception		
<b>MLPP and Confidential Access Level Information</b>		
MLPP Domain	< None >	▼
Confidential Access Mode	< None >	▼
Confidential Access Level	< None >	▼

9. Select **Device Security Profile** as configured earlier from the drop-down menu. *Crestron* was used in this test.
10. Select **SIP Profile** as configured earlier from the drop-down menu. *Standard SIP Profile\_Crestron* was used in this example.
11. Select **Digest User**: select the End User configured earlier from the drop-down menu. *Mercury\_2600* was selected in this example for the first Crestron Mercury device, and *Mercury\_2602* for the second Crestron Mercury device.
12. Click **Save**.

13. Add a **DN** to this phone. DN 2600 was configured for one of the Crestron Mercury devices in this test. Similarly, DN 2602 was added to the other Crestron Mercury device.

*Cisco UCM: Directory Number Configuration: Add DN to Crestron Device: Third-party SIP Device (1/5)*

**Directory Number Configuration** Related Links: [Configure Device \(SEP00107F8B67B8\)](#)

---

**Status**

Status: Ready

---

**Directory Number Information**

Directory Number*	2600	<input type="checkbox"/> Urgent Priority
Route Partition	< None >	
Description		
Alerting Name	Mercury2600	
ASCII Alerting Name	Mercury2600	
External Call Control Profile	< None >	
Line Group	Crestron	<input type="button" value="Edit Line Group"/>
Associated Devices	SEP00107F8B67B8	<input type="button" value="Edit Device"/> <input type="button" value="Edit Line Appearance"/>
▼ ▲		
Dissociate Devices		

---

**Directory Number Settings**

Voice Mail Profile	< None >	(Choose <None> to use system default)
Calling Search Space	< None >	
BLF Presence Group*	Standard Presence group	
User Hold MOH Audio Source	< None >	
Network Hold MOH Audio Source	< None >	

Reject Anonymous Calls

Cisco UCM: Directory Number Configuration: Add DN to Crestron Device: Third-party SIP Device (2/5)

<b>Enterprise Alternate Number</b>			
Add Enterprise Alternate Number			
<b>+E.164 Alternate Number</b>			
Add +E.164 Alternate Number			
<b>Directory URIs</b>			
Primary	URI	Partition	Adverti Global via IL
<input checked="" type="radio"/>	<input type="text"/>	< None >	<input checked="" type="checkbox"/>
Add Row			
<b>PSTN Failover for Enterprise Alternate Number, +E.164 Alternate Number, and URI Dialing</b>			
Advertised Failover Number < None >			
<b>AAR Settings</b>			
	Voice Mail	AAR Destination Mask	AAR Group
AAR	<input type="checkbox"/> or	<input type="text"/>	< None >
<input checked="" type="checkbox"/> Retain this destination in the call forwarding history			
<b>Call Forward and Call Pickup Settings</b>			
	Voice Mail	Destination	Calling Search Space
Calling Search Space Activation Policy			Use System Default

*Cisco UCM: Directory Number Configuration: Add DN to Crestron Device: Third-party SIP Device (3/5)*

Forward All	<input type="checkbox"/> or	<input type="text"/>	< None >
Secondary Calling Search Space for Forward All			< None >
Forward Busy Internal	<input checked="" type="checkbox"/> or	<input type="text"/>	< None >
Forward Busy External	<input checked="" type="checkbox"/> or	<input type="text"/>	< None >
Forward No Answer Internal	<input checked="" type="checkbox"/> or	<input type="text"/>	< None >
Forward No Answer External	<input checked="" type="checkbox"/> or	<input type="text"/>	< None >
Forward No Coverage Internal	<input type="checkbox"/> or	<input type="text"/>	< None >
Forward No Coverage External	<input type="checkbox"/> or	<input type="text"/>	< None >
Forward on CTI Failure	<input type="checkbox"/> or	<input type="text"/>	< None >
Forward Unregistered Internal	<input checked="" type="checkbox"/> or	<input type="text"/>	< None >
Forward Unregistered External	<input checked="" type="checkbox"/> or	<input type="text"/>	< None >
No Answer Ring Duration (seconds)	<input type="text" value="6"/>		
Call Pickup Group			< None >

*Cisco UCM: Directory Number Configuration: Add DN to Crestron Device: Third-party SIP Device (4/5)*

Park Monitoring			
	Voice Mail	Destination	Calling Search Space
Park Monitoring Forward No Retrieve Destination External	<input type="checkbox"/> or	<input type="text"/>	< None > A blank value means to call the parker's line.
Park Monitoring Forward No Retrieve Destination Internal	<input type="checkbox"/> or	<input type="text"/>	< None > A blank value means to call the parker's line.
Park Monitoring Reversion Timer		<input type="text"/>	A blank value will use value set in Park Monitoring Reversion Timer service parameter

MLPP Alternate Party And Confidential Access Level Settings	
Target (Destination)	<input type="text"/>
MLPP Calling Search Space	< None >
MLPP No Answer Ring Duration (seconds)	<input type="text"/>
Confidential Access Mode	< None >
Confidential Access Level	< None >
Call Control Agent Profile	< None >

Cisco UCM: Directory Number Configuration: Add DN to Crestron Device: Third-party SIP Device (5/5)

<b>Line 1 on Device SEP00107F8B67B8</b>	
Display (Caller ID)	Mercury2600 <small>Display text for a line appearance is intended for displaying text such as a name instead of a directory number for calls. If you specify a number, the person receiving a call may not see the proper identity of the caller.</small>
ASCII Display (Caller ID)	Mercury2600
External Phone Number Mask	9722652600
Monitoring Calling Search Space	< None >
<b>Multiple Call/Call Waiting Settings on Device SEP00107F8B67B8</b>	
<small>Note: The range to select the Max Number of calls is: 1-2</small>	
Maximum Number of Calls*	1
Busy Trigger*	1 (Less than or equal to Max. Calls)
<b>Forwarded Call Information Display on Device SEP00107F8B67B8</b>	
<input checked="" type="checkbox"/> Caller Name	
<input type="checkbox"/> Caller Number	
<input type="checkbox"/> Redirected Number	
<input checked="" type="checkbox"/> Dialed Number	
<b>Users Associated with Line</b>	
Associate End Users	
Save Delete Reset Apply Config Add New	

## Configure Media Resource Group and Media Resource Group List

A Media Resource Group (MRG) 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 a 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. The page title is "Media Resource Group Configuration". The navigation bar shows the user is logged in as "administrator". The breadcrumb trail is "System > Call Routing > Media Resources > Advanced Features > Device > Application > User Management > Bulk Administration > Help". The page includes a "Related Links" section with "Back To Find/List" and "Go". The configuration sections are:

- 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**. (These are assumed to have been added earlier and are available for use/registered with this Cisco UCM.)
4. Next, configure the Media Resource Group List **MRGL**.
  - a. Select **Media Resources > Media Resource Group List**.
  - b. Click **Add New**.

Cisco UCM: Media Resource Group List Configuration

The screenshot shows the 'Media Resource Group List Configuration' page in Cisco Unified CM Administration. At the top, the navigation bar includes 'System', 'Call Routing', 'Media Resources', 'Advanced Features', 'Device', 'Application', 'User Management', 'Bulk Administration', and 'Help'. The page title is 'Media Resource Group List Configuration' with a 'Related Links: Back To Find/List' button. Below the title are icons for 'Save', 'Delete', 'Copy', and 'Add New'. The 'Status' section shows 'Status: Ready'. The 'Media Resource Group List Status' section shows 'Media Resource Group List: MRGL (used by 23 devices)'. The 'Media Resource Group List Information' section has a 'Name\*' field containing 'MRGL'. The 'Media Resource Groups for this List' section has two lists: 'Available Media Resource Groups' (empty) and 'Selected Media Resource Groups' (containing 'MRG'). At the bottom are buttons for 'Save', 'Delete', 'Copy', and 'Add New'.

- c. Provide the **Name: MRGL** and select the media resource groups from the **Available Media Resource Groups**.

Cisco UCM: Find and List Device Pools

The screenshot shows the 'Find and List Device Pools' page in Cisco Unified CM Administration. At the top, the navigation bar includes 'System', 'Call Routing', 'Media Resources', 'Advanced Features', 'Device', 'Application', 'User Management', and 'Bulk Administration'. The page title is 'Find and List Device Pools' with buttons for '+ Add New', 'Select All', 'Clear All', and 'Delete Selected'. The 'Status' section shows '1 records found'. Below is a table with columns: 'Name', 'Cisco Unified CM Group', 'Region', 'Date/Time Group', and 'Copy'. The table has one row with values: 'Default', 'Default', 'Default', 'CMLocal'. At the bottom are buttons for 'Add New', 'Select All', 'Clear All', and 'Delete Selected'.

Name	Cisco Unified CM Group	Region	Date/Time Group	Copy
Default	Default	Default	CMLocal	

## Cisco UCM: Device Pool Configuration: Update MRGL on Defalut

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

### Device Pool Configuration

Related Links: [Back To Find/List](#) ▾ Go

Save Delete Copy Reset Apply Config Add New

**Status**

Status: Ready

**Device Pool Information**

Device Pool: Default (22 members\*\*)

**Device Pool Settings**

Device Pool Name*	Default
Cisco Unified Communications Manager Group*	Default ▾
Calling Search Space for Auto-registration	< None > ▾
Adjunct CSS	< None > ▾
Reverted Call Focus Priority	Default ▾
Intercompany Media Services Enrolled Group	< None > ▾

**Roaming Sensitive Settings**

Date/Time Group*	CMLocal ▾
Region*	Default ▾
Media Resource Group List	MRGL_Secure ▾
Location	< None > ▾
Network Locale	< None > ▾
SRST Reference*	Disable ▾

## Configure Trunks

Two trunks were configured for this validation test:

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

### *Cisco UCM- PSTN Gateway Trunk Configuration*

To create a new trunk, perform the following procedure.

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

## Cisco UCM: Trunk Configuration: Add New Trunk

Cisco Unified CM Administration  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration Go

administrator | Search Documentation | About | Logout

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

**Trunk Configuration** Related Links: Back To Find/List ▾ Go

Next

**Status**  
Status: Ready

**Trunk Information**

Trunk Type*	SIP Trunk
Device Protocol*	SIP
Trunk Service Type*	None(Default)

Next

3. Configure **Trunk Type**: SIP Trunk.
4. Configure **Device Protocol**: SIP.
5. Configure **Trunk Service Type**: None (Default).
6. Click **Next**.

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

Cisco Unified CM Administration  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration Go

administrator | Search Documentation | About | Logout

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

**Trunk Configuration** Related Links: Back To Find/List ▾ Go

Save Delete Reset Add New

**Status**  
Status: Ready

**SIP Trunk Status**

**Service Status:** Full Service  
**Duration:** Time In Full Service: 7 days 7 hours 41 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

Media Termination Point Required

Retry Video Call as Audio

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

<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* <input type="text" value="When using both sRTP and TLS"/>
Route Class Signaling Enabled* <input type="text" value="Default"/>
Use Trusted Relay Point* <input type="text" value="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	<input type="text" value="&lt; None &gt;"/>
------------------------------	---

---

**MLPP and Confidential Access Level Information**

MLPP Domain	<input type="text" value="&lt; None &gt;"/>
Confidential Access Mode	<input type="text" value="&lt; None &gt;"/>
Confidential Access Level	<input type="text" value="&lt; None &gt;"/>

---

**Call Routing Information**

<input checked="" type="checkbox"/> Remote-Party-Id	
<input checked="" type="checkbox"/> Asserted-Identity	
Asserted-Type*	<input type="text" value="Default"/>
SIP Privacy*	<input type="text" value="Default"/>

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

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

**Inbound Calls**

Significant Digits\*

Connected Line ID Presentation\*

Connected Name Presentation\*

Calling Search Space

AAR Calling Search Space

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	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input type="text" value="&lt; None &gt;"/>	<input checked="" type="checkbox"/>

---

**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	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input type="text" value="&lt; None &gt;"/>	<input checked="" type="checkbox"/>

11. Select the Redirecting Diversion Header Delivery – Inbound check box.

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

**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

Redirecting Party Transformation CSS

Use Device Pool Redirecting Party Transformation CSS

---

**Caller Information**

Caller ID DN

Caller Name

12. Select the Redirecting Diversion Header Delivery – Outbound check box.

13. Configure the SIP Information.

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

SIP Information				
<b>Destination</b>				
<input type="checkbox"/> Destination Address is an SRV				
	<b>Destination Address</b>	<b>Destination Address IPv6</b>	<b>Destination Port</b>	<b>Status</b>
1*	10.64.1.72		5060	up
MTP Preferred Originating Codec*	711ulaw			
BLF Presence Group*	Standard Presence group			
SIP Trunk Security Profile*	Non Secure SIP Trunk Profile			
Rerouting Calling Search Space	< None >			
Out-Of-Dialog Refer Calling Search Space	< None >			
SUBSCRIBE Calling Search Space	< None >			
SIP Profile*	Standard SIP Profile_Crestron			<a href="#">View Details</a>
DTMF Signaling Method*	No Preference			
<b>Normalization Script</b>				
Normalization Script	< None >			
<input type="checkbox"/> Enable Trace				
	<b>Parameter Name</b>	<b>Parameter Value</b>		
1			<input type="button" value="+"/>	<input type="button" value="-"/>
<b>Recording Information</b>				
<input checked="" type="radio"/> None				
<input type="radio"/> This trunk connects to a recording-enabled gateway				
<input type="radio"/> This trunk connects to other clusters with recording-enabled gateways				

- a. Enter the **Destination Address** and **Destination Port** of the PSTN Gateway.
- b. Select the default **Non Secure SIP Trunk Profile** as the SIP Trunk Security Profile.
- c. Select the configured **Standard SIP Profile\_Crestron** SIP Profile.

14. 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.

*Cisco UCM: Trunk Configuration: Trunk to Voice Mail System: Unity Connection (1/5)*

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 menu shows 'System', 'Call Routing', 'Media Resources', 'Advanced Features', 'Device', 'Application', 'User Management', and 'Bulk Administration'. The main content area is titled 'Trunk Configuration' and features a 'Related Links' section with a 'Back To Find/List' button. A toolbar contains 'Save', 'Delete', 'Reset', and 'Add New' icons. The 'Device Information' section is expanded, showing the following configuration details:

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 configuration section, two checkboxes are checked: 'Media Termination Point Required' and 'Retry Video Call as Audio'.

Cisco UCM: Trunk Configuration: Trunk to Voice Mail System: Unity Connection (2/5)

<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 Configuration: Trunk to Voice Mail System: Unity Connection (3/5)

**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"/>

**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 < None >

Use Device Pool Connected Party Transformation CSS

**Outbound Calls**

Called Party Transformation CSS < None >

Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS < None >

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

Redirecting Diversion Header Delivery - Outbound

Cisco UCM: Trunk Configuration: Trunk to Voice Mail System: Unity Connection (5/5)

SIP Information			
<b>Destination</b>			
<input type="checkbox"/> Destination Address is an SRV			
	<b>Destination Address</b>	<b>Destination Address IPv6</b>	<b>Destination Port</b>
1 *	10.80.10.5		5060
MTP Preferred Originating Codec*	711ulaw		
BLF Presence Group*	Standard Presence group		
SIP Trunk Security Profile*	Non Secure SIP Trunk Profile		
Rerouting Calling Search Space	< None >		
Out-Of-Dialog Refer Calling Search Space	< None >		
SUBSCRIBE Calling Search Space	< None >		
SIP Profile*	Standard SIP Profile		<a href="#">View Details</a>
DTMF Signaling Method*	RFC 2833		
<b>Normalization Script</b>			
Normalization Script	< None >		
<input type="checkbox"/> Enable Trace			
	<b>Parameter Name</b>	<b>Parameter Value</b>	
1			<input type="button" value="+"/> <input type="button" value="-"/>
<b>Recording Information</b>			
<input checked="" type="radio"/> None			
<input type="radio"/> This trunk connects to a recording-enabled gateway			
<input type="radio"/> This trunk connects to other clusters with recording-enabled gateways			
<b>Geolocation Configuration</b>			
Geolocation	< None >		
Geolocation Filter	< None >		
<input type="checkbox"/> Send Geolocation Information			
<input type="button" value="Save"/> <input type="button" value="Delete"/> <input type="button" value="Reset"/> <input type="button" value="Add New"/>			

## Configure Route Patterns

Route patterns were configured for the following:

- To route calls from the Cisco UCM to the Cisco UBE towards PSTN GW
- To restrict caller id on outgoing calls.

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.@** and **\+\*** were configured to enable outbound dialing from Cisco UCM to PSTN using the access code as **9** and using the **+**.

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

The screenshot shows the 'Route Pattern Configuration' window. At the top, there are 'Related Links: Back To Find/List' and 'Go'. Below that are icons for 'Save', 'Delete', 'Copy', and 'Add New'. The 'Status' section shows 'Status: Ready'. The 'Pattern Definition' section contains the following fields:

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

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

<b>Calling Party Transformations</b>		
<input checked="" 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*	Default	
Calling Name Presentation*	Default	
Calling Party Number Type*	Cisco CallManager	
Calling Party Numbering Plan*	Cisco CallManager	
<b>Connected Party Transformations</b>		
Connected Line ID Presentation*	Default	
Connected Name Presentation*	Default	
<b>Called Party Transformations</b>		
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	
<b>ISDN Network-Specific Facilities Information Element</b>		
Network Service Protocol	-- Not Selected --	
Carrier Identification Code	<input type="text"/>	
Network Service	Service Parameter Name	Service Param
-- Not Selected --	< Not Exist >	<input type="text"/>

Cisco UCM: Pattern Configuration: Outbound Dialing Using a + (1/2)

**Route Pattern Configuration** Related Links: [Back To Find/List](#)

---

**Status**

Update successful

---

**Pattern Definition**

Route Pattern*	<input type="text" value="\+*"/>
Route Partition	< None >
Description	<input type="text" value="Dial out using a +"/>
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*	PSTN <a href="#">(Edit)</a>
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern <input type="text" value="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*	<input type="text" value="0"/>
<input type="checkbox"/> Require Client Matter Code	

Cisco UCM: Pattern Configuration: Outbound Dialing Using a + (2/2)

<b>Calling Party Transformations</b>		
<input checked="" 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*	Default	
Calling Name Presentation*	Default	
Calling Party Number Type*	Cisco CallManager	
Calling Party Numbering Plan*	Cisco CallManager	
<b>Connected Party Transformations</b>		
Connected Line ID Presentation*	Default	
Connected Name Presentation*	Default	
<b>Called Party Transformations</b>		
Discard Digits	< None >	
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	
<b>ISDN Network-Specific Facilities Information Element</b>		
Network Service Protocol	-- Not Selected --	
Carrier Identification Code	<input type="text"/>	
Network Service	Service Parameter Name	Service Parameter Va
-- Not Selected --	< Not Exist >	<input type="text"/>
Save Delete Copy Add New		

Similarly, the route pattern of \*6.@ was configured to restrict caller ID on outbound calls.

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

The screenshot shows the 'Route Pattern Configuration' page in Cisco UCM. The interface includes a top navigation bar with 'Related Links: Back To Find/List' and a 'Go' button. Below this is a toolbar with 'Save', 'Delete', 'Copy', and 'Add New' options. The main content area is divided into two sections: 'Status' and 'Pattern Definition'. The 'Status' section shows 'Status: Ready'. The 'Pattern Definition' section contains the following fields and options:

- Route Pattern\***: \*6.@
- Route Partition**: < None >
- Description**: (empty)
- Numbering Plan\***: NANP
- Route Filter**: < None >
- MLPP Precedence\***: Default
- Apply Call Blocking Percentage**
- Resource Priority Namespace Network Domain**: < None >
- Route Class\***: Default
- Gateway/Route List\***: PSTN (with an [\(Edit\)](#) link)
- Route Option**:
  - Route this pattern**
  - Block this pattern** (with a dropdown menu set to 'No Error')
- Call Classification\***: OffNet
- External Call Control Profile**: < None >
- Allow Device Override**  **Provide Outside Dial Tone**  **Allow Overlap Sending**  **Urgent Priority**
- Require Forced Authorization Code**
- Authorization Level\***: 0
- Require Client Matter Code**

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

<b>Calling Party Transformations</b>		
<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	▼
<b>Connected Party Transformations</b>		
Connected Line ID Presentation*	Default	▼
Connected Name Presentation*	Default	▼
<b>Called Party Transformations</b>		
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	▼
<b>ISDN Network-Specific Facilities Information Element</b>		
Network Service Protocol	-- Not Selected -- ▼	
Carrier Identification Code	<input type="text"/>	
Network Service	Service Parameter Name	Service Param
-- Not Selected -- ▼	< Not Exist >	<input type="text"/>

A route pattern of 7000 was configured to route the voice mail pilot number (7000) to the Unity Connection server.

*Cisco UCM: Route Pattern Configuration: Voice Mail Pilot Number (1/2)*

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", and the subtitle "For Cisco Unified Communications Solutions". The user is logged in as "administrator". A menu bar contains options like System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The main heading is "Route Pattern Configuration", with a "Related Links" section containing "Back To Find/List" and "Go". Below the heading are icons for Save, Delete, Copy, and Add New. The "Status" section shows "Status: Ready". The "Pattern Definition" section contains the following fields:

Route Pattern*	7000
Route Partition	< None >
Description	Voice mail to unity Connection
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 <a href="#">(Edit)</a>
Route Option	<input checked="" type="radio"/> Route this pattern

## Cisco UCM: Route Pattern Configuration: Voice Mail Pilot Number (2/2)

<input type="radio"/> Block this pattern <span>No Error</span>		
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*	0	
<input type="checkbox"/> Require Client Matter Code		
<b>Calling Party Transformations</b>		
<input type="checkbox"/> 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	
<b>Connected Party Transformations</b>		
Connected Line ID Presentation*	Default	
Connected Name Presentation*	Default	
<b>Called Party Transformations</b>		
Discard Digits	< None >	
Called Party Transform Mask		
Prefix Digits (Outgoing Calls)		
Called Party Number Type*	Cisco CallManager	
Called Party Numbering Plan*	Cisco CallManager	
<b>ISDN Network-Specific Facilities Information Element</b>		
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"/>		

## Voice Mail Configuration

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

### *Configure Voice Mail Pilot and Voice Mail Profile on Cisco UCM*

Use the following procedure to configure voice mail pilot and voice mail profile on Cisco UCM.

1. Navigate to **Advanced Features > Voice Mail > Voice Mail Pilot**.

### Cisco UCM: Voice Mail Pilot Configuration: Add Voice Mail Pilot Number

The screenshot shows the Cisco Unified CM Administration interface for Voice Mail Pilot Configuration. The page title is "Voice Mail Pilot Configuration" and it includes a "Related Links" section with "Back To Find/List" and "Go". The "Status" section shows "Status: Ready". The "Voice Mail Pilot Information" section contains the following fields:

- Voice Mail Pilot Number: 7000
- Calling Search Space: < None >
- Description: Unity Connection VM
- Make this the default Voice Mail Pilot for the system

At the bottom, there are buttons for "Save", "Delete", and "Add New".

2. Add a new pilot number. 7000 was configured for this example.
3. Select the **Make this the default Voice Mail Pilot for the system** check box.
4. Configure a **Voice Mail Profile** with this pilot number . VM\_profile\_clus20 was configured for this example.

### Cisco UCM: Voice Mail Profile Configuration

The screenshot shows the Cisco Unified CM Administration interface for Voice Mail Profile Configuration. The page title is "Voice Mail Profile Configuration" and it includes a "Related Links" section with "Back To Find/List" and "Go". The "Status" section shows "Status: Ready". The "Voice Mail Profile Information" section contains the following fields:

- Voice Mail Profile: VM\_profile\_clus20 (used by 1059 devices)
- Voice Mail Profile Name\*: VM\_profile\_clus20
- Description:
- Voice Mail Pilot\*\*: 7000/< None >
- Voice Mail Box Mask:
- Make this the default Voice Mail Profile for the System

At the bottom, there are buttons for "Save", "Delete", "Copy", "Reset", "Apply Config", and "Add New".

5. Check the **Make this the default Voice Mail Pilot for the system** check box.

## Configuration on Unity Connection - Add New Phone System

To configure a new phone system, after logging into Unity Connection, perform the following procedure.

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

### Cisco Unity Connection: Telephony Integrations: Phone System

The screenshot displays the Cisco Unity Connection Administration web interface. The left-hand navigation pane shows the 'Cisco Unity Connection' tree with 'Telephony Integrations' expanded to 'Phone System'. The main content area is titled 'Phone System' and contains the following configuration options:

- Phone System Name\***: A text input field containing 'Crestron'.
- 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
  - 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:
  - Guard Time:  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:
    - Ending Time:

At the bottom of the configuration area are buttons for **Save**, **Delete**, **Previous**, and **Next**.

3. Configure the **Phone System Name**: *Crestron* was used in this example.
4. Click **Save**.

On the **Phone System Basics** page, in the **Related Links** drop-down box, select **Add Port Group**, and select **Go**.

*Cisco Unity Connection: Telephony Integrations: Port Group*

The screenshot shows the Cisco Unity Connection Administration interface. The left sidebar contains a navigation menu 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 'Telephony Integrations' section is expanded, showing 'Phone System' and 'Port Group'. The main content area is titled 'New Port Group' and includes a 'Save' button, a 'Phone System' dropdown menu set to 'Crestron', and 'Create From' radio buttons for 'Port Group Type' (selected) and 'SIP'. Below this is the 'Port Group Description' section with fields for 'Display Name\*' (Crestron-1), 'Authenticate with SIP Server' (checkbox), 'Authentication Username', 'Authentication Password', 'Contact Line Name', 'SIP Security Profile' (5060), and 'SIP Transport Protocol' (TCP). The 'Primary Server Settings' section includes 'IPv4 Address or Host Name' (10.80.10.2), 'IPv6 Address or Host Name', and 'Port' (5060). A 'Save' button is at the bottom. A note at the bottom states: 'Fields marked with an asterisk (\*) are required.'

1. **Phone System:** Select the one created earlier (Crestron).
2. **Create From:** Select **Port Group Type** and select **SIP** from the drop-down menu.
3. **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.
4. Click **Save**.

On the “Port Group Basics” page, in the **Related Links** drop-down menu, select **Add Ports** and select **Go**.

*Cisco Unity Connection: Telephony Integrations: Port Group Added, Related Links to Add Port*

**Cisco Unity Connection Administration**  
For Cisco Unified Communications Solutions

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

**Port Group Basics (Crestron-1)**  
Search Port Groups | Port Group Basics (Crestron-1)  
Related Links: Add Ports | Go

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.
- ℹ Created Port Group(s)

**Port Group**

Display Name\* | Crestron-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

Fields marked with an asterisk (\*) are required.

On the **New Port** page, configure the settings and select **Save**.

*Cisco Unity Connection: Telephony Integrations: Port Group: Add New Port*

**Cisco Unity Connection Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unity Connection Administration | Go  
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

**New Port** Search Ports | New Port  
Related Links: Check Telephony Configuration | Go

Port Reset Help

Save

**New Phone System Port**

Enabled

Number of Ports: 1

Phone System: Crestron

Port Group: Crestron-1

Server: clus20unity.skypelabsj.local

**Port Behavior**

Answer Calls

Perform Message Notification

Send MWI Requests (may also be disabled by the port group)

Allow TRAP Connections

Save

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

1. Navigate to **Telephony Integrations > Phone System > CUCM11.0**.

*Cisco Unity Connection: Telephony Integrations: Phone System: Edit AXL servers*

**Cisco Unity Connection Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unity Connection Administration | Go  
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

**Edit AXL Servers** Search Phone Systems | Phone System Basics (Crestron) | Edit AXL Servers  
Related Links: Check Telephony Configuration | Go

Phone System Edit Refresh Help

Save

**AXL Servers**

Delete Selected Add New

	Order	IP Address	Port	
<input type="checkbox"/>	1	10.80.10.3	5060	Test
<input type="checkbox"/>	0	10.80.10.2	5060	Test

Delete Selected Add New

**AXL Server Settings**

Username: administrator

Password: .....

Cisco Unified Communications Manager Version: 5.0 or Greater (SSL)

Save

2. On the **Phone System Basics**, click **Edit > Cisco Unified Communications Manager AXL Servers**.
3. Click **Add New**, or in the second row, configure the Cisco UCM Subscriber IP and port. *10.80.10.3* and *5060* were used in this example.
4. Click **Save**.

### Configure a Voice Mail User

To configure a new user that would have a voice mail box, after logging into Unity Connection, follow this procedure.

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

#### Cisco Unity Connection: Users: Add User

The screenshot shows the Cisco Unity Connection Administration interface. The left sidebar is expanded to show the 'Users' menu. 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\*: Mercury2600
  - First Name: [Empty]
  - Last Name: [Empty]
  - Display Name: [Empty]
  - SMTP Address: [Empty] @clus20unity.lab.tekvizion.com
- Mailbox Store:** Unity Messaging Database -1
- Phone:**
  - Extension\*: 2600
  - Cross-Server Transfer Extension or URI: [Empty]
  - Outgoing Fax Number: [Empty]
  - Corporate Email Address: [Empty]

A 'Save' button is located at the bottom of the form.

3. Configure a **Based on Template** from the drop-down menu. *Voice Mailusertemplate* was used in this example.
4. Configure an **Alias**: *Mercury2600* 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**. Select the Phone System configured earlier from drop-down list. *Crestron* was used in this example.

Cisco Unity Connection: Users: Assign Phone System to User

The screenshot shows the Cisco Unity Connection Administration interface. The main content area is titled "Edit User Basics (Mercury2600)". The interface includes a navigation menu on the left and a main form area. The form contains several sections:

- Name:** Fields for Alias\* (Mercury2600), First Name, Last Name, Display Name (Mercury2600), SMTP Address (mercury2600@clus20unity.lab.tekvizion.com), Initials, Title, and Employee ID.
- LDAP Integration Status:** Radio buttons for "Integrate with LDAP Directory" and "Do Not Integrate with LDAP Directory" (selected).
- Phone:** Fields for Extension\* (2600), Cross-Server Transfer Extension or URI, Outgoing Fax Number, Outgoing Fax Server (--- Not Selected ---), Partition (clus20unity Partition), Search Scope (clus20unity Search Space), Phone System (Crestron), Class of Service (Voice Mail User COS), and Active Schedule (Weekdays).

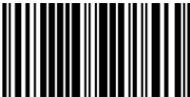
Buttons for "Save", "Delete", "Previous", and "Next" are visible at the top of the form area. A "View" button is located at the bottom right of the form.

8. Click **Save**.

This page is intentionally left blank.

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**Configuration Guide – DOC. 7993A**  
**(2048879)**  
**05.17**  
Specifications subject to  
change without notice.