



DSP-1282 & DSP-1283
Crestron Avia™ DSP with Cisco®
Unified Communications Manager 11.0

Configuration Guide
Crestron Electronics, Inc.

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DSP-1282 & DSP-1283: SIP Endpoint with Cisco® Unified Communications Manager 11.0

Introduction

This configuration guide describes the procedures required to configure Crestron Avia™ Digital Signal Processor (DSP) devices. The devices operate on the Cisco® Unified Communications Manager (UCM) as basic Session Initiation Protocol (SIP) endpoints.

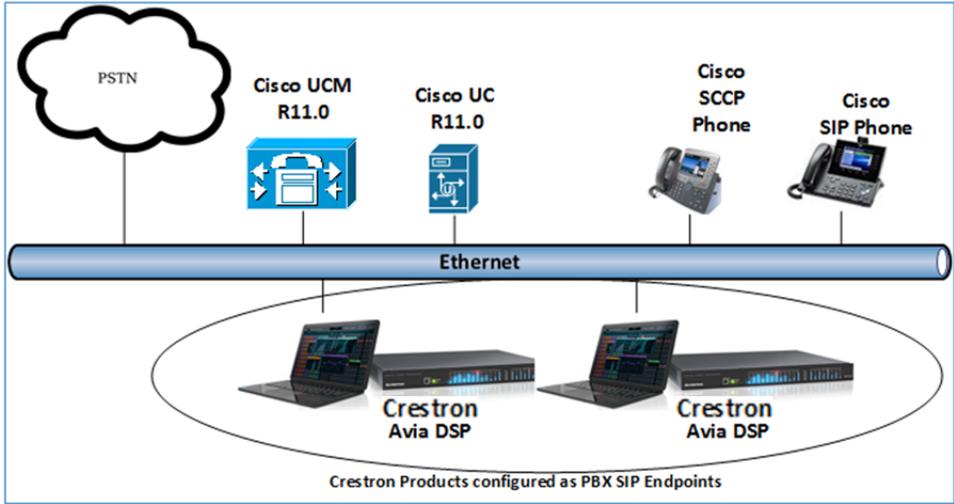
Audience

The intended audience includes those attempting to configure and use Crestron Avia DSP devices as SIP endpoints registered to the Cisco Unified Communications (Cisco UCM).

Topology

The diagram below shows the network topology for integration of a Crestron Avia DSP endpoint with the Cisco UCM.

SIP Endpoint Integration with - Reference Network



The lab network consists of the following components:

- Cisco UCM cluster for voice features
- Cisco Signaling Connection Control Part (SCCP) and SIP phones
- Cisco Unity Connection as the voice mail system
- Crestron DSP as SIP endpoints

Software Requirements

- Cisco Unified Communications Manager v11.0.1.20000-2
- Cisco Unity Connection v 11.0.1.20000-2
- Crestron Avia DSP devices v1.00.092

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 Avia DSP devices (2):
 - Microphones for the DSP (2)
 - Speakers for the DSP (2)
 - Amplifiers for the DSP (2)

Product Description

The Crestron Avia DSP products (DSP-1282 and DSP-1283, specifically) consist of a family of programmable digital audio signal processors intended for the commercial sound market. Each version provides 12 analog mic/line inputs and eight analog line outputs. The devices include a Local Area Network (LAN) connection and a Universal Serial Bus (USB) connection for programming and control. The programmable signal flow is a fixed topology with user-configurable input and output processing chains using a library of preset signal-specific DSP blocks.

Use the Crestron Avia tool to:

- Discover the device on the network
- Configure the SIP parameters
- Configure the mixers to allow 2-way communication on a SIP call

Save the audio configuration along with the SIP configuration as a project file. The project file can be loaded onto all of the DSPs that receive similar settings on a given project. Minor modifications may be necessary.

Use the Crestron Toolbox™ software to discover and control all Crestron devices on the network.

During the integration test, Crestron Toolbox can:

- Discover devices on the network
- Console connect to the devices
- Configure the Ethernet settings
- Upgrade firmware

Summary

This document describes how to configure the Crestron Avia DSP devices on the Cisco UCM as basic SIP endpoints. It also provides information on how to register devices to the Cisco UCM with digest authentication.

Supported features include:

- Registration with digest authentication
- Basic calls with G711u and G711a codecs
- Dual-Tone Multi-Frequency (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
- Voice mail access and interaction
- DND (Do Not Disturb)

Unsupported features include:

- Caller ID presentation
- Call hold and resume
- Call forwarding on the device (forwarding can be configured on the Private Branch Exchange (PBX) for the Domain Name (DN) assigned to the endpoint)
- Call waiting
- Conference
- Attended call transfer
- Early attended call transfer
- Blind call transfer
- Shared line (configuration of shared line on device)
- Initiating call park
- Message Waiting Indicator (MWI)

Known issues and limitations include:

- No support for caller ID on the Crestron Avia DSP.
- No support for MWI on the Crestron Avia DSP.
- No support for Music on Hold (MoH) in scenarios where the Cisco UCM sends an **INVITE** with a **sendonly** Session Description Protocol (SDP) to initiate a call hold.
- The DSP fails to play a reorder tone when a call from the DSP to a PBX extension times out after the called party does not answer.
- When in an alert state, the DSP fails to respond when added to a conference.

Crestron Avia DSP Configuration

This section provides the following details:

- How to set up connections to the amplifier and speaker
- How to access the DSP on the network (once powered)
- How to configure the DSP for registration and integration with the Cisco UCM

Connections

Make the following connections:

- Connect microphone to DSP MIC/LINE INPUTS port 1
- Connect DSP LINE OUTPUTS port 1 to "Audio In" on amplifier
- Connect "Audio Out" of amplifier to speaker
- Connect LAN port to network
- Connect VOIP port to network

Device Discovery/Access

Use the Crestron Toolbox and the Crestron Avia tool to discover and access the connected LAN and/or VOIP ports) DSP devices.

Use the Help menu to assist when performing the discovery and configuration procedure.

Set Up SIP Interface and Routes

The DSP units have separate network interfaces for Voice over Internet Protocol (VoIP) and LAN on the rear panel. Configure either one for SIP calling. The default configuration binds SIP calling to the LAN interface. An optional console command binds the SIP interface to the VoIP connector. Configure all VoIP connections on a separate Virtual Local Area Network (VLAN) or subnet. VoIP connections cannot be on the same subnet as the LAN connection.

Ethernet

Use the **Ethernet** command to turn the VoIP port on/off.

```
DSP-1281>Ethernet ?
ETHERNET [<device_num> ON | OFF [/now]]
Device_num - 0 n
ON - enables VoI
OFF - disables VoIP
/now - take effect without a reboot
No parameter - displays the current setting
```

The VoIP port is off by default. The LAN port is not selectable.

```
<device_num> = 0 selects the LAN port
<device_num> = 1 selects the VoIP port
```

SIP Interface

Use the **sipinterface** command to bind all SIP activity, data, and traffic to the selected port. If a VLAN or exclusive VoIP network is available, bind to the VoIP port (recommended).

```
DSP-1281>sipinterface ?
Get or Set SIP Interface
SIPINTERFACE [LAN | VOIP]
LAN - normal LAN port
VOIP - VOIP port
No Parameter - Displays current setting
```

Set Up Routes

If the configured VoIP port is the SIP interface, add a static route to ensure that all SIP routing is via the VoIP port.

The following console commands (**routeadd**, **routedel**, **routeprint**, and **routeprint**) support the static IP routing configuration:

```
DSP-1282>routeadd ?
ROUTEADD <destination> <netmask> <gateway> [/FORCE]
destination - destination IP address in dot decimal notation
netmask - netmask in dot decimal notation
gateway - gateway in dot decimal notation
/FORCE - force to add/delete even if failed to persist to NVRAM
```

```
DSP-1282>routedel ?
ROUTEDELETE <destination> <netmask> <gateway> [/FORCE]} | </ALL>
destination - destination IP address in dot decimal notation
netmask - netmask in dot decimal notation
gateway - gateway in dot decimal notation
/FORCE - force to add/delete even if failed to persist to NVRAM
/ALL - delete all routes from NVRAM
```

```
DSP-1282>routeprint ?
ROUTEPRINT - shows current routes
```

```
DSP-1282>routeprint ?
ROUTETRACE <IPaddress>
IPaddress - IP address in dot decimal notation
```

Device Configuration

The basic setup for a phone call requires:

- An analog input (such as from a microphone) routed out through the phone line
- Audio coming in from the phone line routed to an analog output (such as to an amplifier or speaker)

Configure the DSP Device

Use the Crestron Avia tool to select and configure the DSP device.

Input Configuration

To configure the analog input:

1. Click **Signal**.

Crestron Avia tool: Audio Input Configuration (1/4)



2. Under **Analog In 1** (first row), double click **Gain**. In the new window set the following:
 - a. Click **Mute** to **Off**.
 - b. Select **33** for the **Analog Gain**.
 - c. If a condenser microphone is being used, click **+48V** (phantom power) to **On**.

Crestron Avia Tool: Audio Input Configuration (2/4)



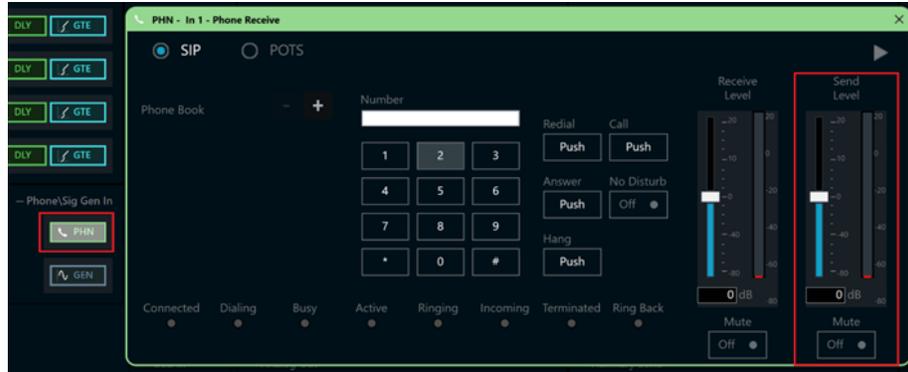
3. Under **Analog In 1** (first row), click **Ref/Phone Out** (right-most column) and enter **0** as the decibel value.

Crestron Avia Tool: Audio Input Configuration (3/4)



4. Under **Phone\Sig Gen In**, click **PHN**. In the new window set the following:
 - a. Move the **Send Level** slider to **0 db**.
 - b. Click **Mute** to **Off**.

Crestron Avia Tool: Audio Input Configuration (4/4)



Output Configuration

To configure the analog output:

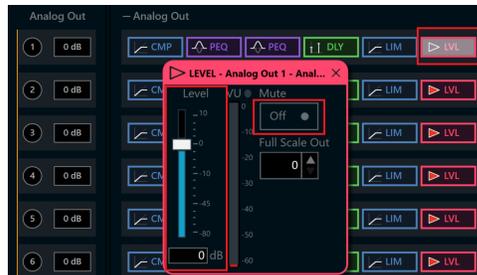
1. Under **Phone In 1** (first row), click **Analog Out** (left-most column) and enter **0** as the decibel value.

Crestron Avia Tool: Audio Output Configuration (1/3)



2. Under **Analog Out 1**, double click **LVL**. In the new window set the following:
 - a. Move the **Level** slider to **0 db**.
 - b. Click **Mute** to **Off**.

Crestron Avia Tool: Audio Output Configuration (2/3)



3. Under **Phone\Sig Gen In**, click **PHN**. In the new window set the following:
 - a. Move the **Receive Level** slider to **0 db**.
 - b. Click **Mute** to **Off**.

Crestron Avia Tool: Audio Output Configuration (3/3)



Configure the SIP Parameters

From the open **PHN - In 1 - Phone Receive** window, select and configure the SIP parameters.

1. With **SIP** selected, click the chevron at the right top corner to expand the window.
Crestron Avia Tool: Phone Dialer, SIP Parameters Configuration



2. Enter the extension configured on Cisco UCM for the **Local Extension** for this device. This example uses **2500**.
3. Enter the Cisco UCM publisher IP for the **SIP Server IP Address**. This example uses **10.80.25.2**.
4. Enter the SIP server port (**5060**) for the **Port**.
5. Enter the same end user name configured for the Cisco UCM for the **SIP Server User Name**. This example uses **crestron_avia**.
6. Enter the same password as configured for the Cisco UCM end user digest credentials for the **SIP Server Password**.

Cisco UCM Configuration

This section describes the Cisco UCM configuration necessary to integrate Crestron devices as SIP endpoints.

NOTE: Confirm that the general installation and basic Cisco UCM configuration have been administered.

Configure the User

To configure the end user:

1. Click **User Management > End User**.
2. Click **Add New**.

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

Status
Status: Ready

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 [View Details](#)

3. Enter a unique end user identification name for the **User ID**. This example uses **DSP1** and **DSP2** **crestron_avia** and **crestron_avia2** for the two DSP devices.
4. Enter a **Password**. This example uses **123456**, which is the same password used on the device against the SIP server password.
5. Enter the same password for **Confirm Password**.
6. Enter the end user's last name for the **Last Name**. This example uses **AviaDSP**.
7. Enter a string of alphanumeric characters for the **Digest Credentials**.
8. Enter the same string for **Confirm Digest Credentials**.
9. Click **Save**.

Cisco UCM: End Users Configured for all DSP Devices

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 subtitle "For Cisco Unified Communications Solutions". The user is logged in as "administrator". The main menu includes System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The "Find and List Users" section is active, showing 2 records found. A search filter is applied to the "User ID" field, set to "begins with cres". The table below lists two users: "crestron_avia" and "crestron_avia2".

Status
2 records found

User (1 - 2 of 2) Rows per Page 50

Find User where User ID begins with cres Find Clear Filter

<input type="checkbox"/>	User ID ^	Meeting Number	First Name	Last Name	Department	Directory URI	User Status
<input type="checkbox"/>	crestron_avia		Crestron	AviaDSP			Enabled Local User
<input type="checkbox"/>	crestron_avia2		DSP128	Avia2			Enabled Local User

Add New Select All Clear All Delete Selected

Configure a SIP Profile

This example configures a new SIP Profile: **Standard SIP Profile_Test**.

To add a new SIP Profile:

1. Click **Device > Device Settings > SIP Profile**.
2. Click **Add New**.

Cisco UCM: SIP Profile Configuration (1/4)

The screenshot displays the Cisco Unified CM Administration interface for SIP Profile Configuration. The page title is "SIP Profile Configuration" and it includes a navigation menu with options like System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, and Bulk Administration. The "Device" menu item is highlighted. Below the navigation is a toolbar with icons for Save, Delete, Copy, Reset, Apply Config, and Add New. The main configuration area is divided into two sections: "SIP Profile Information" and "SDP Information".

SIP Profile Information

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

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 Performance

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

Require SDP Inactive Exchange for Mid-Call Media Change
 Allow RR/RS bandwidth modifier (RFC 3556)

3. Enter **Standard SIP Profile_Test** for the **Name** (for this example).

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><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*	<input type="text" value="Never"/>
Resource Priority Namespace List	<input type="text" value="< None >"/>
SIP Rel1XX Options*	<input type="text" value="Disabled"/>
Video Call Traffic Class*	<input type="text" value="Mixed"/>
Calling Line Identification Presentation*	<input type="text" value="Default"/>
Session Refresh Method*	<input type="text" value="Invite"/>
Early Offer support for voice and video calls*	<input type="text" value="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)*	<input type="text" value="60"/>
Ping Interval for Out-of-service Trunks (seconds)*	<input type="text" value="120"/>
Ping Retry Timer (milliseconds)*	<input type="text" value="500"/>
Ping Retry Count*	<input type="text" value="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	

4. Select **Best Effort (no MTP inserted)** for **Early Offer support for voice and video calls**.
5. Leave all other fields at the default values.
6. Click **Save**.
7. Click **Apply Config**.

Configure Phone Security Profile

To configure the Phone Security Profile:

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

Cisco UCM: Phone Security Profile

The screenshot displays the Cisco Unified CM Administration interface for configuring a Phone Security Profile. The page title is "Phone Security Profile Configuration". The navigation menu includes System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The toolbar contains icons for Save, Delete, Copy, Reset, Apply Config, and Add New. The configuration fields are as follows:

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
<input checked="" type="checkbox"/> Enable Digest Authentication	

Parameters used in Phone:

SIP Phone Port*	5060
-----------------	------

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

3. Enter **Crestron** for the **Name** (for this example).
4. Select **TCP+UDP** for the **Transport Type**.
5. Check **Enable Digest Authentication**.
6. Click **Save**.

Configure the Crestron Device as a Third-party SIP Device

To configure the DSP device as a third-party SIP device:

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

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

The screenshot shows the 'Phone Configuration' page in Cisco UCM. The 'Phone Type' section is highlighted with a red box, showing 'Product Type: Third-party SIP Device (Basic)' and 'Device Protocol: SIP'. The 'Device Information' section is also highlighted with a red box, showing a warning 'Device is not trusted' and various configuration fields. The fields are as follows:

Field	Value
MAC Address*	00107F05227A
Description	SEP00107F05227A
Device Pool*	Default
Common Device Configuration	< None >
Phone Button Template*	Third-party SIP Device (Basic)
Common Phone Profile*	Standard Common Phone Profile
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*	crestron_avia
Use Trusted Relay Point*	Default
Always Use Prime Line*	Default

3. In the **Phone Type** section, select **Third-party SIP Device (Basic)** for the **Product Type**.
4. Click **Next**.
5. Enter the MAC address of the DSP for the **MAC Address**.
6. Select **Default** for the **Device Pool**.
7. Select **Third-party SIP Device (Basic)** for the **Phone Button Template**.
8. Click **User** for the **Owner**.
9. Select the End User configured earlier for the **Owner User ID**. This example selects **crestron_avia** for the first Crestron Avia DSP device and **crestron_avia2** for the second Crestron Avia DSP device.

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

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		
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	▼ View Details
Digest User	crestron_avia	▼
<input type="checkbox"/> Media Termination Point Required		
<input type="checkbox"/> Unattended Port		

10. Select **Crestron** (configured earlier for this example) for the **Device Security Profile**.
11. Select **Standard SIP Profile_Test** (configured earlier for this example) for the **SIP Profile**.
12. Select **crestron_avia** for the first Crestron Avia DSP device and **crestron_avia2** for the second (configured earlier for this example) for the **Digest User**.
13. Click **Save**.
14. Add a DN to this phone. This example configures DN 2500 for one of the Crestron Avia DSP devices and DN 2501 for the other.

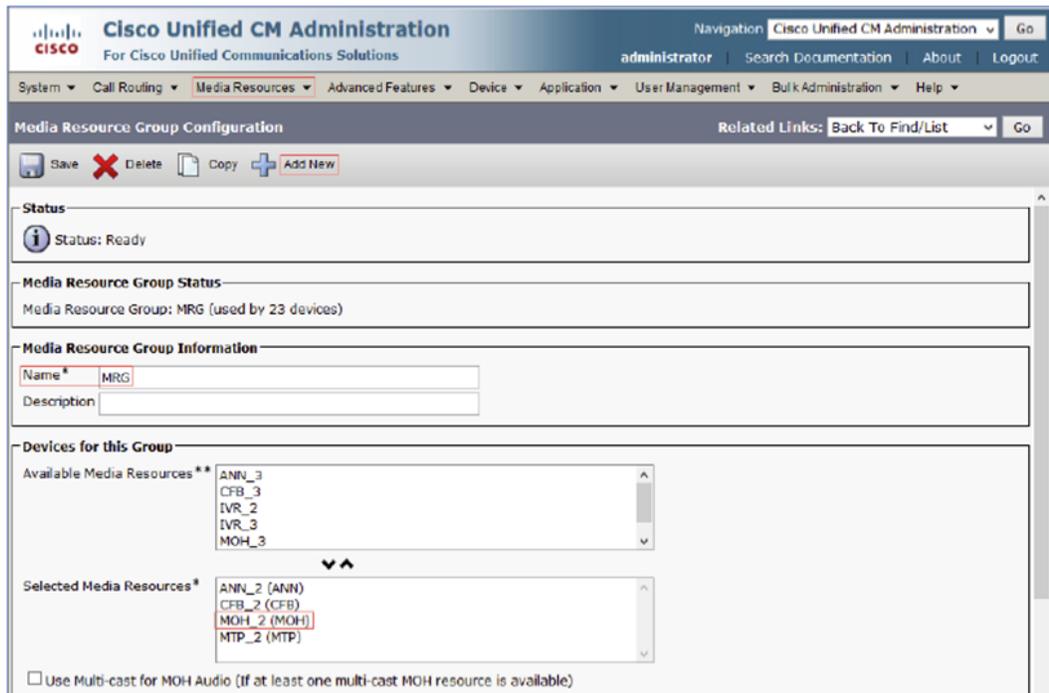
Configure Media Resource Group

A Media Resource Group (MRG) includes Music on Hold servers, conference bridges, and media termination points that may test the Cisco UCM or service provider features.

To configure a Media Resource Group (for this example):

1. Click **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 status is "Ready". The Media Resource Group is named "MRG" and is used by 23 devices. The "Media Resource Group Information" section shows the name "MRG" and a description field. The "Devices for this Group" section contains two lists: "Available Media Resources" and "Selected Media Resources". The "Available Media Resources" list includes ANN_3, CFB_3, IVR_2, IVR_3, and MOH_3. The "Selected Media Resources" list includes ANN_2 (ANN), CFB_2 (CFB), MOH_2 (MOH), and MTP_2 (MTP). A checkbox for "Use Multi-cast for MOH Audio" is present at the bottom.

3. Enter **MRG** for the **Name** (for this example).
4. Transfer media resources between the two lists. Resources (added earlier) are available for use with the Cisco UCM.

Configure the Media Resource Group List

To configure a Media Resource Group List (for this example):

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

Cisco UCM: Media Resource Group List Configuration

The screenshot shows the Cisco Unified CM Administration web interface. The page title is "Media Resource Group List Configuration". The navigation menu includes "System", "Call Routing", "Media Resources", "Advanced Features", "Device", "Application", "User Management", "Bulk Administration", and "Help". The user is logged in as "administrator". The page has a "Save", "Delete", "Copy", and "Add New" toolbar. 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 with the value "MRGL". The "Media Resource Groups for this List" section has two lists: "Available Media Resource Groups" and "Selected Media Resource Groups". The "Selected Media Resource Groups" list contains the value "MRG". There are up and down arrows between the two lists to facilitate moving items.

3. Enter **MRGL** for the **Name** (for this example).
4. Transfer media resource groups between the two lists.
 - a. In the **Media Resource Groups for this List** section, select **MRG** from the **Available Media Resource Groups** list.
 - b. Click **V** (between the two lists) to move the selected resource to **Selected Media Resource Groups** (for this example).

Configure the Duplex Streaming Parameter

To configure the duplex streaming parameter:

1. Click **System > Service Parameters**.
2. Select **Cisco UCM publisher** for the **Server**.
3. Select **Call Manager (Active)** for the **Service**.
4. Set **Duplex Streaming Enabled** to **True** for this example to enable the device to hear MoH when put on hold. When set to **False**, the device user hears silence when the call is put on hold

Configure Trunks

This example configures two trunks:

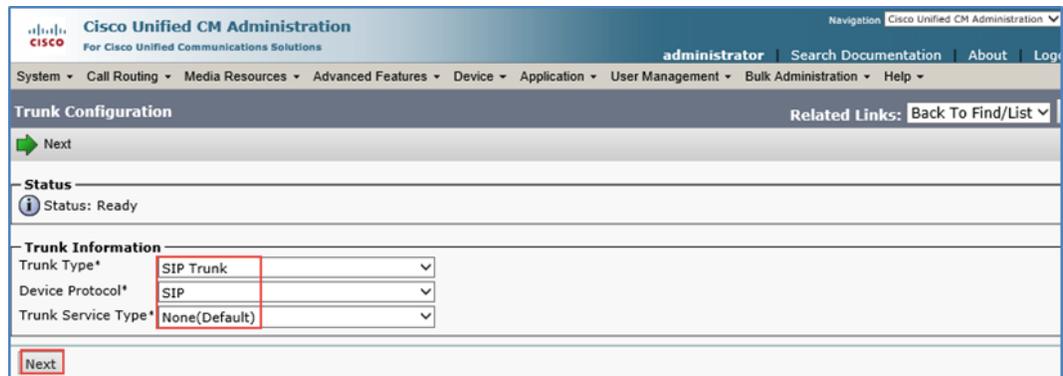
- Between the Cisco UCM and the PSTN gateway for calls to the PSTN
- Between the Cisco UCM and Cisco Unity Connection for voice mail

Configure the Cisco UCM - PSTN Gateway Trunk

To create a new trunk:

1. Click **Device > Trunk**.
2. Click **Add New**.

Cisco UCM: Add New Trunk



The screenshot shows the 'Cisco Unified CM Administration' interface. The page title is 'Trunk Configuration'. The 'Status' section shows '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 of the form.

3. In the **Trunk Information** section, do the following:
 - a. Select **SIP Trunk** for the **Trunk Type**.
 - b. Select **SIP** for the **Device Protocol**.
 - c. Select **None(Default)** for the **Trunk Service Type**.
4. Click **Next**.

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

The screenshot shows the 'Trunk Configuration' page for a SIP Trunk. At the top, there are buttons for 'Save', 'Delete', 'Reset', and 'Add New'. Below this is a 'Status' section showing 'Status: Ready'. The 'SIP Trunk Status' section shows 'Service Status: Full Service' and 'Duration: Time In Full Service: 0 day 22 hours 10 minutes'. The 'Device Information' section contains the following fields:

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. Enter a unique SIP Trunk name for the **Device Name**. This example uses **PSTN**. A **Description** is optional.
6. Select **Default** for the **Device Pool** (for this example).
7. Select **MRGL** for the **Media Resource Group List**.

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

Media Termination Point Required
 Retry Video Call as Audio
 Path Replacement Support
 Transmit UTF-8 for Calling Party Name
 Transmit UTF-8 Names in QSIG APDU
 Unattended Port
 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*
 Route Class Signaling Enabled*
 Use Trusted Relay Point*
 PSTN Access
 Run On All Active Unified CM Nodes

Intercompany Media Engine (IME)

E.164 Transformation Profile

MLPP and Confidential Access Level Information

MLPP Domain
 Confidential Access Mode
 Confidential Access Level

- Uncheck **Media Termination Point Required**.

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

Call Routing Information

Remote-Party-Id
 Asserted-Identity
 Asserted-Type*
 SIP Privacy*

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	Default	0	< None >	<input checked="" type="checkbox"/>

- Check **Redirecting Diversion Header Delivery - Inbound**.

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

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

Redirecting Party Transformation CSS < None >

Use Device Pool Redirecting Party Transformation CSS

Caller Information

Caller ID DN

Caller Name

10. Check **Redirecting Diversion Header Delivery - Outbound**.

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

SIP Information

Destination

Destination Address is an SRV

	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 >

Enable Trace

	Parameter Name	Parameter Value
1		

11. In the **SIP Information** section:
 - a. Enter **10.64.1.72** and **5060** for the **Destination Address** and **Destination Port**, respectively.
 - b. Select **Non Secure SIP Trunk Profile_Crestron** for the **SIP Trunk Security Profile**.
 - c. Select **Standard SIP Profile_Test** for the **SIP Profile**.
12. Click **Save**.

Configure Cisco UCM - Unity Connection Trunk

Configure a new trunk from Cisco UCM to the Unity Connection server, similar to the PSTN gateway trunk configuration. The following images illustrate the trunk parameter settings.

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

The screenshot shows the Cisco Unified CM Administration interface for configuring a SIP Trunk. The page title is "Trunk Configuration" and it includes navigation menus for System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, and Bulk Administration. The "Device Information" section is expanded, showing the following configuration:

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

Additional options are checked:

- Media Termination Point Required
- Retry Video Call as Audio

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

The screenshot shows advanced configuration options for the trunk. The following options are visible:

- Path Replacement Support
- Transmit UTF-8 for Calling Party Name
- Transmit UTF-8 Names in QSIG APDU
- Unattended Port
- 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
- PSTN Access
- 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 Voice Mail System - Unity Connection (3/6)

Call Routing Information

Remote-Party-Id

Asserted-Identity

Asserted-Type*

SIP Privacy*

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="< None >"/>	<input checked="" type="checkbox"/>

Cisco UCM: Trunk to Voice Mail 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	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input type="text" value="< 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 Voice Mail 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 Voice Mail 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

Configure the following route patterns.

- Route calls from the Cisco UCM to the PSTN.
- Restrict Caller ID on outgoing calls.
- Access voice mail.

To configure route patterns:

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

PSTN Route Pattern

Configure the 9.@ route pattern to enable outbound calling from Cisco UCM to PSTN using 9 as the access code.

The screenshots that follow show the configuration.

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

The screenshot displays the Cisco Unified CM Administration interface for configuring a Route Pattern. The page title is "Route Pattern Configuration". The status is "Ready".

Pattern Definition

- 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

Calling Party Transformations

- Use Calling Party's External Phone Number Mask
- Calling Party Transform Mask: (empty)
- Prefix Digits (Outgoing Calls): (empty)
- 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 >	
Save Delete Copy Add New		

Restricted Caller ID Route Pattern

Configure the *67.@ route pattern to restrict Caller ID on outbound calls.

The screenshots that follow 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 Document
System Call Routing Media Resources Advanced Features Device Application User Management Bulk Administration Help		
Route Pattern Configuration		Related Links:
Save Delete Copy 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 >	

Voice Mail Pilot Number Route Pattern

Configure the **2900** route pattern to route the voice mail pilot number (2900) to the Unity Connection server.

The screenshots that follow show the configuration.

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

Route Pattern Configuration		Related Links: Back To Find/List ▼ Go
Save <input type="checkbox"/> Delete <input checked="" type="checkbox"/> Copy <input type="checkbox"/> Add New <input checked="" type="checkbox"/>		
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 - Voice Mail Pilot Number (2/2)

Block this pattern | No Error

Call Classification*

External Call Control Profile

Allow Device Override
 Provide Outside Dial Tone
 Allow Overlap Sending
 Urgent Priority

Require Forced Authorization Code

Authorization Level*

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*

Calling Name Presentation*

Calling Party Number Type*

Calling Party Numbering Plan*

Connected Party Transformations

Connected Line ID Presentation*

Connected Name Presentation*

Called Party Transformations

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

Called Party Number Type*

Called Party Numbering Plan*

ISDN Network-Specific Facilities Information Element

Network Service Protocol

Carrier Identification Code

Network Service	Service Parameter Name	Service Parameter Value
<input type="text" value="-- Not Selected --"/>	<input type="text" value="< Not Exist >"/>	<input type="text"/>

Configure Voice Mail

Perform a Cisco UCM - Cisco Unity Connection SIP integration to test voice mail scenarios.

Configure Voice Mail Pilot and Voice Mail Profile on Cisco UCM

To configure voice mail pilot:

1. Click **Advanced Features** > **Voice Mail** > **Voice Mail Pilot**.
2. Click **Add New**.

Cisco UCM: Voice Mail Pilot Number Configuration

The screenshot shows the Cisco Unified CM Administration interface. The top navigation bar includes 'System', 'Call Routing', 'Media Resources', 'Advanced Features', 'Device', 'Application', 'User Management', 'Bulk Administration', and 'Help'. The 'Advanced Features' menu is highlighted with a red box. Below the navigation bar, the 'Voice Mail Pilot Configuration' page is displayed. It features a 'Save', 'Delete', and 'Add New' button bar. The 'Status' section shows 'Status: Ready'. The 'Voice Mail Pilot Information' section is highlighted with a red box and contains the following fields: 'Voice Mail Pilot Number' (2900), 'Calling Search Space' (< None >), 'Description' (empty), and a checked checkbox 'Make this the default Voice Mail Pilot for the system'. At the bottom, there are 'Save', 'Delete', and 'Add New' buttons.

3. Enter a new pilot number for the **Voice Mail Pilot Number**. This example uses **2900**.
4. Check **Make this the default Voice Mail Pilot for the system**.

Configure a voice mail profile with this pilot number as shown below.

Cisco UCM: Voice Mail Profile Configuration

The screenshot shows the Cisco Unified CM Administration interface. The top navigation bar includes 'System', 'Call Routing', 'Media Resources', 'Advanced Features', 'Device', 'Application', 'User Management', 'Bulk Administration', and 'Help'. The 'Voice Mail Profile Configuration' page is displayed. It features a 'Save' button. The 'Status' section shows 'Status: Ready'. The 'Voice Mail Profile Information' section is highlighted with a red box and contains the following fields: 'Voice Mail Profile Name*' (UnityConnection), 'Description' (empty), 'Voice Mail Pilot**' (2900/< None >), 'Voice Mail Box Mask' (empty), and a checked checkbox 'Make this the default Voice Mail Profile for the System'. At the bottom, there is a 'Save' button.

Configure New Phone System on Unity Connection

To configure a new phone system, after logging into Unity Connection:

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

Cisco Unity Connection: Phone System

The screenshot displays the Cisco Unity Connection Administration interface. The left sidebar shows a navigation tree with 'Telephony Integrations > Phone System' selected. The main content area is titled 'Phone System Basics (CUCM11.0)'. It includes a 'Phone System Name*' field containing 'CUCM11.0', a 'Default TRAP Phone System' checkbox, 'Message Waiting Indicators' settings, 'Call Loop Detection by Using DTMF' settings, 'Call Loop Detection by Using Extension' settings, 'Phone View Settings', and 'Outgoing Call Restrictions'. The 'Outgoing Call Restrictions' section has 'Enable outgoing calls' selected. The 'Phone System Name*' field is highlighted with a red box. The 'Save' button is also highlighted with a red box. The interface includes 'Save', 'Delete', 'Previous', and 'Next' buttons at the top and bottom.

3. Enter **CUCM11.0** for the **Phone System Name** (for this example).
4. Click **Save**.

On the **Phone System Basics** page, in the **Related Links** section, select **Add Port Group** and then click **Go**.

Cisco Unity Connection: Add New 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, and Telephony Integrations. The main content area is titled 'New Port Group' and includes a 'Save' button at the top. Below this, the 'New Port Group' section has a 'Phone System' dropdown menu set to 'CUCM11.0'. The 'Create From' section has two radio buttons: 'Port Group Type SIP' (selected) and 'Port Group'. The 'Port Group Description' section includes a 'Display Name*' field with 'CUCM11.0-1', an 'Authenticate with SIP Server' checkbox, and fields for 'Authentication Username', 'Authentication Password', and 'Contact Line Name'. The 'Primary Server Settings' section includes a 'SIP Security Profile' dropdown set to '5060', a 'SIP Transport Protocol' dropdown set to 'TCP', and an 'IPv4 Address or Host Name' field with '10.80.25.2'. There are also fields for 'IPv6 Address or Host Name' and 'Port' (set to '5060'). A 'Save' button is located at the bottom of the form. A note at the bottom states 'Fields marked with an asterisk (*) are required.'

To add a new port group:

1. Select **CUCM11.0** (created earlier) for the **Phone System**.
2. Click **Port Group Type** for **Create From**, and select **SIP**.
3. Enter the IP address (or host name) of the primary Cisco UCM server integrated with Cisco Unity Connection for the **IPv4 Address or Host Name**. This example uses **10.80.25.2**.
4. Click **Save**.

On the **Phone System Basics** page, in the **Related Links** section, select **Add Ports** and then click **Go**.

Cisco Unity Connection: Related Links to Add Port

The screenshot shows the Cisco Unity Connection Administration interface. The main content area is titled "Port Group Basics (CUCM11.0-1)". In the top right corner, there is a "Related Links" dropdown menu with "Add Ports" selected and highlighted. Below this, the "Status" section contains a warning icon and the text: "The phone system cannot take calls if it has no ports. Use the Related Links to add ports." The "Port Group" section includes fields for "Display Name*" (CUCM11.0-1), "Integration Method" (SIP), and "Reset Status" (Reset Not Required). The "Session Initiation Protocol (SIP) Settings" section has checkboxes for "Register with SIP Server" and "Authenticate with SIP Server", along with input fields for "Authentication Username", "Authentication Password", and "Contact Line Name". The "SIP Security Profile" is set to "5050" and "SIP Transport Protocol" is set to "TCP". The "Advertised Codec Settings" section features a table with columns "Display Name" and "Packet Size".

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

The "Message Waiting Indicator Settings" section includes a checked box for "Enable Message Waiting Indicators" and input fields for "Delay between Requests" (0 milliseconds), "Maximum Concurrent Requests" (0), "Retries After Successful Attempt" (0), and "Retry Interval After Successful Attempt" (5 milliseconds).

On the **New Port** page, configure the settings as shown below, and then click **Save**.

Cisco Unity Connection: Add New Port

The screenshot shows the Cisco Unity Connection Administration interface. The left sidebar contains a navigation tree 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. Under Telephony Integrations, 'Phone System' is expanded to show 'Port Group', 'Port', 'Speech Connect Port', and 'Trunk'. The main content area is titled 'New Port' and includes a 'Port' tab, 'Reset', and 'Help' options. A 'Status' box contains two warning messages: 'Because it has no port groups, PhoneSystem is not listed in the Phone system field.' and 'Because it has no port groups, test is not listed in the Phone system field.' Below this is a 'Save' button. The 'New Phone System Port' section includes a checked 'Enabled' checkbox, a 'Number of Ports' input field with the value '1', a 'Phone System' dropdown menu set to 'CUCM11.0', a 'Port Group' dropdown menu set to 'CUCM11.0-1', and a 'Server' dropdown menu set to 'dus35cuc.job.tskvizion.com'. The 'Port Behavior' section has four checked options: 'Answer Calls', 'Perform Message Notification', 'Send MWI Requests (may also be disabled by the port group)', and 'Allow TRAP Connections'. A 'Save' button is located at the bottom of this section.

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

1. Click **Telephony Integrations > Phone System > CUCM11.0**.
2. On the **Phone System Basics** page, click **Edit > Cisco Unified Communications Manager AXL Servers**.
3. Click **Add New**.

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' and 'Phone System' highlighted. The main content area is titled 'Edit AXL Servers' and includes a search bar, a 'Save' button, and a table of AXL servers. The table has columns for 'Order', 'IP Address', and 'Port'. The second row is highlighted with a red box, showing an IP address of 10.80.25.3 and a port of 5060. Below the table are 'AXL Server Settings' fields for Username, Password, and Cisco Unified Communications Manager Version.

Order	IP Address	Port	Test
0	10.80.25.2	5060	Test
1	10.80.25.3	5060	Test

4. Enter the Cisco UCM subscriber IP Address and Port in the second row. This example uses **10.80.25.3** and **5060**, respectively.
5. Click **Save**.

Configure a Voice Mail User

To configure a new user with a voice mail box, after logging into Unity Connection:

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

Cisco Unity Connection: Add User

The screenshot shows the Cisco Unity Connection Administration web interface. The left sidebar is expanded to show the 'Users' menu. The main content area is titled 'New User' and contains a form for creating a new user. The form includes the following fields and values:

- User Type:** User With Mailbox
- Based on Template*:** voicemailusertemplate
- Alias*:** Crestron_Avia
- Extension*:** 2500
- Mailbox Store:** Unity Messaging Database -1

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

3. Select **voicemailusertemplate** for **Based on Template** (for this example).
4. Enter **Crestron_Avia** for the **Alias** (for this example).
5. Enter **2500** for the **Extension** (for this example).
6. Click **Save**.

Cisco Unity Connection: Assign Phone System to User

The screenshot shows the 'Edit User Basics' page for user 'Crestron_Avia'. The left sidebar contains 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_Avia)' and includes a menu with 'User', 'Edit', 'Refresh', and 'Help'. Below the menu are buttons for 'Save', 'Delete', 'Previous', and 'Next'. The 'Name' section contains fields for 'Alias*' (Crestron_Avia), 'First Name', 'Last Name', 'Display Name' (Crestron_Avia), 'SMTP Address' (crestron_avia@clus35cuc.lab.tekvizion.com), 'Initials', 'Title', and 'Employee ID'. The 'LDAP Integration Status' section has two radio buttons: 'Integrate with LDAP Directory' (unselected) and 'Do Not Integrate with LDAP Directory' (selected). The 'Phone' section includes fields for 'Extension*' (2500), '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), and 'Active Schedule' (Weekdays). A 'View' button is next to the 'Active Schedule' field. At the bottom, there is a checkbox for 'Set for Self-enrollment at Next Sign-In'.

7. On the screen that follows, select **CUCM11.0** (configured earlier for this example) for the **Phone System**.
8. Click **Save**.

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