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

Configuration Guide

Crestron Electronics, Inc.

Contents

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.

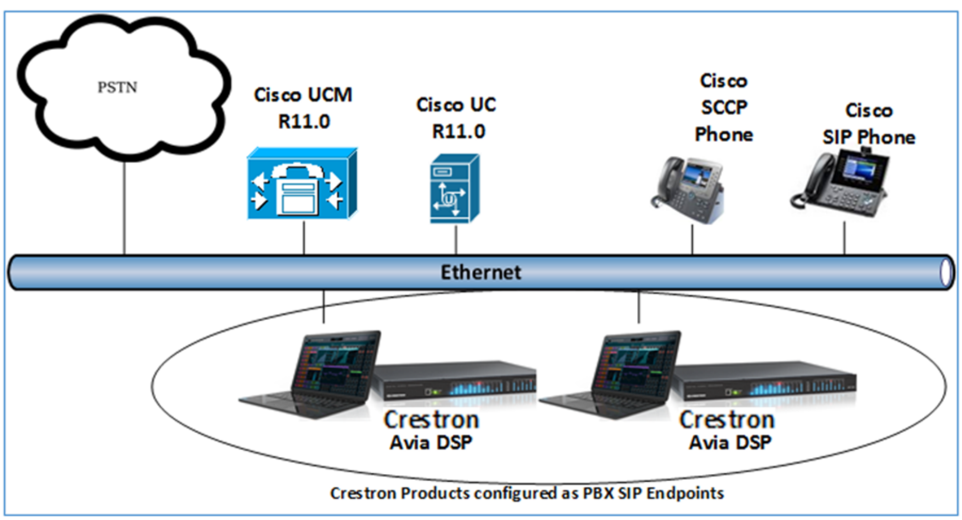
Audience

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 too 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 **routetrace**) 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>routetrace ?
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

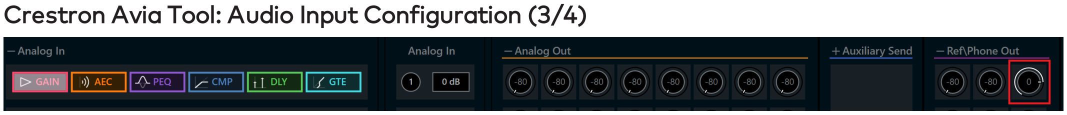
Use the Crestron Avia tool to select and configure the DSP device.

Input Configuration

To configure the analog input:

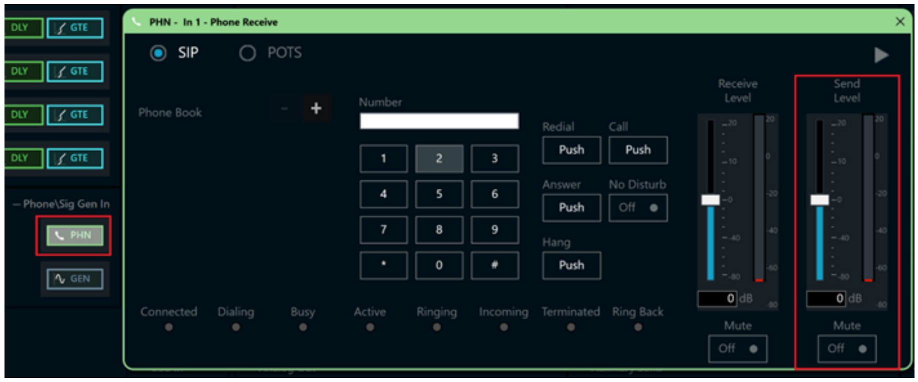
1. Click **Signal**.
   Crestron Avia tool: Audio Input Configuration (1/4)
   

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)

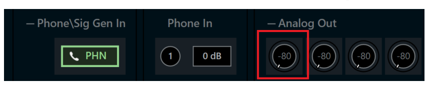


Output Configuration

To configure the analog output:

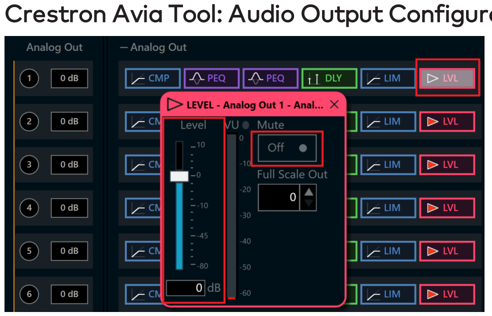
1. Under **Phone In 1** (first row), click **Analog Out** (left-most column) and enter **0** as the decibel value.

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**.

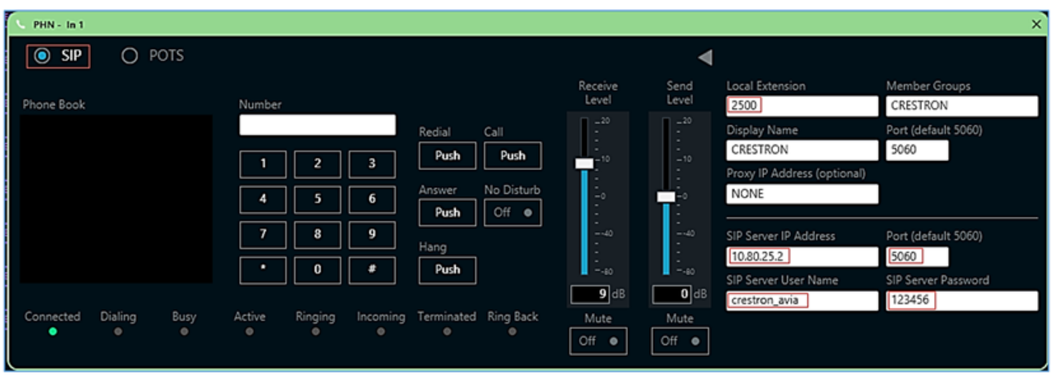
    Crestron Avia Tool: Audio Output Configuration (3/3)

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 Cicso 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.

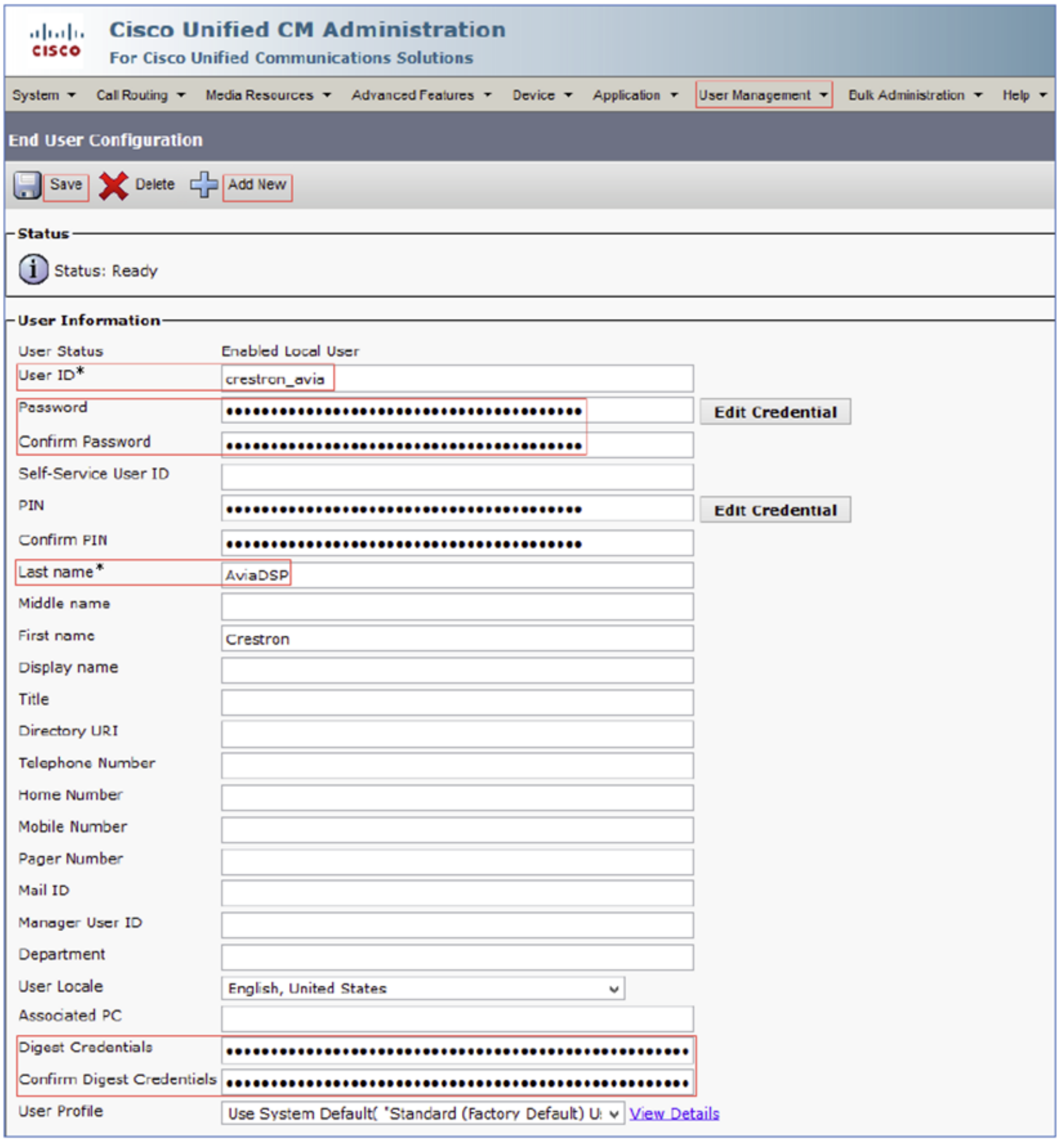
NOTE: Confirm that the general installation and basic Cisco UCM configuration have been administered.

Configure the User

To configure the end user:

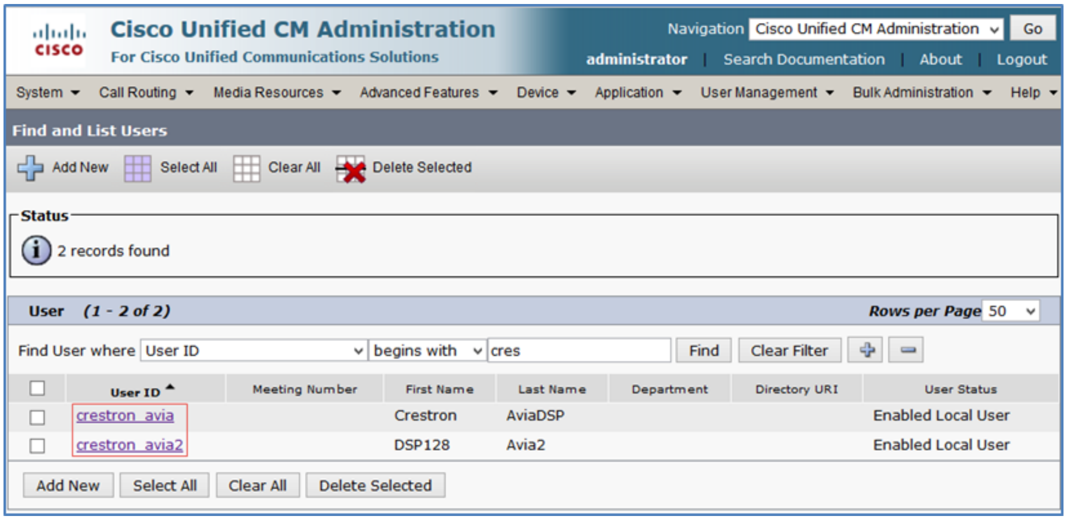
1. Click **User Management** > **End User**.
2. Click **Add New**.

Cisco UCM: End User Configuration



3. Enter a unique end user identification name for the **User ID**. This example uses **DSP1** and **DSP2crestron_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



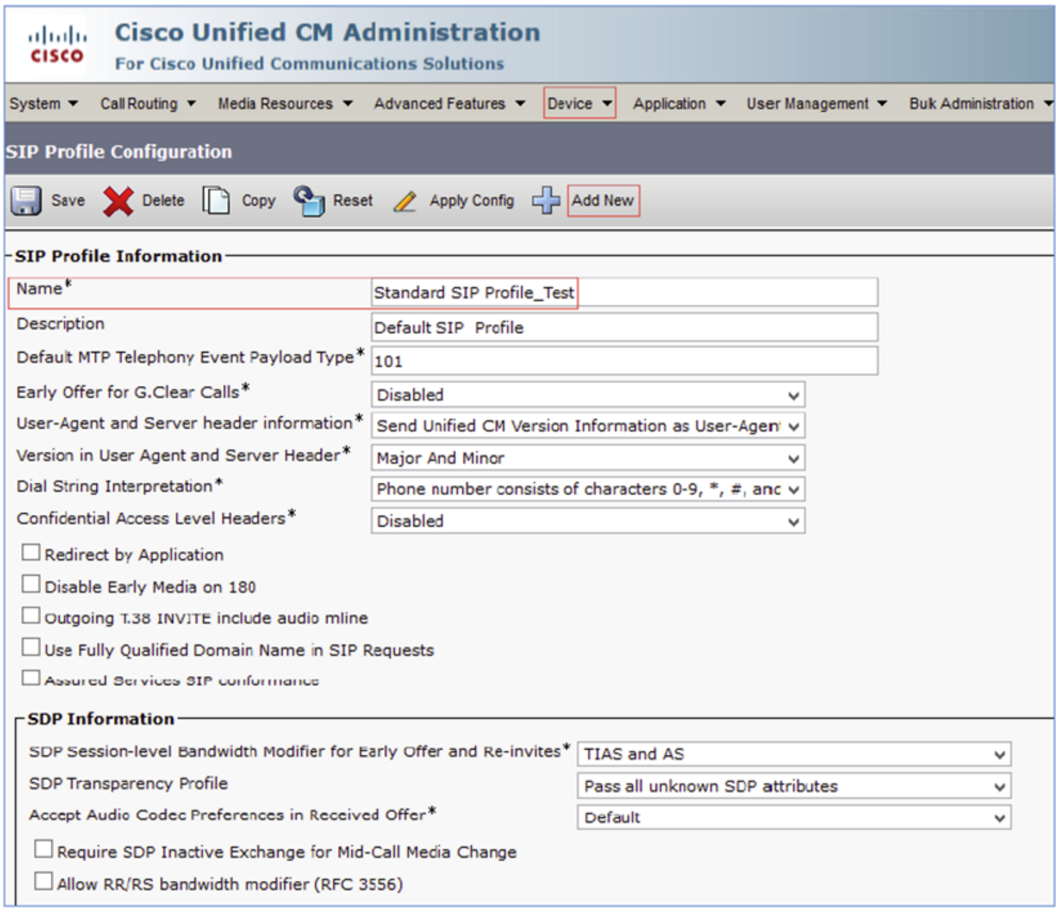
Configure a SIP Profile

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)



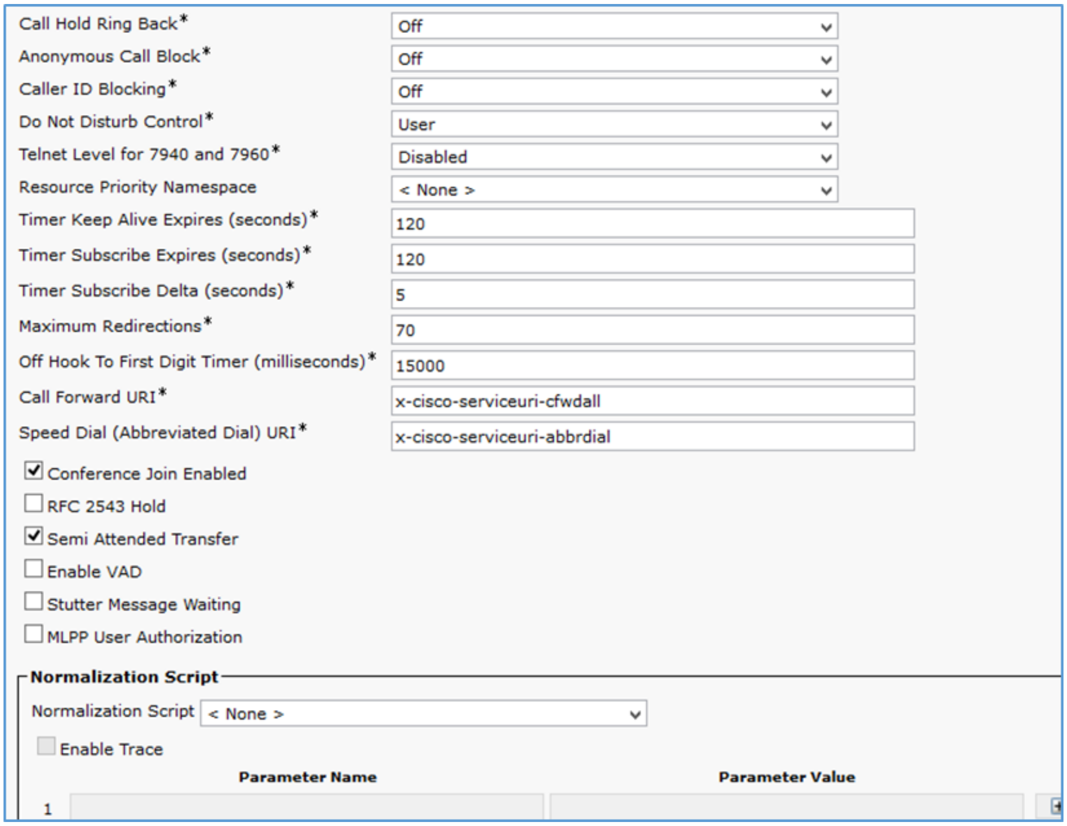
3. Enter **Standard SIP Profile_Test** for the **Name** (for this example).

Cisco UCM: SIP Profile Configuration (2/4)

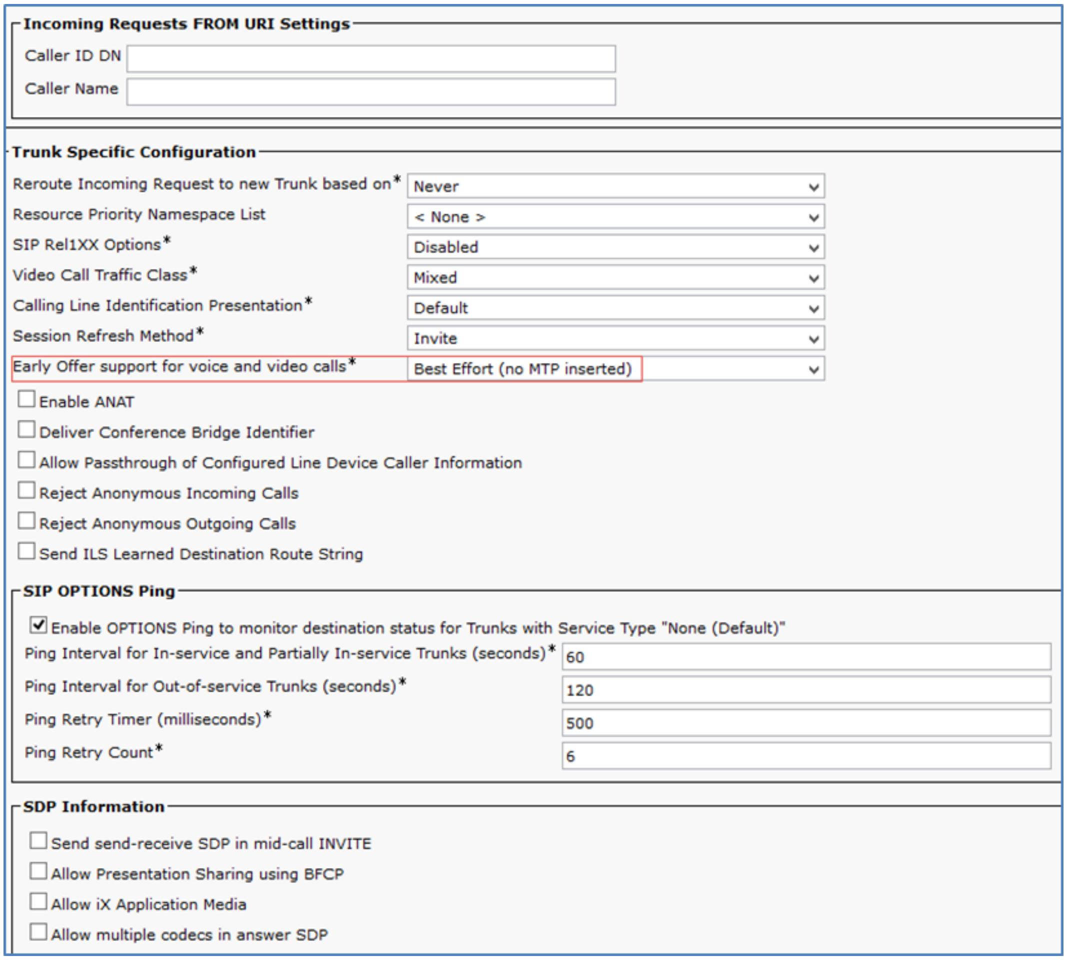
Parameters used in Phone

Timer Invite Expires (seconds)*	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	(●) Common Port Range for Audio and Video ( ) 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
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

Cisco UCM: SIP Profile Configuration (3/4)



Cisco UCM: SIP Profile Configuration (4/4)



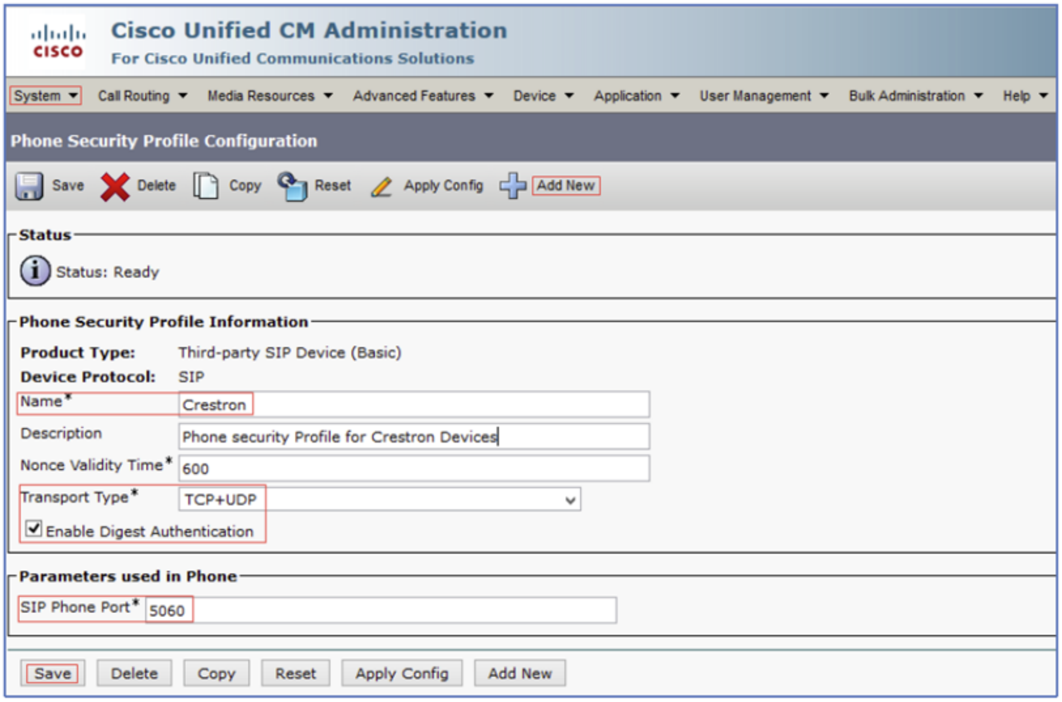
4. Select **Best Effort (no MTP inserted)** for **Early Offer support for voice and video calls**.
5. Leave all other fields at the default values.
6. Click **Save**.
7. Click **Apply Config**.

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

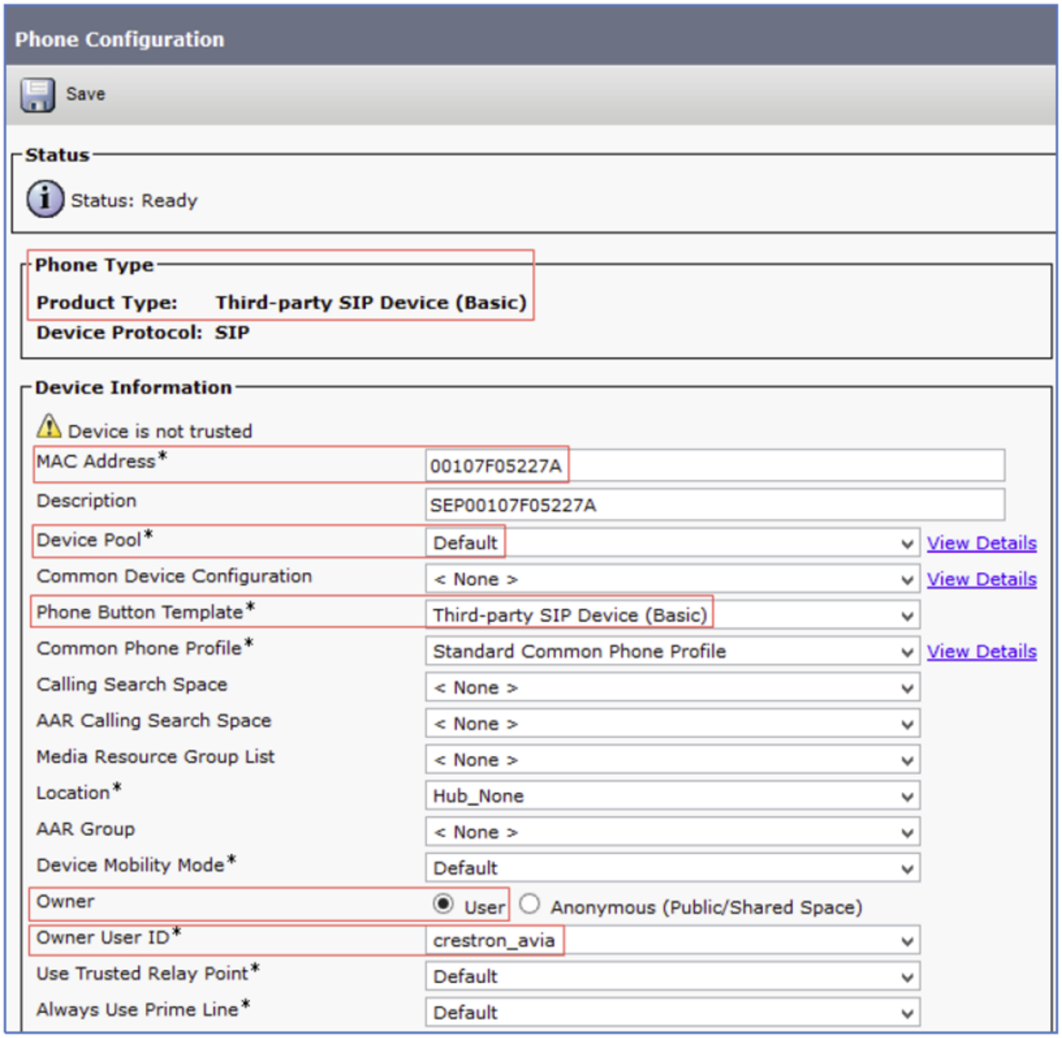
3. Enter **Crestron** for the **Name** (for this example).
4. Select **TCP+UDP** for the **Transport Type**.
5. Check **Enable Digest Authentication**.
6. Click **Save**.

Configure the Crestron Device as a Third-party SIP Device

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

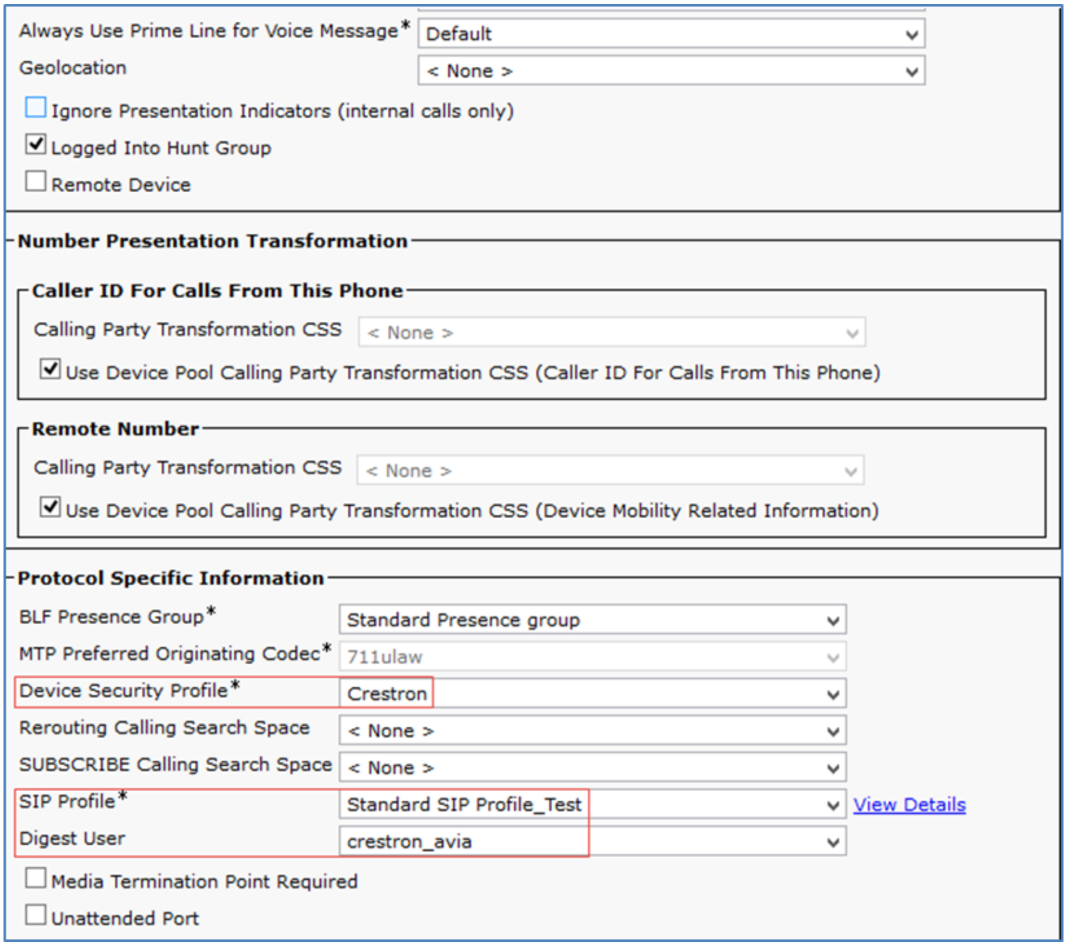
1. Click **Device** > **Phone**.
2. Click **Add New**.

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



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)



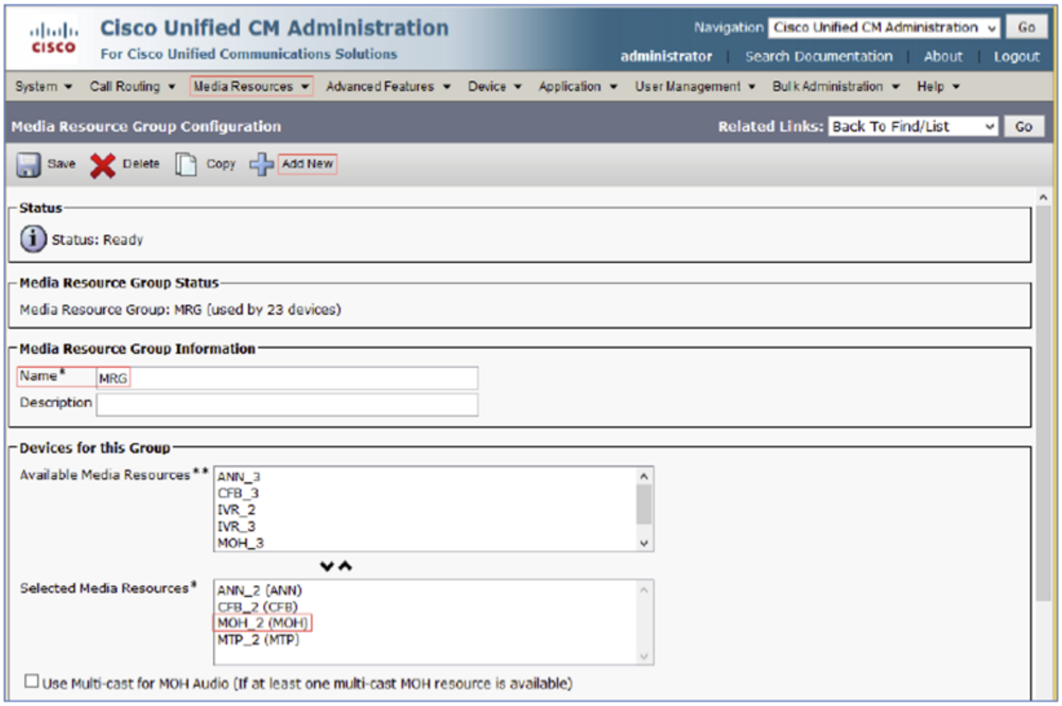
10. Select **Crestron** (configured earlier for this example) for the **Device Security Profile**.
11. Select **Standard SIP Profile_Test** (configured earlier for this example) for the **SIP Profile**.
12. Select **crestron_avia** for the first Crestron Avia DSP device and **crestron_avia2** for the 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



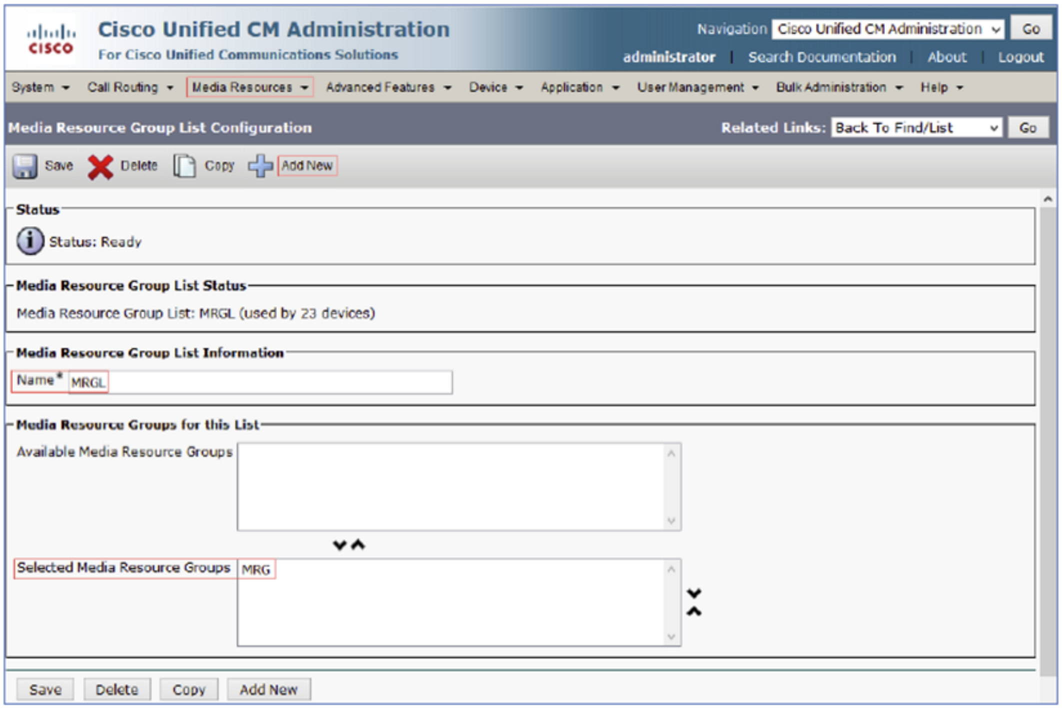
3. Enter **MRG** for the **Name** (for this example).
4. Transfer media resources between the two lists. Resources (added earlier) are available for use with the Cisco UCM.

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



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:

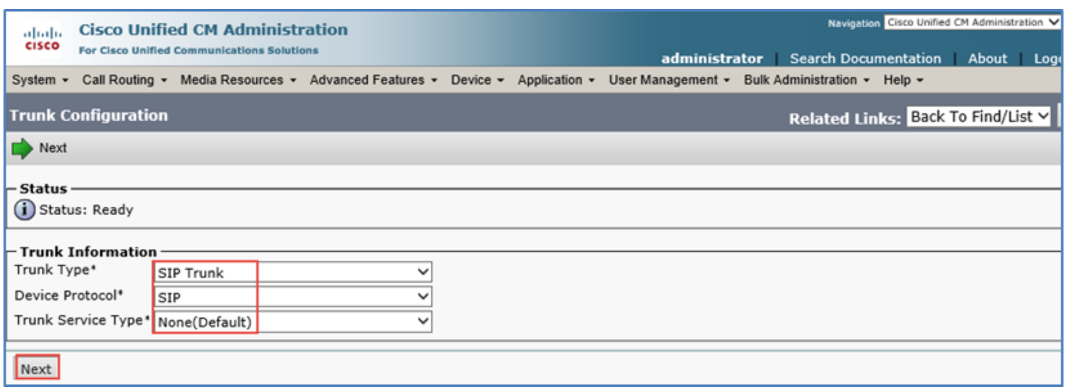
- Between the Cisco UCM and the PSTN gateway for calls to the PSTN
- Between the Cisco UCM and Cisco Unity Connection for voice mail

Configure the Cisco UCM - PSTN Gateway Trunk

To create a new trunk:

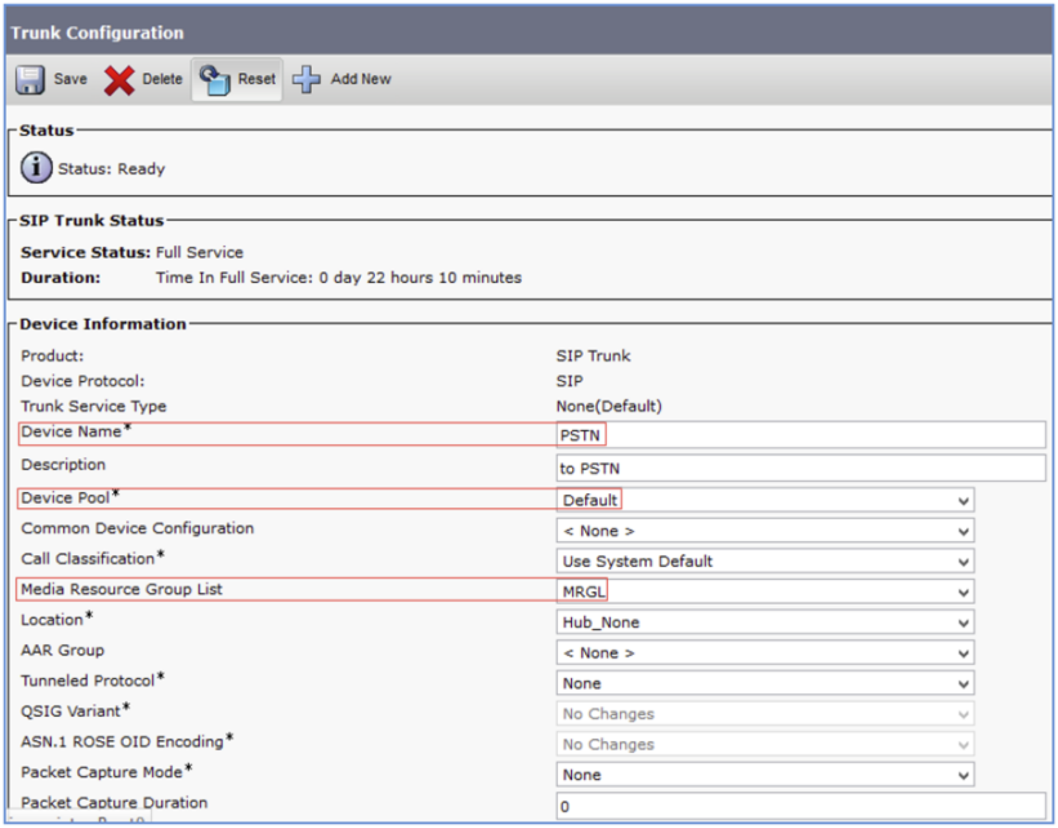
1. Click **Device** > **Trunk**.
2. Click **Add New**.

Cisco UCM: Add New Trunk



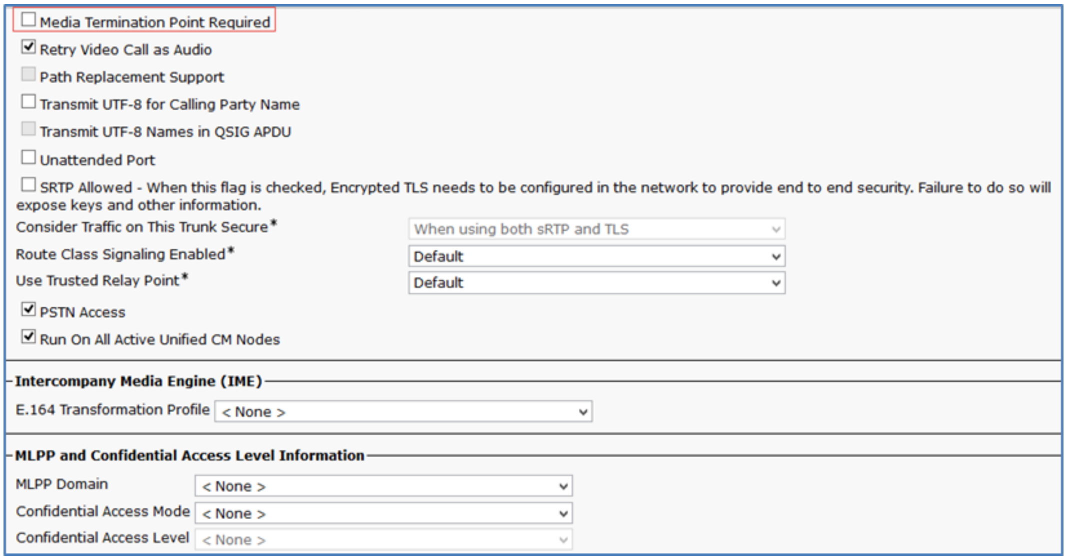
3. In the **Trunk Information** section, do the following:
   a. Select **SIP Trunk** for the **Trunk Type**.
   b. Select **SIP** for the **Device Protocol**.
   c. Select **None(Default)** for the **Trunk Service Type**.
4. Click **Next**.

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



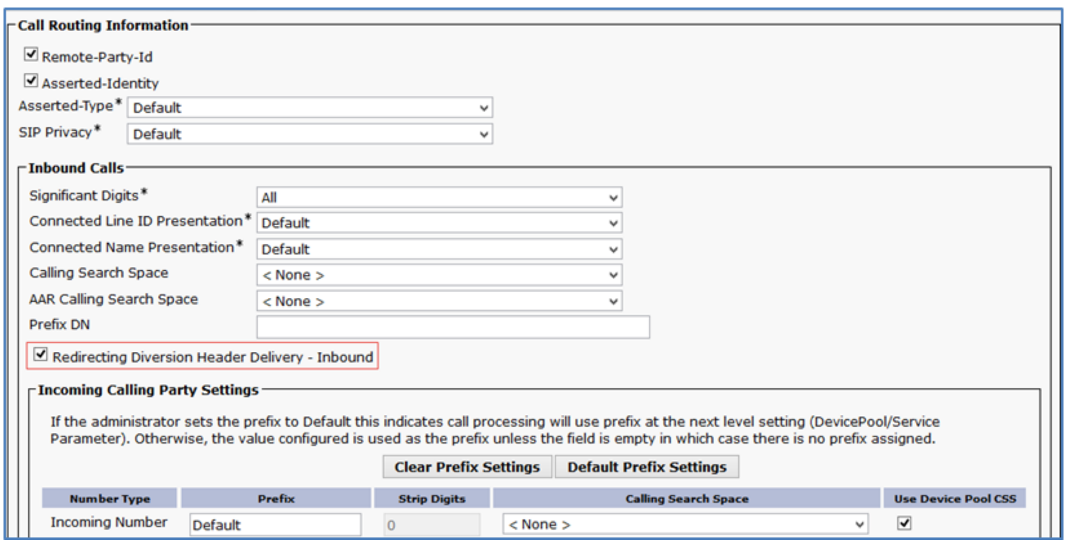
5. Enter a unique SIP Trunk name for the **Device Name**. This example uses **PSTN**. A **Description** is optional.
6. Select **Default** for the **Device Pool** (for this example).
7. Select **MRGL** for the **Media Resource Group List**.

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



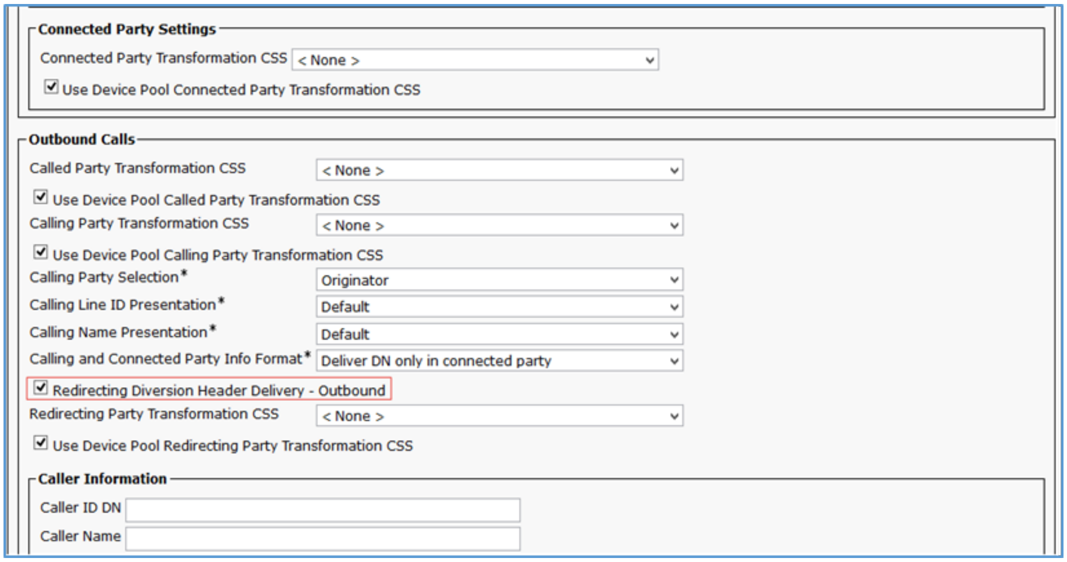
8. Uncheck **Media Termination Point Required**.

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



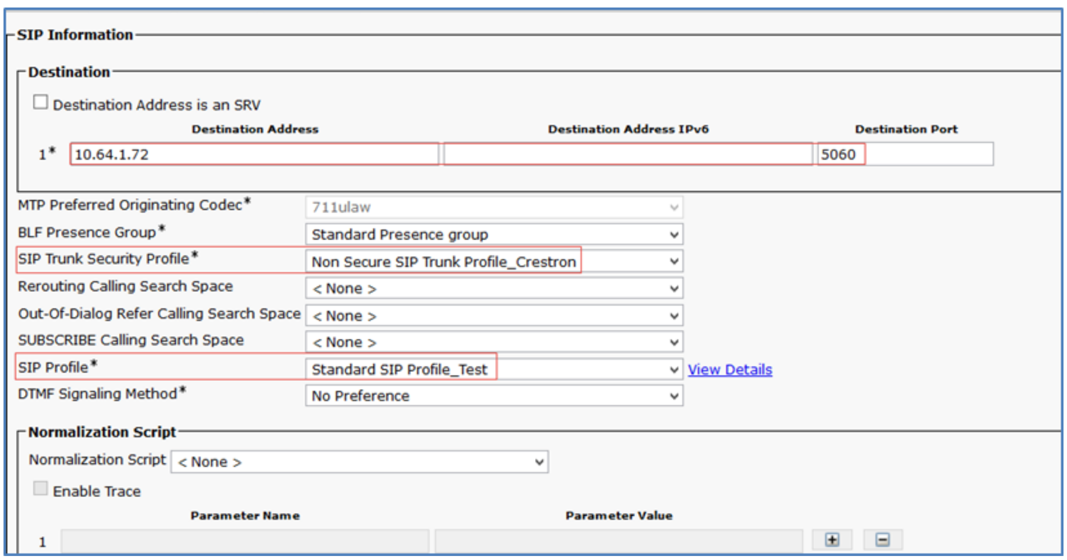
9. Check **Redirecting Diversion Header Delivery - Inbound**.

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



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

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

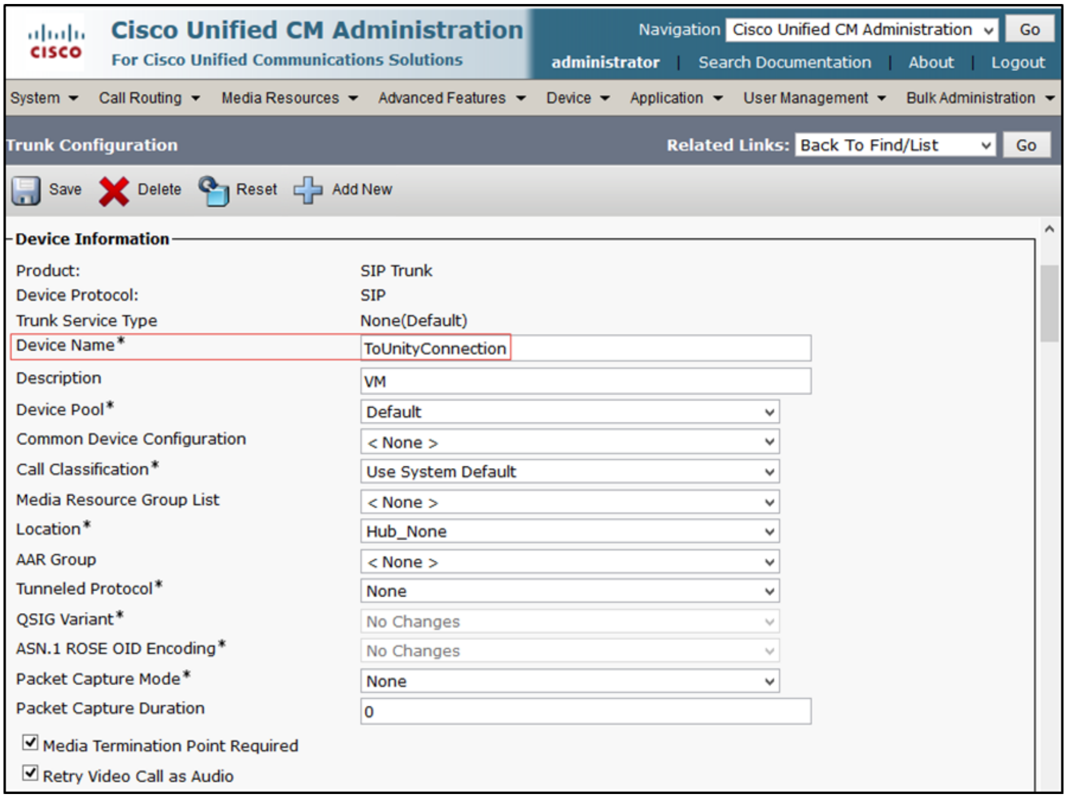


11. In the **SIP Information** section:
    a. Enter **10.64.1.72** and **5060**for the **Destination Address** and **Destination Port**, respectively.
    b. Select **Non Secure SIP Trunk Profile_Crestron** for the **SIP Trunk Security Profile**.
    c. Select **Standard SIP Profile_Test** for the **SIP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

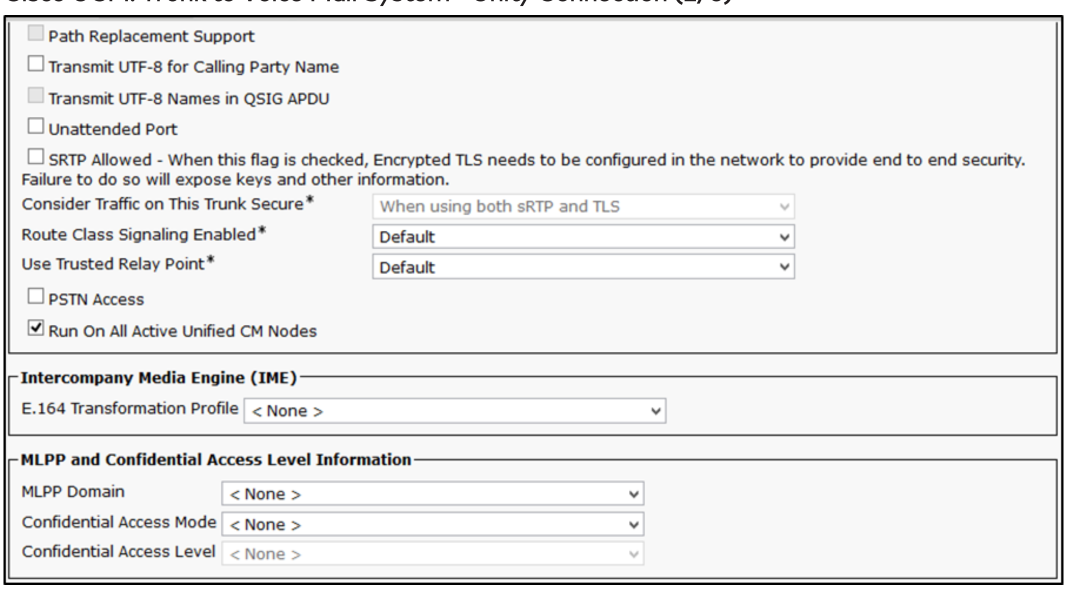
Configure Cisco UCM - Unity Connection Trunk

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.

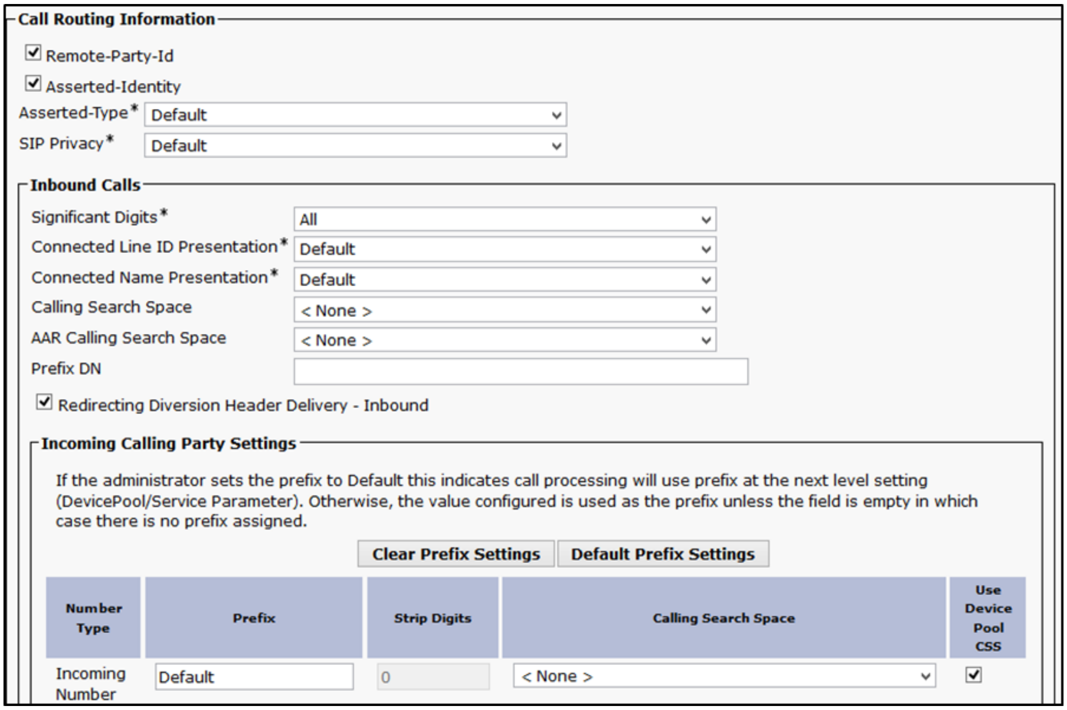
Cisco UCM: Trunk to Voice Mail System - Unity Connection (1/6)



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



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



Call Routing Information

- [x] Remote-Party-Id
- [x] 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	

- [x] Redirecting Diversion Header Delivery - Inbound

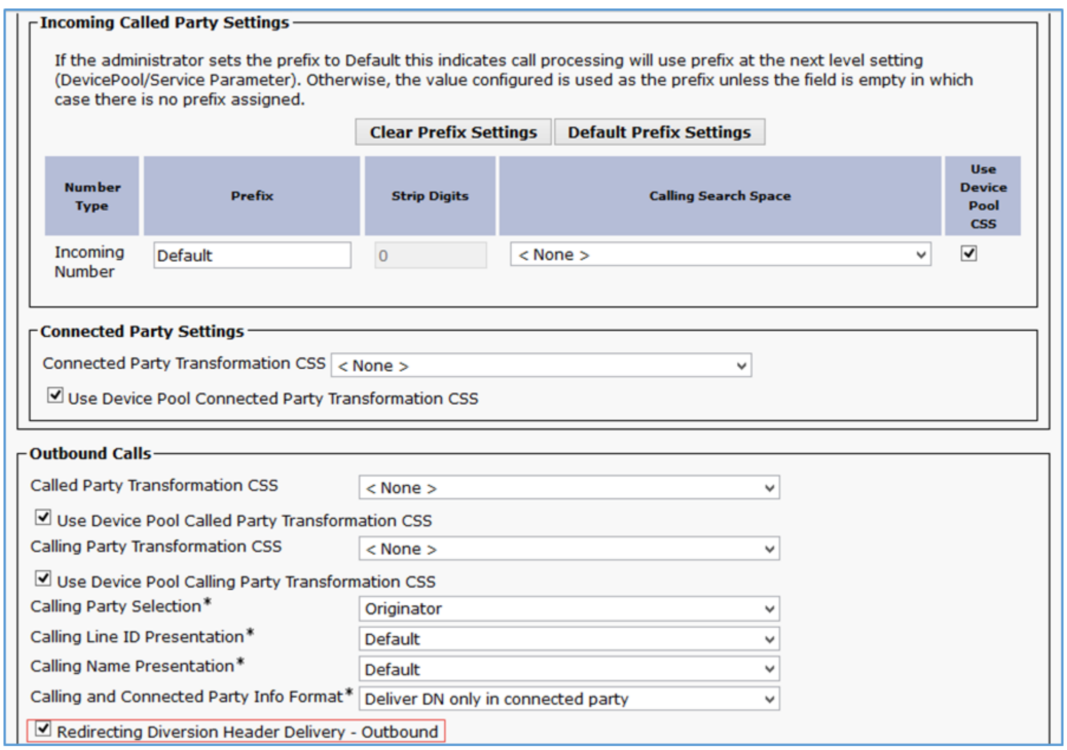
Incoming Calling Party Settings

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

Clear Prefix Settings | Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	☑

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.

Clear Prefix Settings | Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	☑

Connected Party Settings

Connected Party Transformation CSS	< None >

- [x] Use Device Pool Connected Party Transformation CSS

Outbound Calls

Called Party Transformation CSS	< None >

- [x] Use Device Pool Called Party Transformation CSS

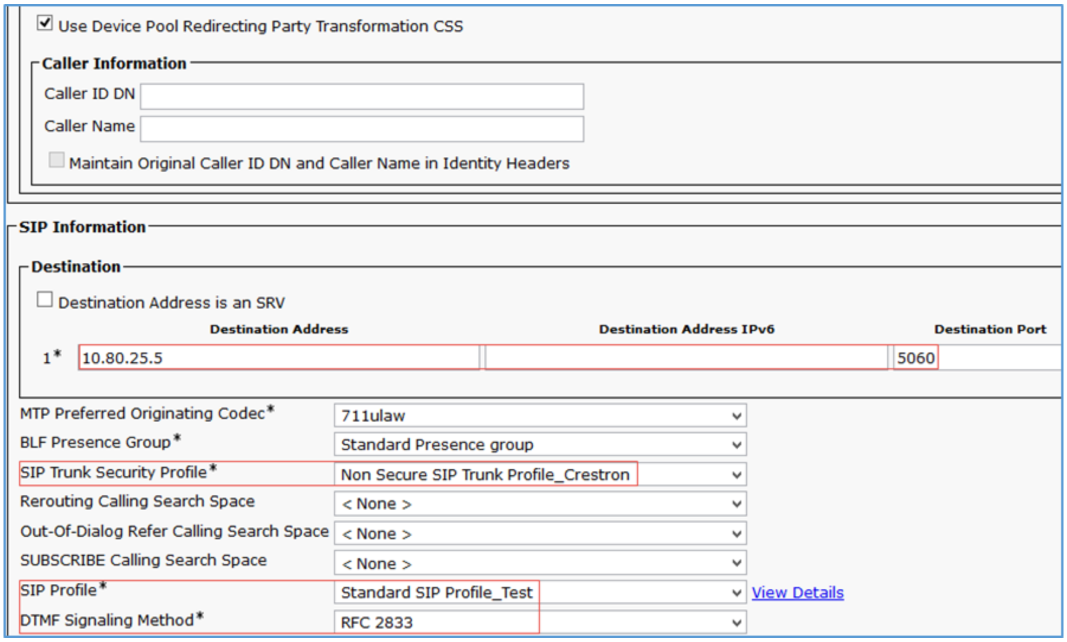
Calling Party Transformation CSS	< None >

- [x] Use Device Pool Calling Party Transformation CSS

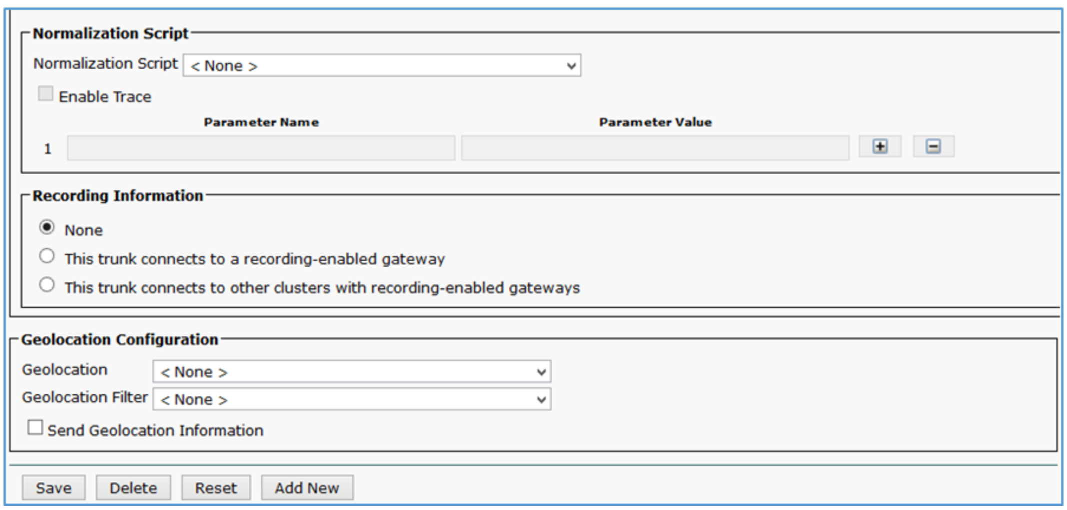
Calling Party Selection*	Originator
Calling Line ID Presentation*	Default
Calling Name Presentation*	Default
Calling and Connected Party Info Format*	Deliver DN only in connected party

- [x] Redirecting Diversion Header Delivery - Outbound

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



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



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:

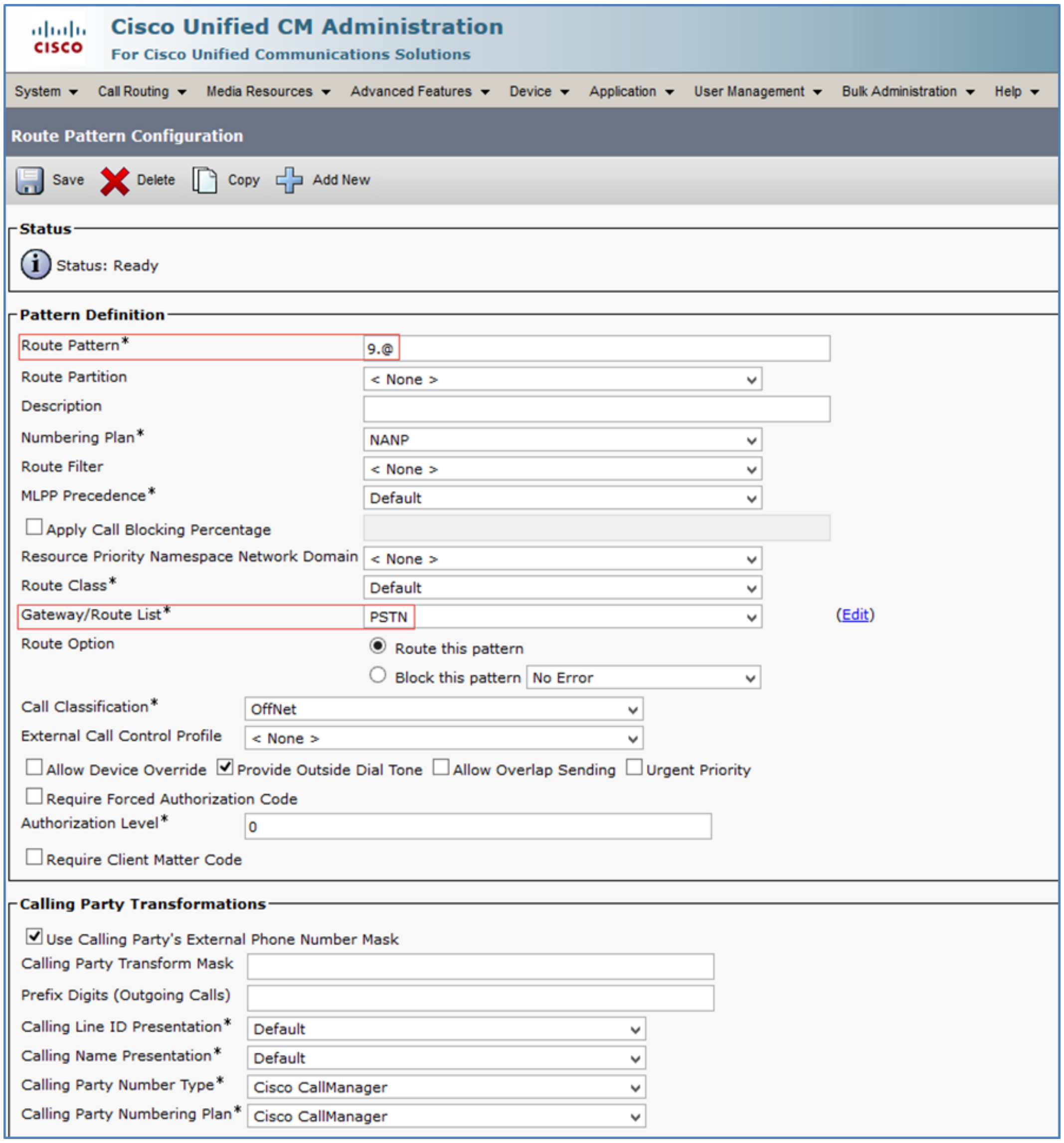
1. Click **Call Routing** > **Route/Hunt** > **Route Pattern**.
2. Click **Add New**.
3. Enter the desired information and then click **Save**.

PSTN Route Pattern

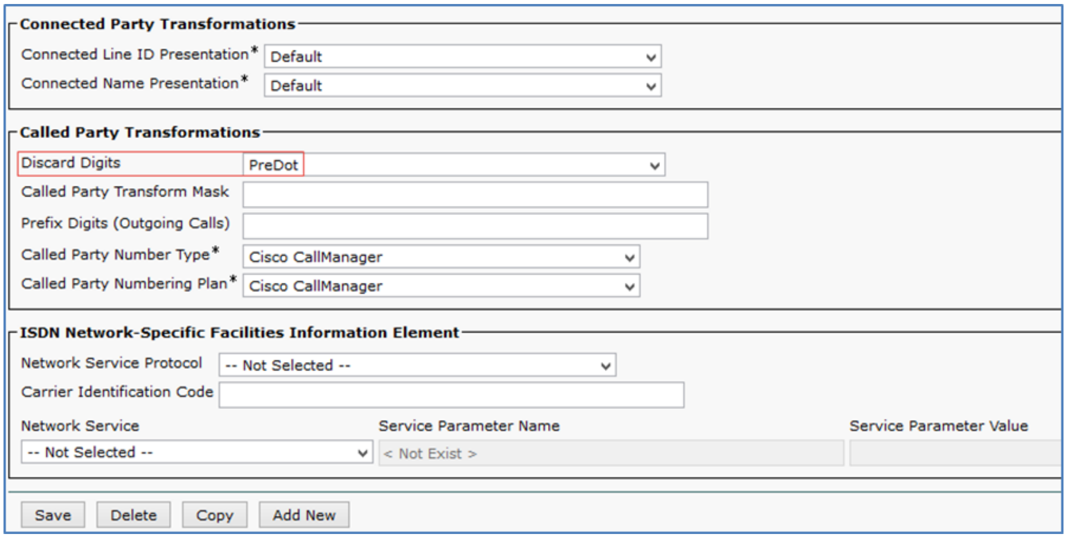
Configure the **9.@** route pattern to enable outbound calling from Cisco UCM to PSTN using 9 as the access code.

The screenshots that follow show the configuration.

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



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

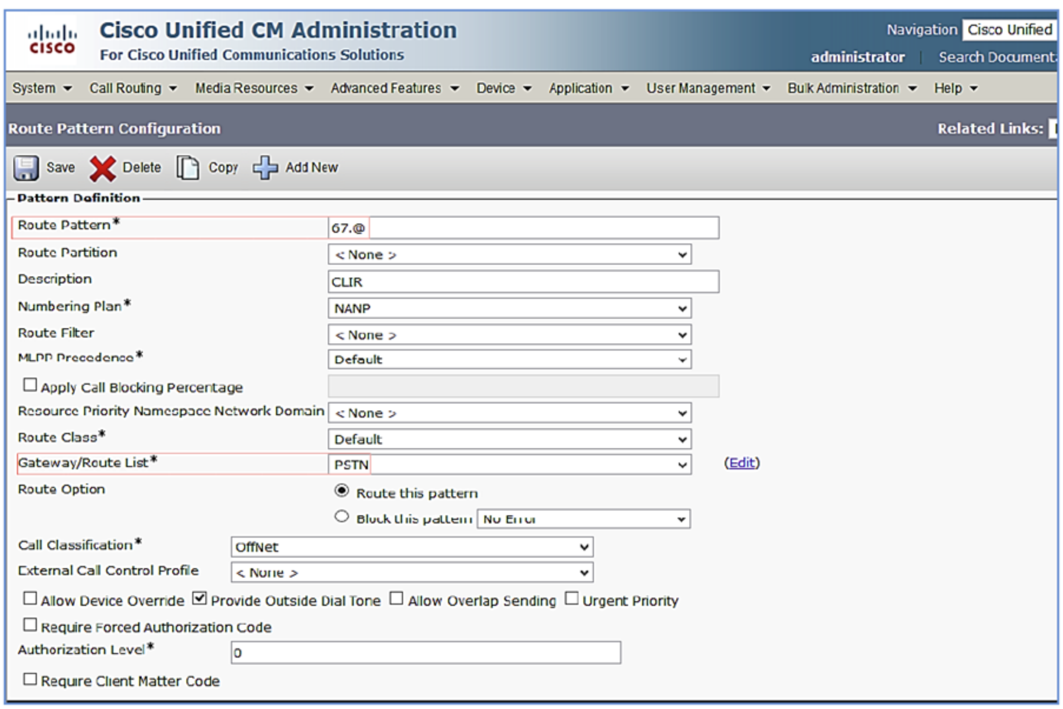


Restricted Caller ID Route Pattern

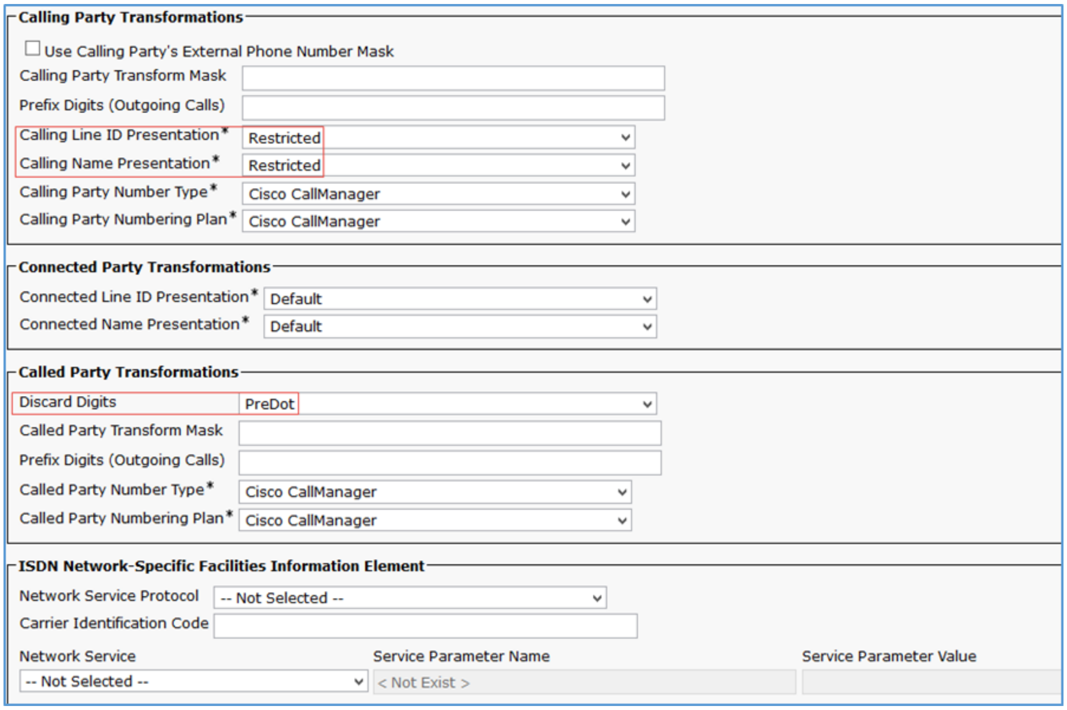
Configure the ***67.@** route pattern to restrict Caller ID on outbound calls.

The screenshots that follow show the configuration.

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



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

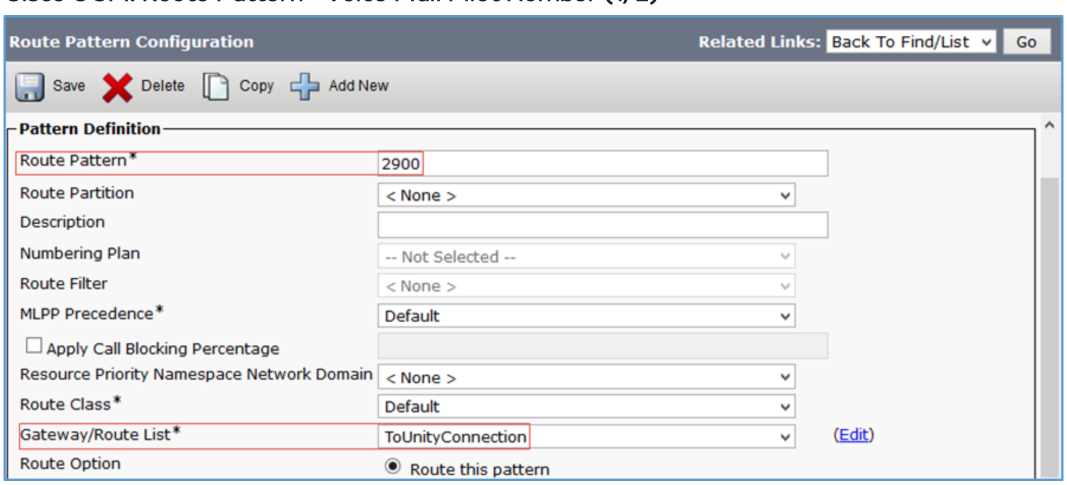


Voice Mail Pilot Number Route Pattern

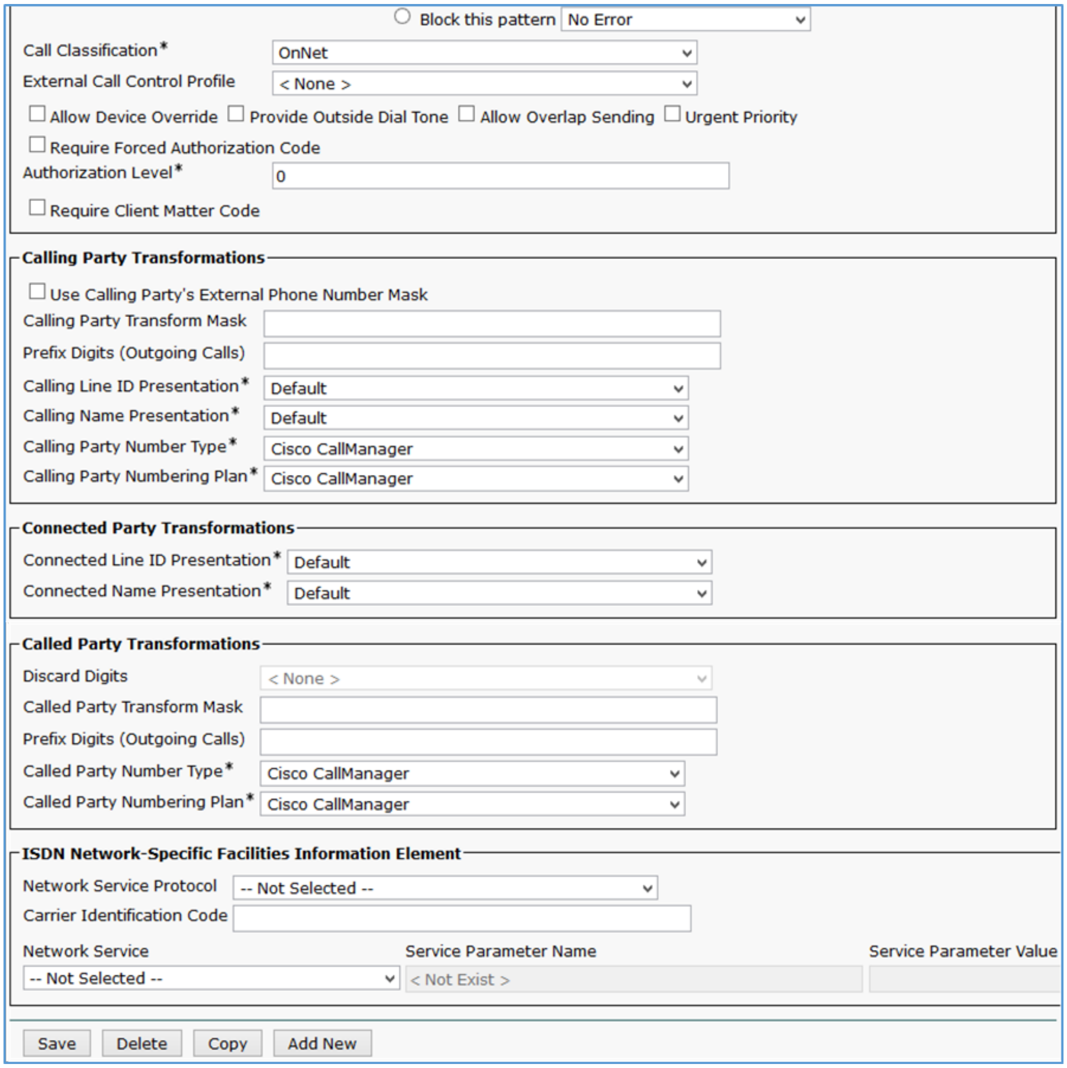
Configure the **2900** route pattern to route the voice mail pilot number (2900) to the Unity Connection server.

The screenshots that follow show the configuration.

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



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



Configure Voice Mail

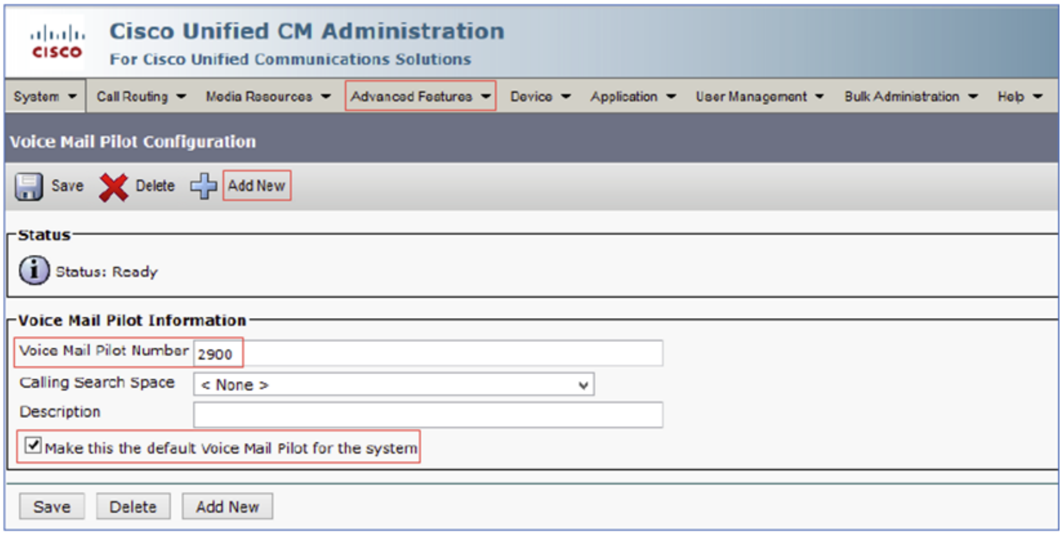
Perform a Cisco UCM - Cisco Unity Connection SIP integration to test voice mail scenarios.

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**.

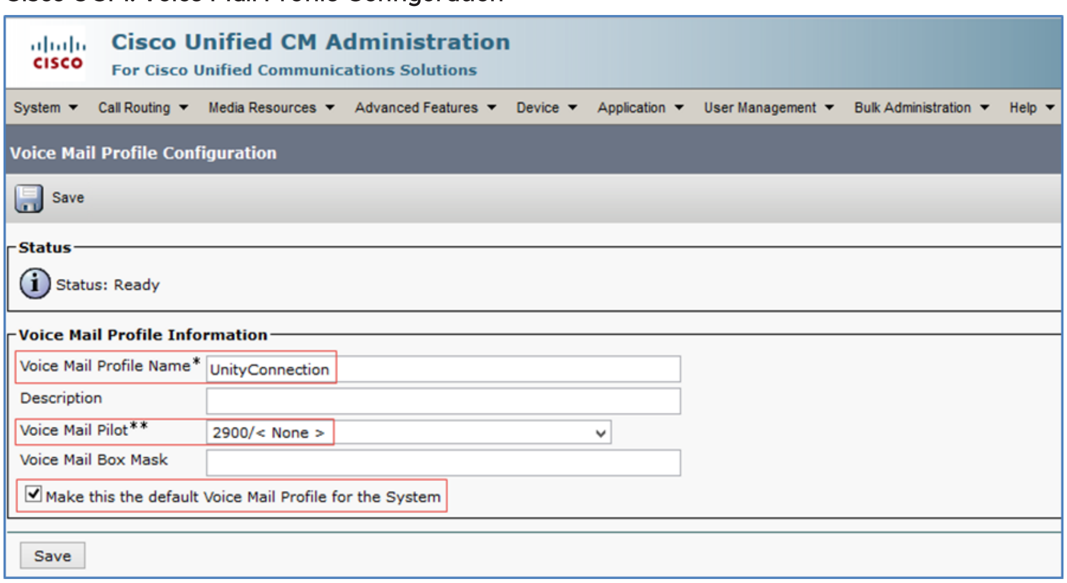
Cisco UCM: Voice Mail Pilot Number Configuration



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

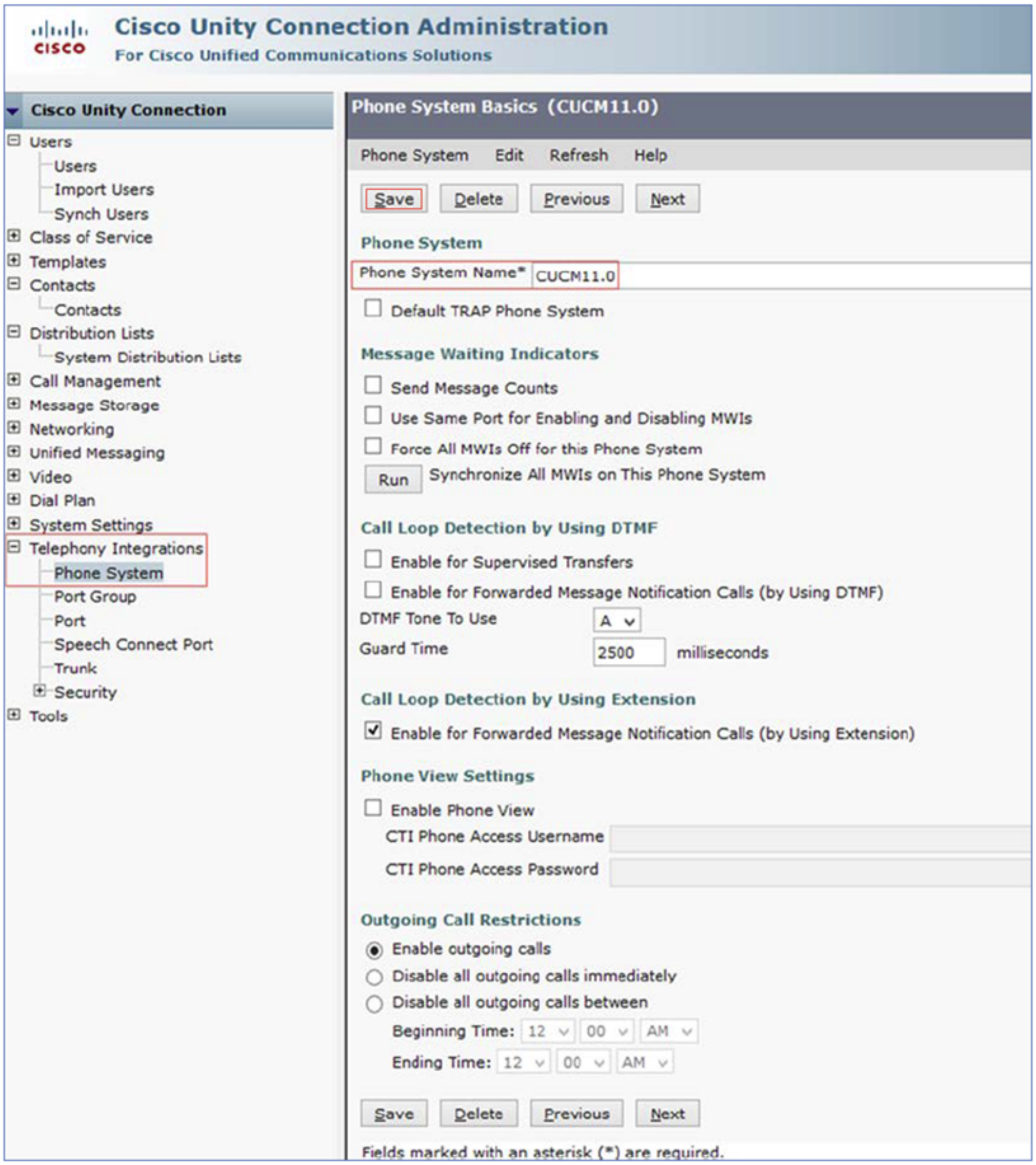


Configure New Phone System on Unity Connection

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

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

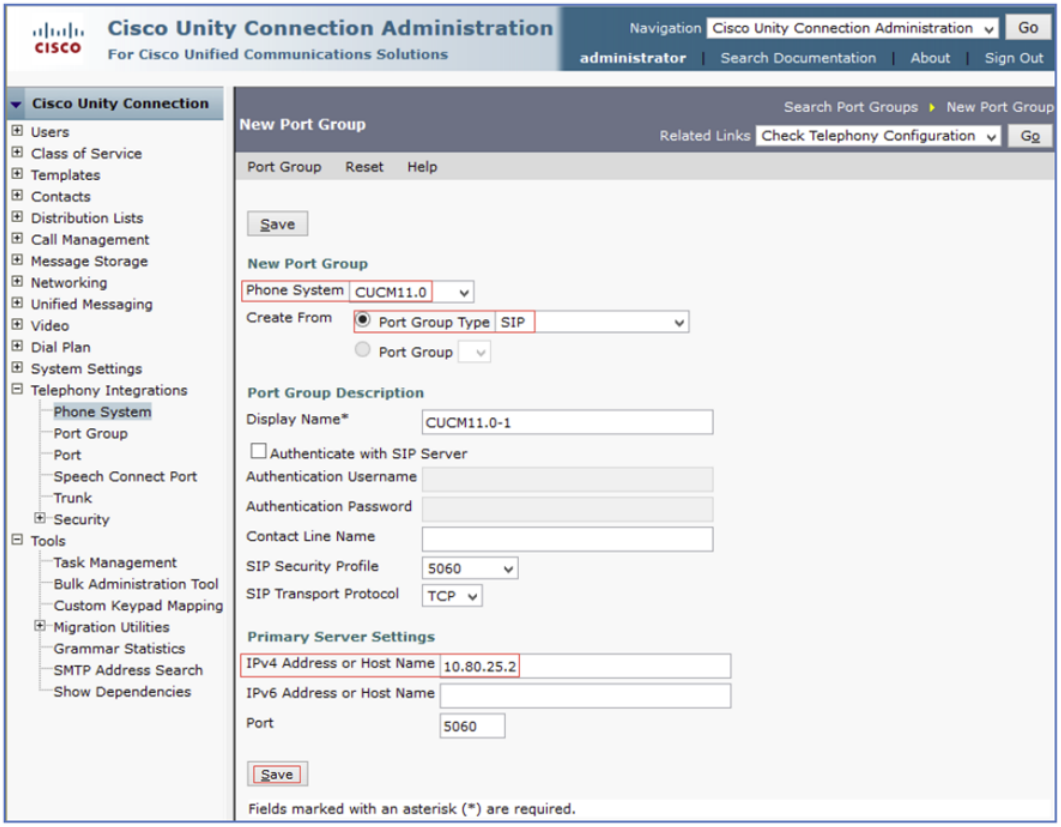
   Cisco Unity Connection: Phone System



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

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

Cisco Unity Connection: Add New Port Group

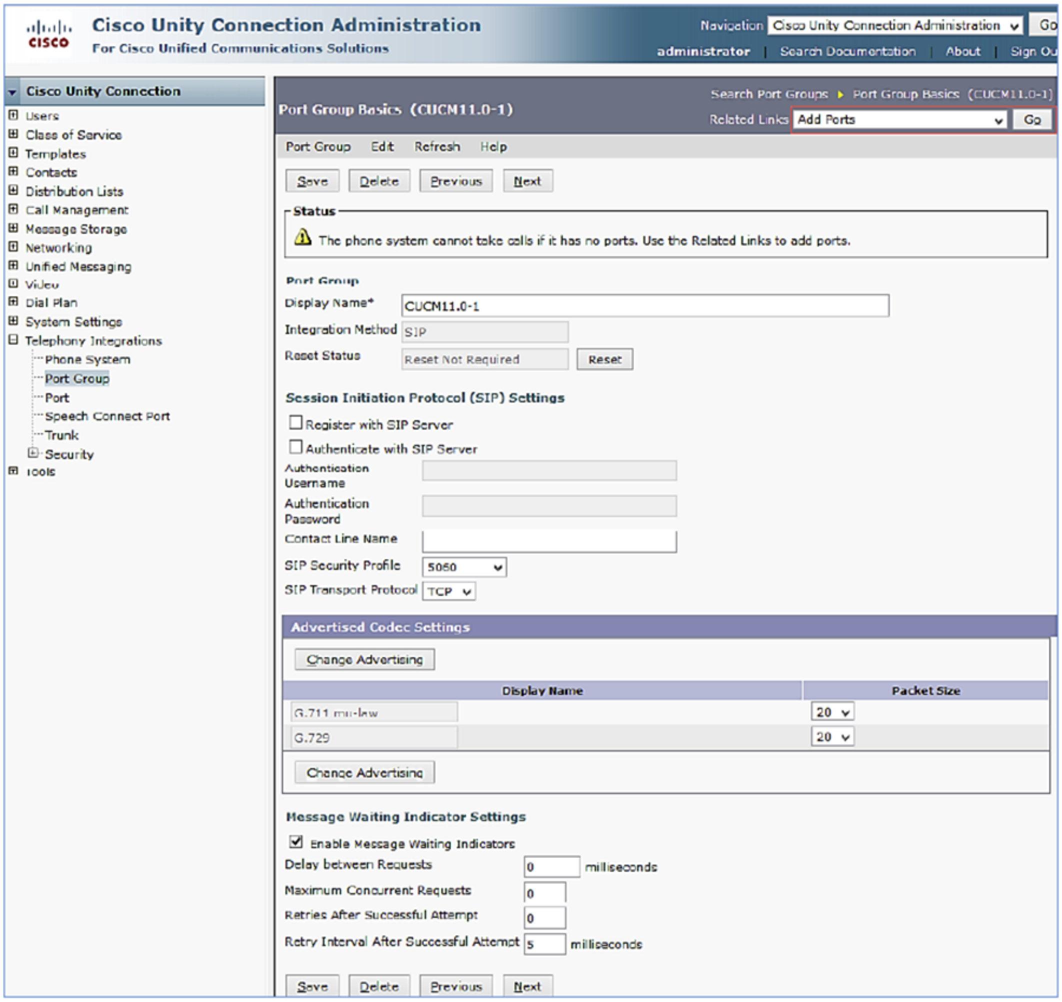


To add a new port group:

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**.

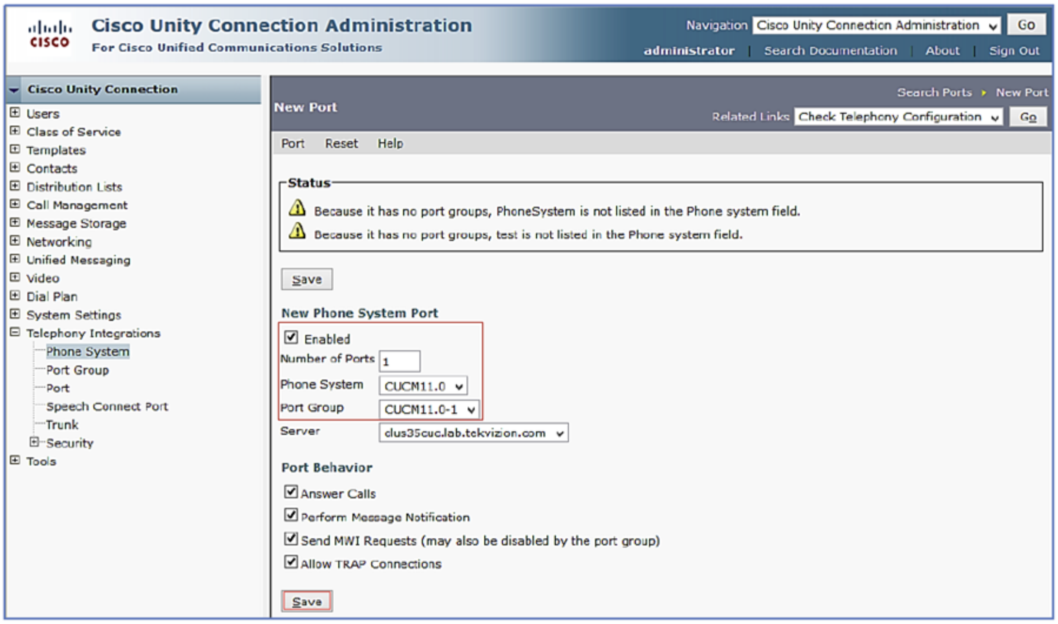
On the **Phone System Basics** page, in the **Related Link**s section, select **Add Ports** and then click **Go**.

Cisco Unity Connection: Related Links to Add Port



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

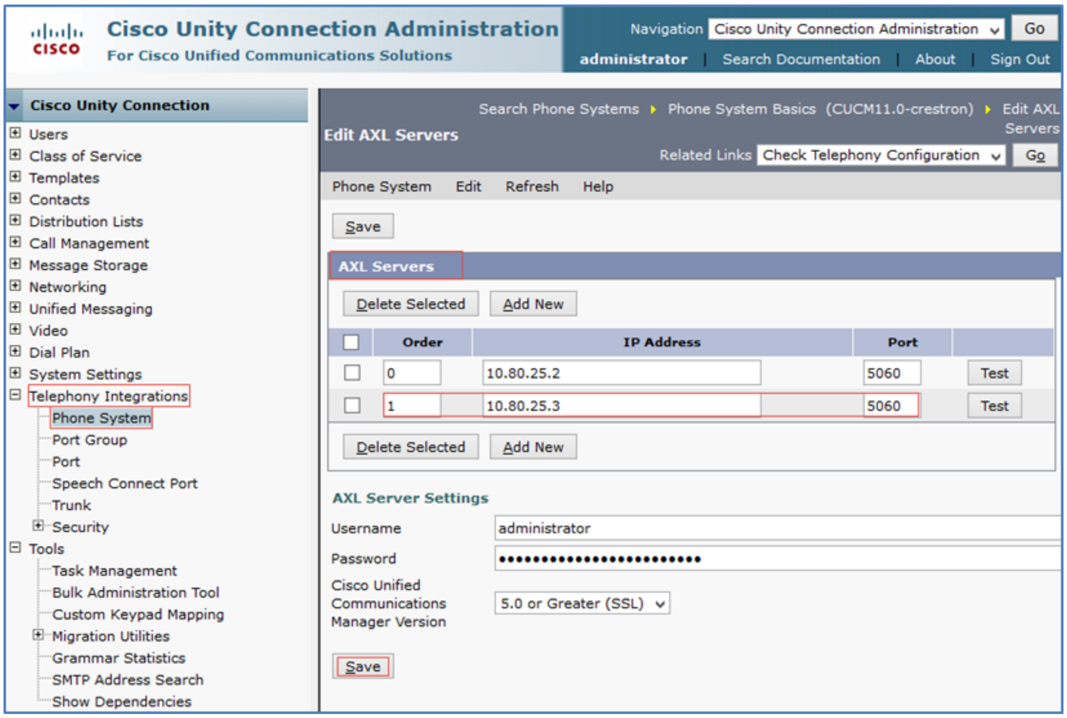
Cisco Unity Connection: Add New Port



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



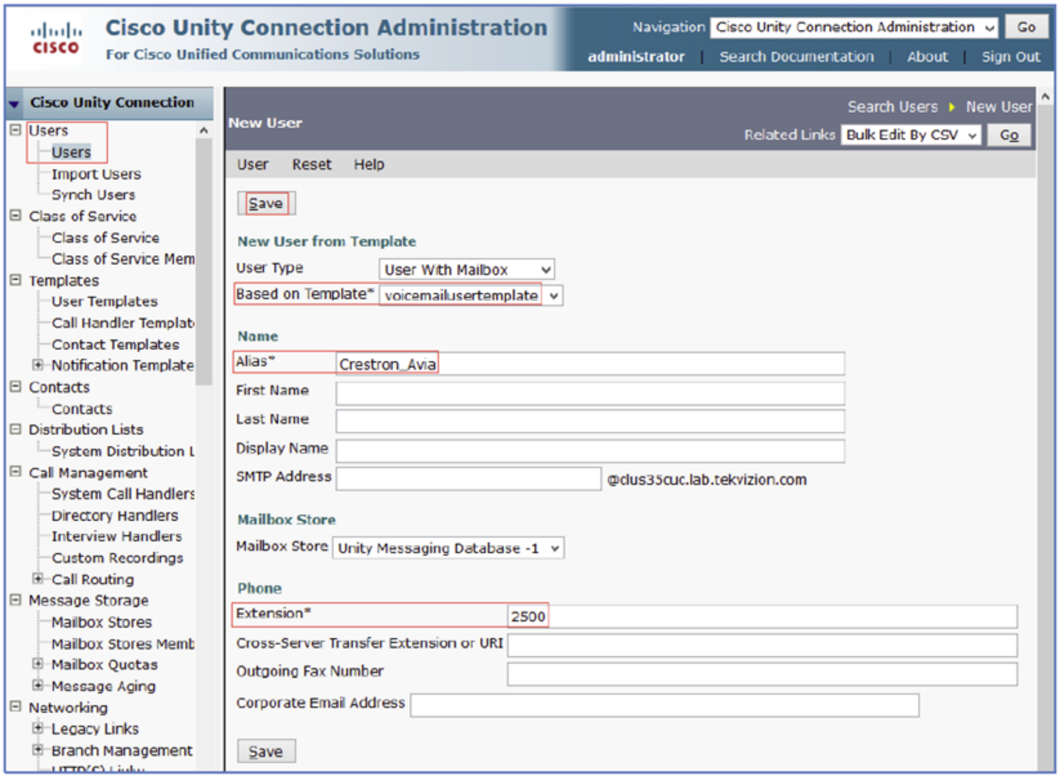
4. Enter the Cisco UCM subscriber IP Address and Port in the second row. This example uses **10.80.25.3** and **5060**, respectively.
5. Click **Save**.

Configure a Voice Mail User

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

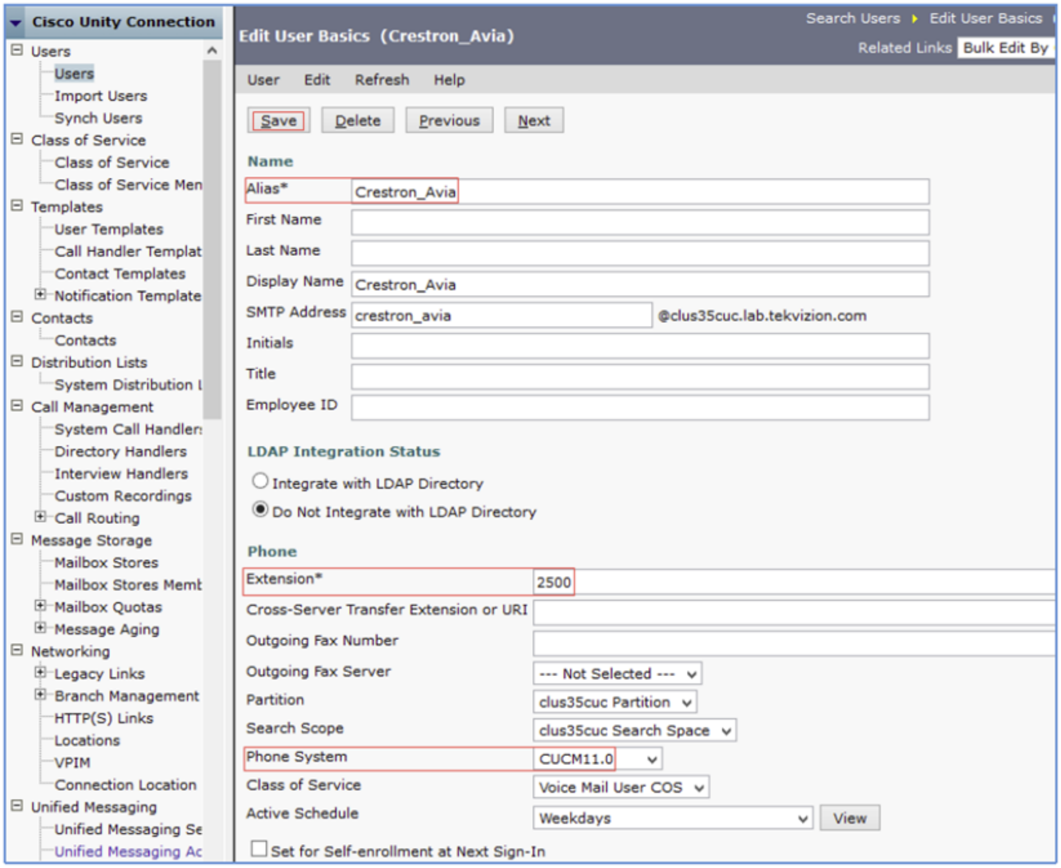
1. Click **Users** > **Users**.
2. Click **Add New**.

Cisco Unity Connection: Add User



3. Select **voicmailusertemplate** 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



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

Crestron Electronics, Inc.
15 Volvo Drive, Rockleigh, NJ 07647
Tel: 888.CRESTRON
Fax: 201.767.7656
www.crestron.com

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10.18
Specifications subject to
change without notice.