



Crestron Mercury® Tabletop Conference System  
(CCS-UC-1 & CCS-UC-1-X)

Non Secure SIP Endpoint with Cisco® 12.5 Unified  
Communication Manager (CUCM)

Configuration Guide

Prepared by tekVizion for Crestron Electronics, Inc.



## Original Instructions

The U.S. English version of this document is the original instructions.  
All other languages are a translation of the original instructions.

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## Revision History

Revision	Date	Author	Description
1.0	September 29, 2021	tekVizion	Initial Release

# Introduction

This configuration guide describes the necessary procedure to configure a Crestron Mercury® device to register to the Cisco® Unify Communication Manager (CUCM) as a Non Secure SIP Endpoint.

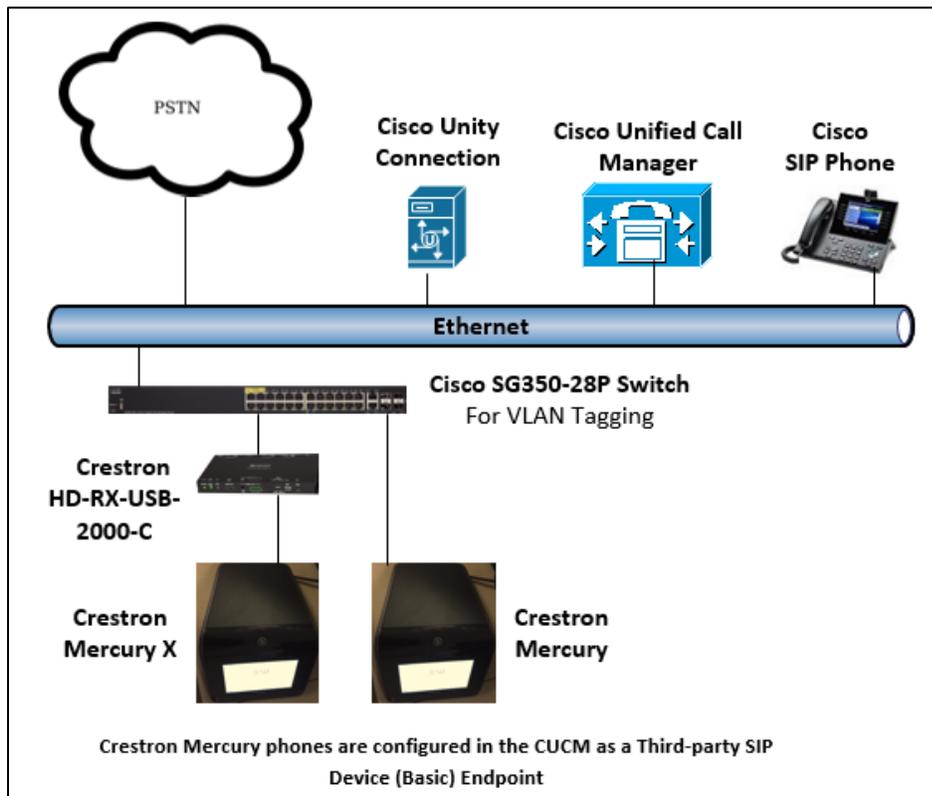
## Audience

This document is intended for users attempting to configure and use Crestron Mercury as Secure SIP Endpoints registering to Cisco CUCM 12.5.

## Topology

The network topology for the Crestron Mercury Endpoint to operate with Cisco CUCM is shown below.

### Crestron Mercury: SIP Endpoint Integration with CUCM: Reference Network



The lab network consists of the following components:

- Cisco Unified Communications Manager (Cisco CUCM) cluster for voice features
- Cisco Unity Connection as the voice mail system
- Cisco SIP phones
- Cisco SG350-28P Switch (For VLAN Tagging Configuration)
- Crestron Mercury and Crestron Mercury X as the SIP Endpoints
- Crestron HD-RX-USB-2000-C – used when connecting to the AUX Port on the Crestron Mercury X

## Software Requirements

- Cisco Unified Communication Manager v 12.5.1.12900-115
- Cisco Unity Connection v 12.5.1.12900-56
- Cisco SG350-28P v 2.4.5.71
- Crestron Mercury devices v 1.4736.00054
- Cisco SIP Phone v sip9971.9-4-2SR4-1

## Hardware Requirements

- Cisco UCS-C240-M3S VMWare Host running ESXi 5.5
- Cisco 3845 as PSTN Gateway
- Cisco SG350-28P (For VLAN Tagging Configuration)
- Cisco Phone : models - 9971 (SIP)
- Crestron Mercury CCS-US-1
- Crestron Mercury X CCS-US-1-EXT
- Crestron HD-RX-USB-2000-C – Needed when using the AUX Port on the Crestron Mercury X and also provides connections for front of the room displays

## 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® wireless presentation product for conference rooms.

Call dialing options on this device include Bluetooth® connectivity, USB and regular audio using a dial pad, though each dialing option is exclusive.

This device can be discovered via Crestron Toolbox™ software, though most of the configuration is performed via a local web interface. An Ethernet port on the device is used to provide power and network connectivity to make audio calls via SIP.

## Summary

The Crestron Mercury devices were configured on the Cisco CUCM as a Basic, Third-party SIP Device, endpoints since they support only a single line/extension. The devices successfully registered to the Cisco CUCM with digest authentication.

The Crestron Mercury CCS-UC-1 & CCS-UC-1-X phones in Non Secure mode are configured on the Cisco CUCM as a Basic, Third-party SIP Device, endpoints since they support only a single line/extension. The devices successfully registered to the Cisco CUCM with digest authentication.

The sections below describe the features that are supported/not supported and known issues/limitations on the Crestron Mercury phone.

## Features Supported

- VLAN Tagging
- Registration with Digest Authentication
- Basic Calls with G722, G729, G711u and G711a codecs
- DTMF Out-Of-Band and In-Band DTMF support
- Caller ID (limited to only Calling Number)
- Voice Mail access and interaction
- Early Media support
- Retrieval of a Parked Call
- Transferee in a Call Transfer
- Conference Call Participant
- Member of Shared Line configuration
- Member of a Hunt group

## Features Not Supported

- Caller ID Name presentation (Only the calling party number is displayed)
- Call Hold and Resume
- Call Forwarding on the device (Though forwarding can be configured on the PBX for the DN assigned to the endpoint)
- Call Waiting
- Initiating a Conference Call
- Initiating Attended Call Transfer
- Initiating Early Attended Call Transfer
- Initiating Blind Call Transfer
- Shared Line (configuration of shared line on Crestron Mercury device)
- Call Park (Initiating call park)
- DND (Do Not Disturb)
- Message Waiting Indicator

## Known Issues and Limitations

- None

# Crestron Mercury & Crestron Mercury X Configuration

## Crestron Mercury - Power

The LAN port of the Crestron Mercury device needs to be connected to one PoE+ port to power it up for network connectivity with the Cisco CUCM. The PoE switch should have 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. In the tekVizion™ lab environment, the Crestron Mercury phones are powered by an AC line universal power pack.

## Crestron Mercury X - Power

When using the Crestron Mercury X phone, an AC line universal power pack is needed to power the Crestron Mercury X.

## AUX Port on Crestron Mercury X

The AUX Port is used on the Crestron Mercury X phone. When using the AUX Port on the Crestron Mercury X phone, the HD-RX-USB-2000-C converter box is needed in line with the Ethernet connection.

## Discover/Access the Crestron Mercury

Crestron has a software tool available to discover and access the Crestron Mercury on the network: The Crestron Toolbox.

The Help menu on this tool assists the user through the discovery and configuration procedure.

The Crestron Mercury IP address, Host Name, MAC Address, Serial Number and Firmware Version can be viewed in the System info screen from the Home Screen by pressing and holding the Info link in the bottom left hand corner of the Crestron Mercury phone screen for 10 seconds.

### Crestron Mercury: System Info Screen

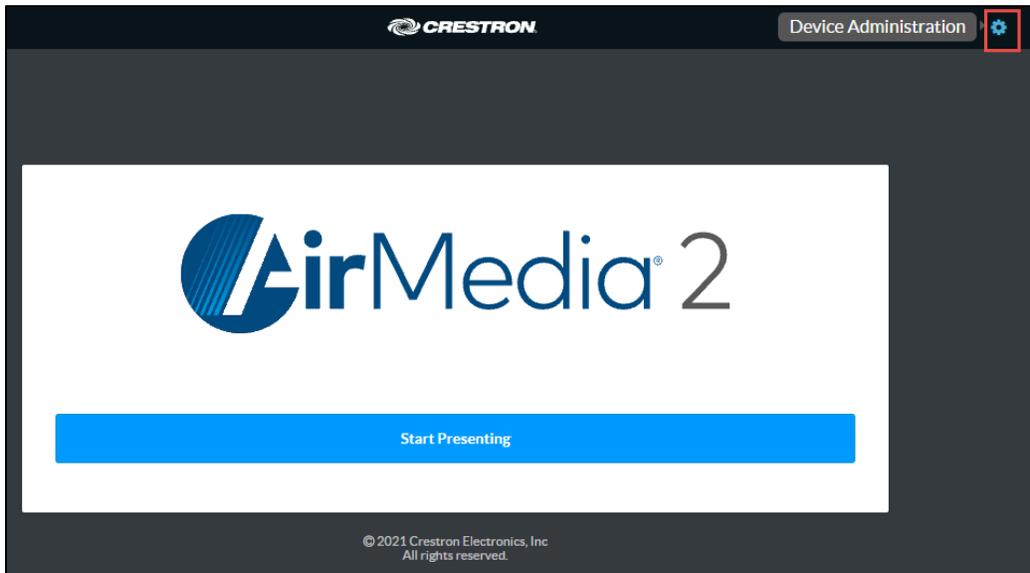


## Crestron Mercury Web UI Sign In

Access the Crestron Mercury Web UI for the device by using an http session with the device's IP address. The initial page that displays is shown below.

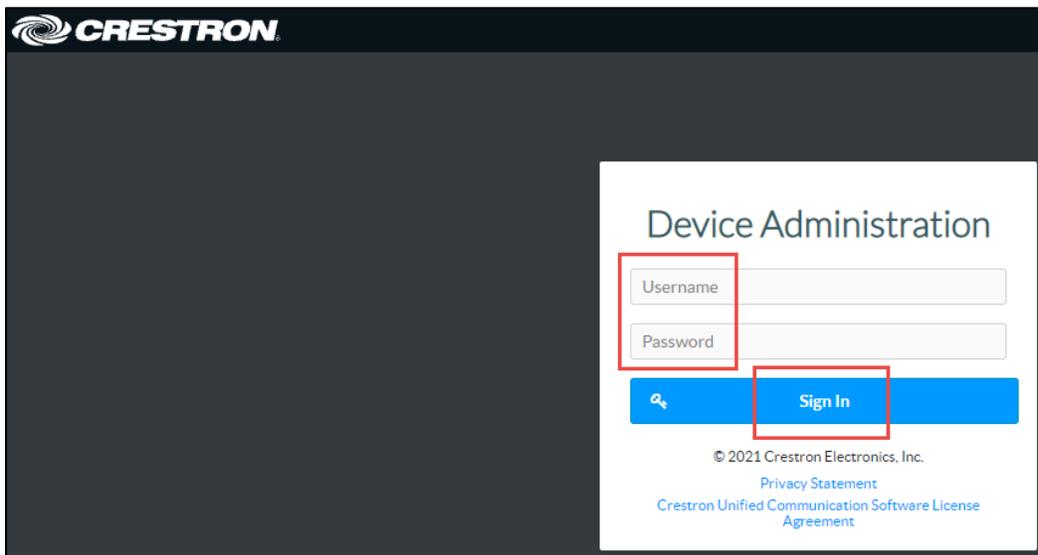
- Select the **Device Administration** link in top right corner.

### Crestron Mercury: Device Administration



1. In the pop-up window provide **login credentials**.
2. Default Crestron Mercury Login credentials are **admin/admin**.
3. Click **Sign In**.

### Crestron Mercury Web UI: Sign In



## Crestron Mercury

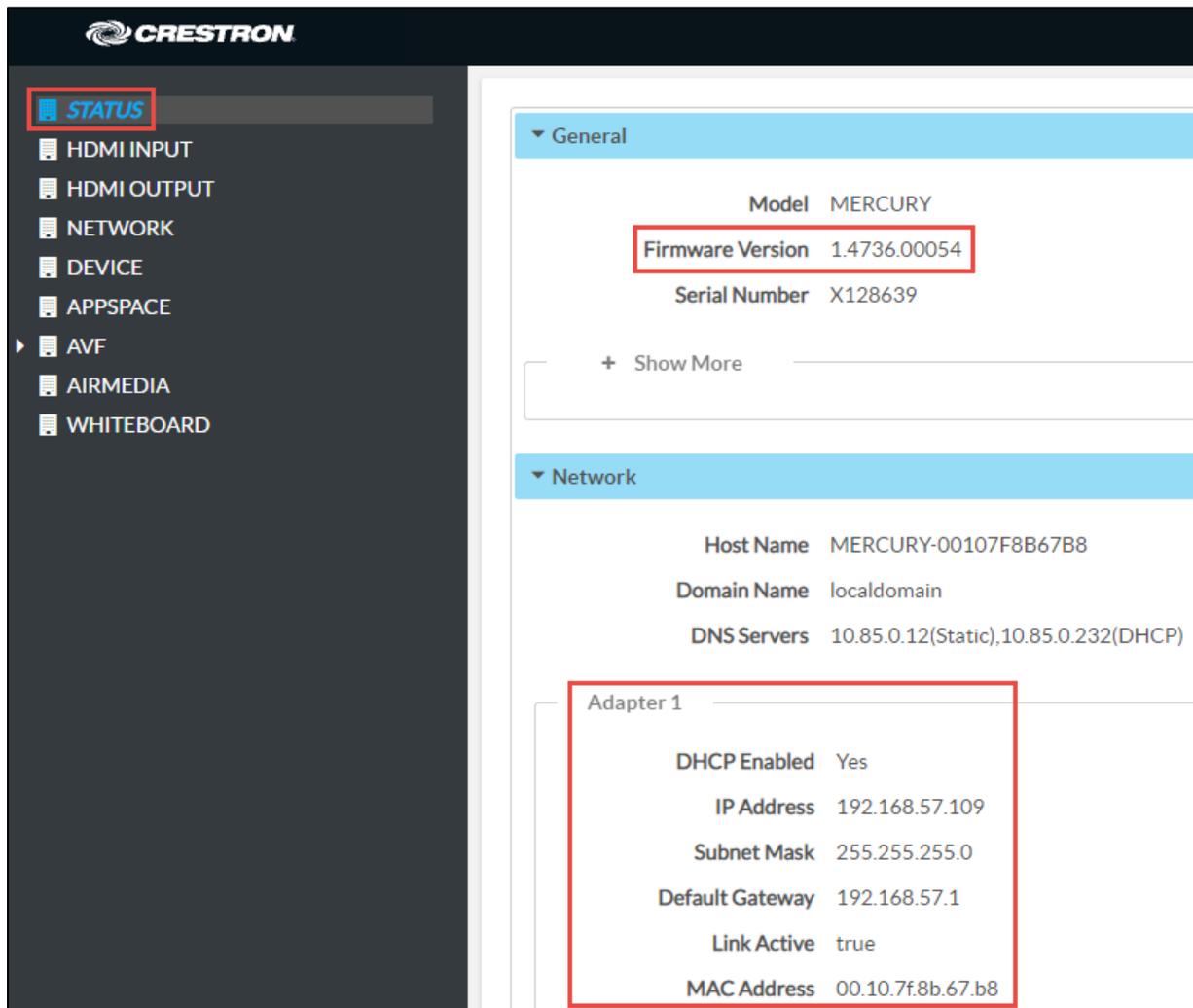
In the tekVizion lab environment, one DUT used as a Crestron Mercury phone with the Ethernet cable connected to the LAN port of the Crestron Mercury. Configuration for this setup is shown below.

### Status

The **Status** screen shown below displays basic device information:

- The **Firmware Version** and **Network** info of the Crestron Mercury are shown here.

#### Crestron Mercury: Status



The screenshot shows the Crestron Mercury web interface. On the left is a navigation menu with 'STATUS' highlighted. The main content area is divided into two sections: 'General' and 'Network'. In the 'General' section, 'Firmware Version' is 1.4736.00054. In the 'Network' section, 'Adapter 1' is shown with DHCP Enabled (Yes), IP Address (192.168.57.109), Subnet Mask (255.255.255.0), Default Gateway (192.168.57.1), Link Active (true), and MAC Address (00.10.7f.8b.67.b8).

Section	Property	Value
General	Model	MERCURY
	Firmware Version	1.4736.00054
	Serial Number	X128639
Network	Host Name	MERCURY-00107F8B67B8
	Domain Name	localdomain
	DNS Servers	10.85.0.12(Static),10.85.0.232(DHCP)
	Adapter 1	
	DHCP Enabled	Yes
	IP Address	192.168.57.109
	Subnet Mask	255.255.255.0
Default Gateway	192.168.57.1	
Link Active	true	
MAC Address	00.10.7f.8b.67.b8	

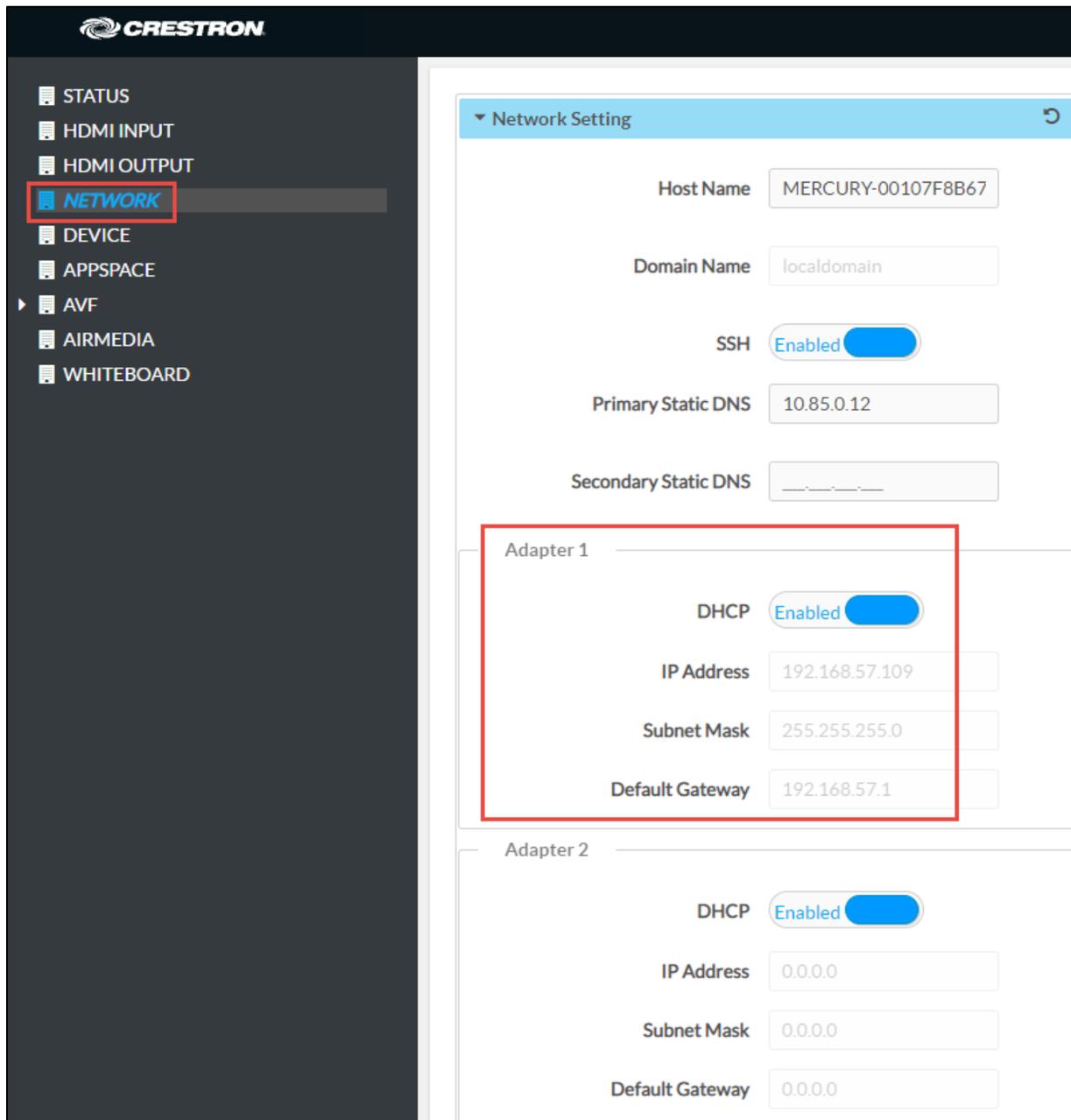
## Network Configuration

The Crestron Mercury Network settings can be configured from the Network page.

On the Crestron Mercury Web UI, navigate to **Network**.

1. **DHCP:** The Crestron Mercury is configured as DHCP.
2. The LAN Port is used on the Crestron Mercury, so **Adapter 1** is configured via DHCP.
3. Click **Save Changes**.

### Crestron Mercury: Network: DHCP



The screenshot displays the Crestron Mercury Network Configuration interface. On the left, a navigation menu lists various system settings, with **NETWORK** highlighted in a red box. The main panel shows the **Network Setting** configuration page. Key settings include:

- Host Name:** MERCURY-00107F8B67
- Domain Name:** localdomain
- SSH:** Enabled (toggle switch)
- Primary Static DNS:** 10.85.0.12
- Secondary Static DNS:** (empty field)

Below these settings, the configuration for **Adapter 1** is shown, highlighted with a red box:

- DHCP:** Enabled (toggle switch)
- IP Address:** 192.168.57.109
- Subnet Mask:** 255.255.255.0
- Default Gateway:** 192.168.57.1

The **Adapter 2** configuration is also visible below:

- DHCP:** Enabled (toggle switch)
- IP Address:** 0.0.0.0
- Subnet Mask:** 0.0.0.0
- Default Gateway:** 0.0.0.0

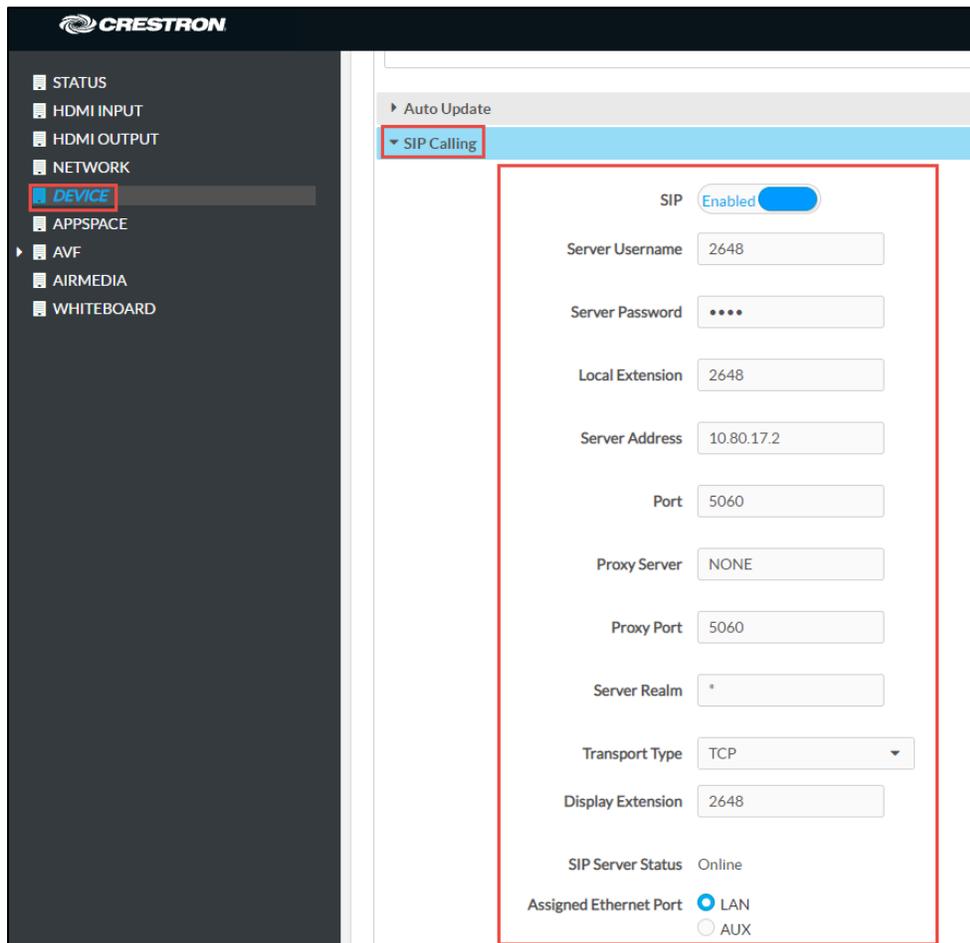
## SIP Calling Parameters

Configure the Crestron Mercury SIP Parameters to enable Crestron Mercury communication with the Cisco CUCM.

From the Crestron Mercury Web UI, navigate to **Device → SIP Calling**.

1. **SIP:** click the box to display **Enabled**.
2. **Server Username:** Enter the end user configured on the Cisco CUCM for this device, **(2648)**.
3. **Server Password:** User's password as configured on the Cisco CUCM.
4. **Local Extension:** Enter the directory number configured on the Cisco CUCM, **(2648)**.
5. **Server Address:** Enter the IP address of the Cisco CUCM, **(10.80.17.2)**.
6. **Port:** For the Non Secure TCP setup port **5060** is used.
7. **Transport Type:** For the Non Secure setup, **TCP** Transport is used.
8. **Display Extension:** **2648** is used.
9. **Assigned Ethernet Port** is set to **LAN**.
10. Click **Save Changes**.
11. **SIP Server Status** shows **Online** when successfully registered to the PBX.

### Crestron Mercury: Device: SIP Calling



The screenshot displays the Crestron Mercury Web UI configuration page for SIP Calling. The left sidebar shows the navigation menu with 'DEVICE' selected. The main content area is titled 'SIP Calling' and contains the following configuration fields:

- SIP:** Enabled (toggle)
- Server Username:** 2648
- Server Password:** Masked with dots
- Local Extension:** 2648
- Server Address:** 10.80.17.2
- Port:** 5060
- Proxy Server:** NONE
- Proxy Port:** 5060
- Server Realm:** \*
- Transport Type:** TCP (dropdown)
- Display Extension:** 2648
- SIP Server Status:** Online
- Assigned Ethernet Port:** LAN (radio button selected), AUX (radio button unselected)

## Crestron Mercury X

In the tekVizion lab environment, one DUT is a Crestron Mercury X phone with the Ethernet cable connected to the AUX port. The Crestron HD-RX-USB-2000-C converter box is needed in-line with the Ethernet connection when the AUX port is used. The specific Crestron Mercury X configuration for this setup is shown below, the rest of the configuration is the same as the above Crestron Mercury configuration.

## Network Configuration

The Crestron Mercury Network settings can be configured from the Network page.

On the Crestron Mercury Web UI, navigate to **Network**.

1. **DHCP:** The Crestron Mercury is configured as DHCP. The AUX Port is used on the Crestron Mercury X, so **Adapter 2** is configured as **DHCP**.
2. Click **Save Changes**.

### Crestron Mercury X: Network

**CRESTRON**

- STATUS
- HDMI INPUT
- HDBT OUTPUT
- NETWORK**
- DEVICE
- APPSPACE
- AVF
- AIRMEDIA
- WHITEBOARD

**Network Setting**

Host Name: MERCURY-X-00107FCFI

Domain Name: localdomain

SSH: Enabled

Primary Static DNS: 10.85.0.12

Secondary Static DNS: \_\_\_\_\_

**Adapter 1**

DHCP: Enabled

IP Address: 0.0.0.0

Subnet Mask: 0.0.0.0

Default Gateway: 0.0.0.0

**Adapter 2**

DHCP: Enabled

IP Address: 192.168.57.141

Subnet Mask: 255.255.255.0

Default Gateway: 192.168.57.1

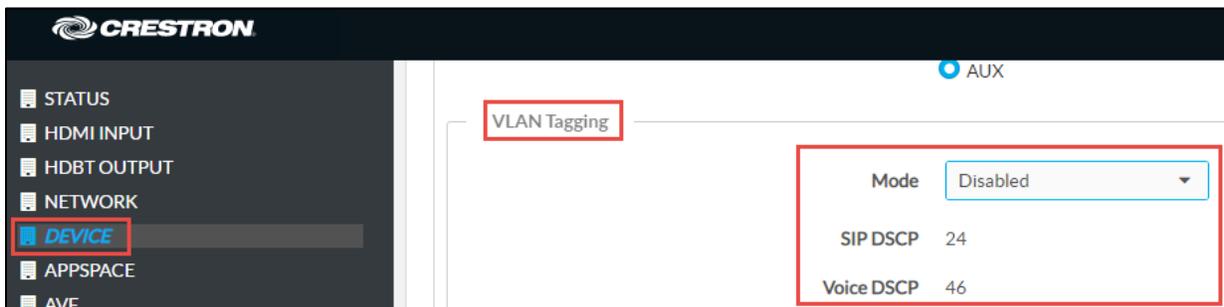
## VLAN Tagging

VLAN Tagging on the Crestron Mercury allows you to assign DSCP values to the SIP and Media messages. It also allows you to assign a Priority value to the VLAN used for the SIP and Media messages. When enabled, VLAN Tagging uses a 2nd IP address that is assigned to the Crestron Mercury phone for the SIP and Media messages. The IP address is assigned by a Local Network Cisco switch (Cisco SG350-28P), providing the VLAN Tagging configuration info to the Crestron Mercury.

The available **VLAN Tagging Mode** settings are shown below: From the Crestron Mercury Web UI, navigate to **Device → SIP Calling**.

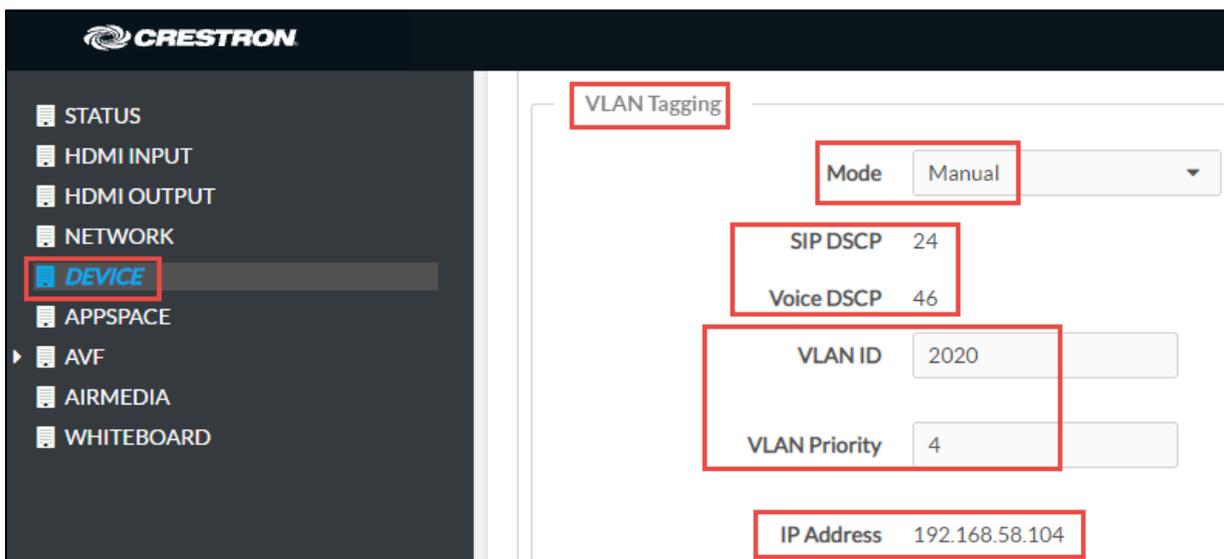
1. **Disabled** – Uses just 1 IP address for the Data IP address SIP and Media. The default DSCP value assigned to SIP is **24** and to Media is **46**. The Priority VLAN value is not assigned to the Messages. The default Crestron Mercury setting is **Disabled**.

### Crestron Mercury: Device: SIP Calling: VLAN Tagging - Disabled



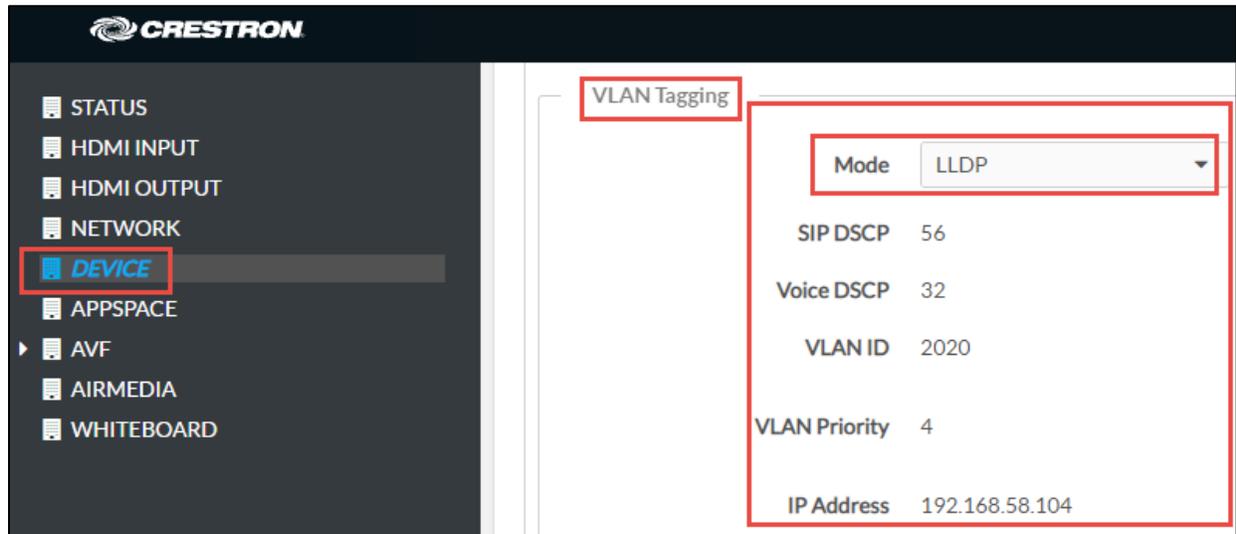
2. **Manual** – Allows you to assign the VLAN ID and VLAN priority to be used by the Crestron Mercury. The default DSCP values (**SIP – 24** and **Voice – 46**) are assigned. The 2<sup>nd</sup> IP address, used for SIP and Media is assigned to the Crestron Mercury by the local network switch with the VLAN Tagging configuration.

### Crestron Mercury: Device: SIP Calling: VLAN Tagging - Manual



3. **LLDP** – Pulls the VLAN Tagging information from the local network switch with the VLAN Tagging configuration.

**Crestron Mercury: Device: SIP Calling: VLAN Tagging - LLDP**



VLAN Tagging	
Mode	LLDP
SIP DSCP	56
Voice DSCP	32
VLAN ID	2020
VLAN Priority	4
IP Address	192.168.58.104

## VLAN Tagging Local Network Switch – Cisco SG350-28P

The tekVizion lab environment used a Cisco SG350-28P switch to provide the 2<sup>nd</sup> IP address used for SIP & Media, and the VLAN Tagging configuration for the Crestron Mercury and Crestron Mercury X phone when **LLDP** is set as the **Mode** for the Crestron Mercury.

The Crestron Mercury phones are connected directly to the Cisco SG350-28P switch in the lab setup.

The Running Configuration for the VLAN Tagging switch is provided below. The following configuration settings are used in the tekVizion lab VLAN Tagging environment.

1. **Voice Vlan ID 2020**
2. **LLDP Med Network-Policy**
  - 3 voice-signaling vlan 2020 vlan-type tagged up 4
  - 4 voice vlan 2020 vlan-type tagged up 4 dscp 32
  - 9 voice-signaling vlan 2020 vlan-type tagged up 4 dscp 32
  - 10 voice vlan 2020 vlan-type tagged up 4 dscp 32
  - 15 voice-signaling vlan 2020 vlan-type tagged up 4 dscp 56
  - 16 voice vlan 2020 vlan-type tagged up 4 dscp 32
3. **Interface Port 4 - Crestron Mercury phone**
  - interface GigabitEthernet4
  - description Crestron Mercury2
  - switchport mode trunk
  - lldp med network-policy add 15
  - lldp med network-policy add 16
4. **Interface Port 7 - Crestron Mercury phone**
  - interface GigabitEthernet7
  - description Crestron Mercury 5
  - switchport mode trunk
  - lldp med network-policy add 15
  - lldp med network-policy add 16

## Cisco SG350\_28P – Running Configuration

```
switch94214e#show run
config-file-header
switch94214e
v2.4.5.71 / RTESLA2.4.5_930_181_144
CLI v1.0
file SSD indicator encrypted
@
ssid-control-start
ssid config
ssid file passphrase control unrestricted
no ssid file integrity control
ssid-control-end cb0a3fdb1f3a1af4e4430033719968c0
!
!
unit-type-control-start
unit-type unit 1 network gi uplink none
unit-type-control-end
!
vlan database
vlan 2,10-11,15,200,2018-2020,4030
exit
voice vlan id 2020
voice vlan oui-table add 0001e3 Siemens_AG_phone_____
voice vlan oui-table add 00036b Cisco_phone_____
voice vlan oui-table add 00096e Avaya _____
voice vlan oui-table add 000fe2 H3C_Aolynk_____
voice vlan oui-table add 0060b9 Philips_and_NEC_AG_phone
voice vlan oui-table add 00d01e Pingtel_phone_____
voice vlan oui-table add 00e075 Polycom/Veritel_phone___
voice vlan oui-table add 00e0bb 3Com_phone_____
no lldp med network-policy voice auto

lldp med network-policy 3 voice-signaling vlan 2020 vlan-type tagged up 4
lldp med network-policy 4 voice vlan 2020 vlan-type tagged up 4 dscp 32
```

```
Ildp med network-policy 9 voice-signaling vlan 2020 vlan-type tagged up 4 dscp 32
```

```
Ildp med network-policy 10 voice vlan 2020 vlan-type tagged up 4 dscp 32
```

```
Ildp med network-policy 15 voice-signaling vlan 2020 vlan-type tagged up 4 dscp 56
```

```
Ildp med network-policy 16 voice vlan 2020 vlan-type tagged up 4 dscp 32
```

```
link-flap prevention disable
```

```
bonjour interface range vlan 1
```

```
hostname switch94214e
```

```
no passwords complexity enable
```

```
ip ssh server
```

```
ip telnet server
```

```
!
```

```
interface vlan 2
```

```
name Data
```

```
!
```

```
interface vlan 15
```

```
name "RSPAN VLAN"
```

```
remote-span
```

```
!
```

```
interface GigabitEthernet1
```

```
description PoE1
```

```
storm-control broadcast level 10
```

```
storm-control multicast level 10
```

```
port security max 10
```

```
port security mode max-addresses
```

```
port security discard trap 60
```

```
spanning-tree portfast
```

```
spanning-tree bpduguard enable
```

```
switchport mode trunk
```

```
switchport trunk allowed vlan remove 2-2019,2021-4094
```

```
macro description "ip_phone_desktop_1 | no_ip_phone_desktop  
ip_phone_desktop" |
```

```
no macro auto smartport
```

```
macro auto smartport type ip_phone_desktop
```

```
!
```



```
interface GigabitEthernet2
description PoE2
storm-control broadcast level 10
storm-control multicast level 10
port security max 10
port security mode max-addresses
port security discard trap 60
spanning-tree portfast
spanning-tree bpduguard enable
switchport mode trunk
switchport trunk allowed vlan remove 2-2019,2021-4094
macro description "ip_phone_desktop_2 | no_ip_phone_desktop"
ip_phone_desktop
macro auto smartport type ip_phone_desktop
!
interface GigabitEthernet3
description Crestron Mercury1
switchport mode trunk
lldp med network-policy add 15
lldp med network-policy add 16
!
interface GigabitEthernet4
description Crestron Mercury2
switchport mode trunk
lldp med network-policy add 15
lldp med network-policy add 16
!
interface GigabitEthernet5
shutdown
description Crestron Mercury3
switchport mode trunk
!
interface GigabitEthernet6
description Crestron Mercury4
switchport mode trunk
lldp med network-policy add 3
```



```
lldp med network-policy add 4
!
interface GigabitEthernet7
description Crestron Mercury5
switchport mode trunk
lldp med network-policy add 15
lldp med network-policy add 16
!
interface GigabitEthernet13
description PoE3
storm-control broadcast level 10
storm-control multicast level 10
port security max 10
port security mode max-addresses
port security discard trap 60
spanning-tree portfast
spanning-tree bpduguard enable
switchport mode trunk
switchport trunk allowed vlan remove 2-2019,2021-4094
macro description "ip_phone_desktop_3 | no_ip_phone_desktop
ip_phone_desktop"
macro auto smartport type ip_phone_desktop
!
interface GigabitEthernet14
description PoE4
storm-control broadcast level 10
storm-control multicast level 10
port security max 10
port security mode max-addresses
port security discard trap 60
spanning-tree portfast
spanning-tree bpduguard enable
switchport mode trunk
switchport trunk allowed vlan remove 2-2019,2021-4094
macro description "ip_phone_desktop_4 | no_ip_phone_desktop
ip_phone_desktop"
!next command is internal.
```



```
macro auto smartport dynamic_type ip_phone_desktop
!
interface GigabitEthernet24
description Wireshark
bridge multicast unregistered filtering
switchport trunk native vlan none
ip igmp version 2
ip igmp query-interval 60
!
interface GigabitEthernet25
description DHCP
spanning-tree link-type point-to-point
switchport mode trunk
macro description switch
!
interface GigabitEthernet26
shutdown
description dhcp1
spanning-tree link-type point-to-point
switchport mode trunk
!
exit
monitor session 1 destination remote vlan 15 reflector-port GigabitEthernet24 network
monitor session 1 source interface GigabitEthernet4 both
monitor session 1 source interface GigabitEthernet7 both
monitor session 1 source interface GigabitEthernet13 both
monitor session 1 source interface GigabitEthernet14 both
```

## Crestron Mercury & Crestron Mercury X - RFC 2833 Support

To configure the RFC 2833 support on the Crestron Mercury, the **Sipaudiomode RFC2833** command is used from the Crestron Mercury CLI and accessed from the Crestron Toolbox. There are 2 options: **ON** or **OFF**.

1. **ON (TRUE)**: Considered Out-of-band, RTP DTMF Events are viewable in the RTP stream. This is the Default setting.
2. **OFF (FALSE)**: Considered In-band, RTP DTMF Events are not viewable in the RTP Stream.

### SipAudioMode RFC2833 On

**Sipaudiomode RFC2833 on** command is used to enable RFC2833 Out-of-band support. The RFC2833 setting can be viewed from the **Sipstate** command.

#### Crestron Mercury X: CLI: RFC 2833 Support

```
MERCURY>sipaudiomode rfc2833 on
RFC2833 support has been turned on.

MERCURY>sipstate

Current SIP States
-----
Server registered      = TRUE
Door station mode     = FALSE
Call in progress      = FALSE
Call hold              = FALSE
Push-To_Talk          = FALSE
Do not disturb         = FALSE
Video started         = FALSE
Video blocked         = FALSE
Video can show        = FALSE
Default ringer        = TRUE
Ring state            = FALSE
Ringback state        = FALSE
Group call flag       = FALSE
User Mute state       = FALSE
Local Mute state      = FALSE
Multicast flag        = FALSE
Support answer        = FALSE
Request auto          = FALSE
Request urgent        = FALSE
RFC 2833 support      = TRUE
Call timeout          = 120 (secs)
Answer timeout        = 0 (secs)
Rewrite CONTACT       = TRUE
Rewrite SDP           = FALSE
Rewrite VIA           = TRUE
Voice-AutoListen      = FALSE
Sound device          = not active
SIP DSCP codepoint    = 56
RTP DSCP codepoint    = 32
Verify server         = FALSE
Verify client         = FALSE
SRTP                  = mandatory
Session Timer         = optional
Early Media           = auto
Video Enable          = auto
Invite Response       = 183
Interface             = LAN_SIPVLAN
Reg Timeout           = 300
```

## Crestron Mercury & Crestron Mercury X - SIP Interface Port

To configure the Crestron Mercury X Assigned Ethernet Port to use the LAN or RX OUT Ethernet ports, use the SIPINTERFACE CLI command. When the HD-RX-USB-2000-C Receiver is connected to the Crestron Mercury X, the **AUX** (RX OUT) port is used.

The **Assigned Ethernet Port** can also be configured from the Crestron Mercury Web UI, in the **SIP Calling** section.

### SipInterFace AUX

**Sipinterface AUX** is used in the Crestron Mercury X CLI to activate the RX OUT Ethernet port as the SIP Interface port to be used. Using the RX OUT Ethernet port allows the internet connection to be routed through the HD-RX-USB-2000-C receiver and then connected to the RX OUT port on the Crestron Mercury X.

#### Crestron Mercury X: CLI: SIPINTERFACE Support

```
MERCURY-X>sipinterface aux
Success: New SIP interface has been set.

MERCURY-X>sipinfo
SIP Parameters
-----
SIP: ENABLED
-----
SIP audio mode: FD
SIP auto mode: NONE
SIP local ext: 2645
SIP local name: CRESTRON
SIP local port: 5060
SIP connection mode: SERVER
SIP page group(s): CRESTRON
SIP realm: *
SIP remote config file: NONE
SIP server name: NONE
SIP server port: 5060
SIP server ip address: 10.80.17.2
SIP server username: 2645
SIP server password: ****
SIP Name server: NONE
SIP proxy server: NONE:5060
SIP STUN server: NONE
SIP STUN domain: NONE
SIP multicast address: 227.1.1.1
SIP multicast port: 1234
SIP transport type: TCP
SIP protocol qos: 24
SIP media port: 40000
SIP rtp qos: 46
SIP session timer: optional
SIP Interface: AUX
SIP registration timeout: 300
```

# Cisco Unified Communications Manager (CUCM)

This section describes the Cisco CUCM configuration necessary to integrate the Crestron Mercury and Crestron Mercury X as a SIP Endpoint.

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

## User Configuration

1. Navigate to **User Management** -> **End User**.
2. Click **Add New**. The End User configuration window appears.
3. **User ID**: Enter a unique end user identification name. Two users were configured for this test: **2645** (*Crestron Mercury X*) and **2648** (*Crestron Mercury*).
4. **Last Name**: Enter the end user last name, **Mercury X**.
5. **Digest Credentials**: This same password will be entered on the Crestron Mercury device for the SIP Server Password. The extension number (**2645 & 2648**) is used for the Password.
6. **Confirm the Digest Credentials**: Re-enter the password configured above.
7. **Password**: the **Digest Credentials** were also used for the **Password**.
8. **Confirm Password**: Re-enter the same password configured above.
9. Click **Save**.

### Cisco CUCM: End User Configuration

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

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

#### End User Configuration

Save Delete Add New

**Status**  
 Status: Ready

**User Information**

User Status	Enabled Local User	
User ID*	<input type="text" value="2645"/>	
Password	<input type="password" value="....."/>	<input type="button" value="Edit Credential"/>
Confirm Password	<input type="password" value="....."/>	
Self-Service User ID	<input type="text" value="2645"/>	
PIN	<input type="password" value="....."/>	<input type="button" value="Edit Credential"/>
Confirm PIN	<input type="password" value="....."/>	
Last name*	<input type="text" value="Mercury X"/>	
Middle name	<input type="text"/>	
First name	<input type="text"/>	
Display name	<input type="text"/>	
Title	<input type="text"/>	
Directory URI	<input type="text"/>	
Telephone Number	<input type="text"/>	
Home Number	<input type="text"/>	
Mobile Number	<input type="text"/>	
Pager Number	<input type="text"/>	
Mail ID	<input type="text"/>	
Manager User ID	<input type="text"/>	
Department	<input type="text"/>	
User Locale	<input type="text" value="&lt; None &gt;"/>	
Associated PC/Site Code	<input type="text"/>	
Digest Credentials	<input type="password" value="....."/>	
Confirm Digest Credentials	<input type="password" value="....."/>	
User Profile	<input type="text" value="Use System Default( 'Standard (Factory Default) Us"/>	<a href="#">View Details</a>
User Rank*	<input type="text" value="1-Default User Rank"/>	

## SIP Profile

SIP Profile is configured for the Crestron Mercury and Crestron Mercury X phones. The Standard SIP Profile is used for the Cisco PBX phone.

### Crestron Standard SIP Profile – Crestron Mercury phones

1. To add a new SIP Profile, Navigate to **Device -> Device Settings-> SIP Profile**.
2. On the screen that appears, click **Add New** and configure the SIP Profile as below.
3. **Name: Crestron Standard SIP Profile**.
4. Configure **Early offer support for voice and video calls \*** as **Best Effort(no MTP inserted)**
5. Retain all other default configuration settings.
6. Then click **Save** and then **Apply Config**.

#### Cisco CUCM: Crestron Standard SIP Profile (1/4)

The screenshot displays the Cisco Unified CM Administration interface for configuring a SIP Profile. The main configuration area is titled "SIP Profile Configuration" and includes a toolbar with "Save", "Delete", "Copy", "Reset", and "Apply Config" buttons. The "Device" menu is open, showing "Device Settings" selected. The "SIP Profile Information" section contains the following fields:

- Name\*: Crestron Standard SIP Profile
- Description: Crestron Standard 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

Below these fields are several checkboxes, all of which are unchecked:

- Redirect by Application
- Disable Early Media on 180
- Outgoing T.38 INVITE include audio mline
- Offer valid IP and Send/Receive mode only for T.38 Fax Relay
- Use Fully Qualified Domain Name in SIP Requests
- Assured Services SIP conformance
- Enable External QoS\*\*

The "SDP Information" section includes:

- 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

At the bottom, there are two more checkboxes, both unchecked:

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

### Cisco CUCM: Crestron Standard SIP Profile (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	<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
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

### Cisco CUCM: Crestron Standard SIP Profile (3/4)

Speed Dial (Abbreviated Dial) URI*	<input type="text" value="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											
<b>Normalization Script</b>											
Normalization Script	<input type="text" value="&lt; None &gt;"/>										
<input type="checkbox"/> Enable Trace											
	<table border="1"><thead><tr><th></th><th>Parameter Name</th><th>Parameter Value</th><th></th><th></th></tr></thead><tbody><tr><td>1</td><td><input type="text"/></td><td><input type="text"/></td><td><input type="button" value="+"/></td><td><input type="button" value="-"/></td></tr></tbody></table>		Parameter Name	Parameter Value			1	<input type="text"/>	<input type="text"/>	<input type="button" value="+"/>	<input type="button" value="-"/>
	Parameter Name	Parameter Value									
1	<input type="text"/>	<input type="text"/>	<input type="button" value="+"/>	<input type="button" value="-"/>							
<b>External Presentation Information</b>											
<input type="checkbox"/> Anonymous External Presentation											
External Presentation Number	<input type="text"/>										
External Presentation Name	<input type="text"/>										
<b>Trunk Specific Configuration</b>											
Reroute Incoming Request to new Trunk based on*	<input type="text" value="Never"/>										
Resource Priority Namespace List	<input type="text" value="&lt; None &gt;"/>										
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"/>										
<b>Early Offer support for voice and video calls*</b>	<input type="text" value="Best Effort (no MTP inserted)"/>										
<input type="checkbox"/> Enable ANAT											
<input type="checkbox"/> Deliver Conference Bridge Identifier											
<input type="checkbox"/> Enable External Presentation Name and Number											
<input type="checkbox"/> Reject Anonymous Incoming Calls											
<input type="checkbox"/> Reject Anonymous Outgoing Calls											
<input type="checkbox"/> Send ILS Learned Destination Route String											
<input type="checkbox"/> Connect Inbound Call before Playing Queuing Announcement											

### Cisco CUCM: Crestron Standard SIP Profile (4/4)

SIP OPTIONS Ping	
<input 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	

<b>Save</b>	Delete	Copy	Reset	<b>Apply Config</b>	Add New
-------------	--------	------	-------	---------------------	---------

### Standard SIP Profile – Cisco PBX phone

- To view the Standard SIP Profile, Navigate to **Device -> Device Settings-> SIP Profile**.
- The Default Standard SIP Profile is shown below.

### Cisco CUCM: Standard SIP Profile (1/4)

The screenshot shows the Cisco Unified CM Administration interface. The 'Device' menu is open, and 'Device Settings' is selected. The 'SIP Profile' sub-menu is also open, showing 'SIP Profile' as the selected option. The main configuration area shows the 'SIP Profile Information' for the 'Standard SIP Profile'.

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

### Cisco CUCM: Standard SIP Profile (2/4)

SDP Information	
SDP Session-level Bandwidth Modifier for Early Offer and Re-invites*	TIAS and AS
SDP Transparency Profile	< None >
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
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

Cisco CUCM: Standard SIP Profile (3/4)

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							
<b>Normalization Script</b>							
Normalization Script	< None >						
<input type="checkbox"/> Enable Trace							
	<table border="1"> <thead> <tr> <th></th> <th>Parameter Name</th> <th>Parameter Value</th> </tr> </thead> <tbody> <tr> <td>1</td> <td></td> <td></td> </tr> </tbody> </table>		Parameter Name	Parameter Value	1		
	Parameter Name	Parameter Value					
1							
<b>External Presentation Information</b>							
<input type="checkbox"/> Anonymous External Presentation							
External Presentation Number							
External Presentation Name							
<b>Trunk Specific Configuration</b>							
Reroute Incoming Request to new Trunk based on*	Never						
Resource Priority Namespace List	< None >						
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*	Disabled (Default value)						
<input type="checkbox"/> Enable ANAT							

### Cisco CUCM: Standard SIP Profile (4/4)

<input type="checkbox"/> Deliver Conference Bridge Identifier	
<input type="checkbox"/> Enable External Presentation Name and Number	
<input type="checkbox"/> Reject Anonymous Incoming Calls	
<input type="checkbox"/> Reject Anonymous Outgoing Calls	
<input type="checkbox"/> Send ILS Learned Destination Route String	
<input type="checkbox"/> Connect Inbound Call before Playing Queuing Announcement	
<b>SIP OPTIONS Ping</b>	
<input 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"/>
<b>SDP Information</b>	
<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	
<input type="button" value="Copy"/> <input type="button" value="Reset"/> <input type="button" value="Apply Config"/> <input type="button" value="Add New"/>	

## Security Profiles

Three Security Profiles were created, one for the Crestron Mercury phones, one for the Cisco 9971 PBX phone and one for the PSTN Trunk.

### Crestron Mercury Phone Security Profile

1. Navigate to **System->Security-> Phone Security Profile**.
2. Click **Add New**.
3. Provide a **Name: *Third-party SIP Device Basic - Standard SIP Non-Secure Profile***.
4. **Transport Type: TCP+UDP**
5. Check the Enable Digest Authentication checkbox
6. Make sure the **SIP Phone Port** is set to **5060**
7. Click **Save**

### Cisco CUCM: Crestron Mercury Security Profile (1/2)

The screenshot shows the Cisco CUCM navigation menu. The 'System' dropdown is selected, and the 'Security' option is highlighted. The 'Security' submenu is open, showing 'Certificate', 'Phone Security Profile', 'SIP Trunk Security Profile', and 'CUMA Server Security Profile'. The 'Phone Security Profile' option is highlighted with a red box.

### Cisco CUCM: Crestron Mercury Security Profile (2/2)

The screenshot shows the 'Phone Security Profile Configuration' page. The page includes a toolbar with 'Copy', 'Reset', 'Apply Config', and 'Add New' buttons. The 'Status' section shows 'Status: Ready'. The 'Phone Security Profile Information' section contains the following fields:

- Product Type:** Third-party SIP Device (Basic)
- Device Protocol:** SIP
- Name\*:** Third-party SIP Device Basic - Standard SIP Non-Secure Profile
- Description:** Third-party SIP Device (Basic) - Standard SIP Non-Secure Profile
- Nonce Validity Time\*:** 600
- Transport Type\*:** TCP+UDP
- Enable Digest Authentication

The 'Parameters used in Phone' section contains the following field:

- SIP Phone Port\*:** 5060

## Cisco 9971 – Security Profile

- Navigate to **System->Security-> Phone Security Profile**
- The Default **Cisco 9971 - Standard SIP Non-Secure Profile** is shown below.

### Cisco CUCM: Cisco 9971 – Standard SIP Non-Secure Profile

#### Phone Security Profile Configuration

Copy Reset Apply Config Add New

---

**Status**

*i* Status: Ready

---

**Phone Security Profile Information**

<b>Product Type:</b>	Cisco 9971
<b>Device Protocol:</b>	SIP
<b>Name *</b>	Cisco 9971 - Standard SIP Non-Secure Profile
<b>Description</b>	Cisco 9971 - Standard SIP Non-Secure Profile
<b>Nonce Validity Time *</b>	600
<b>Device Security Mode</b>	Non Secure
<b>Transport Type *</b>	TCP+UDP
<input type="checkbox"/> Enable Digest Authentication	
<input type="checkbox"/> TFTP Encrypted Config	

---

**Phone Security Profile CAPF Information**

<b>Authentication Mode *</b>	By Null String
<b>Key Order *</b>	RSA Only
<b>RSA Key Size (Bits) *</b>	2048
<b>EC Key Size (Bits)</b>	< None >

Note: These fields are related to the CAPF Information settings on the Phone Configuration page.

---

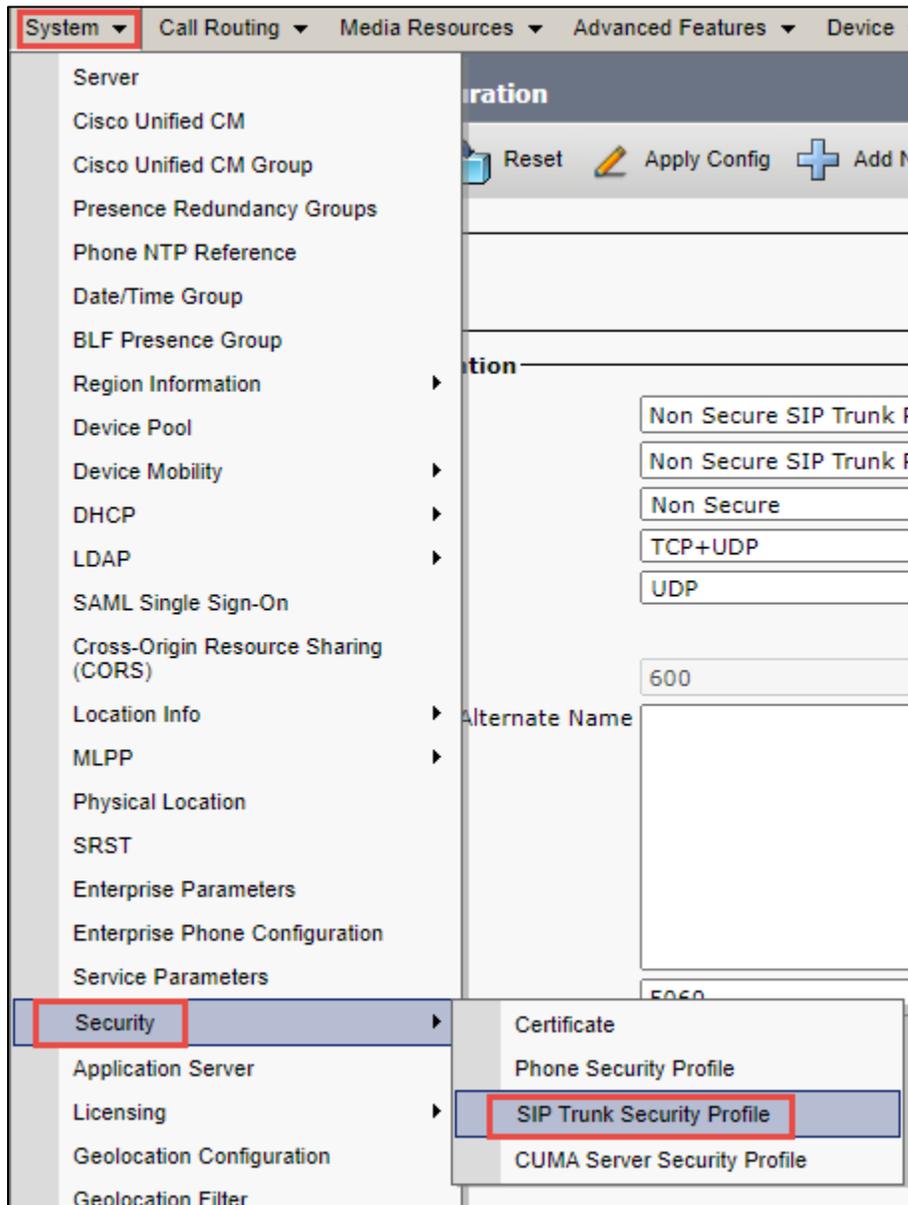
**Parameters used in Phone**

<b>SIP Phone Port *</b>	5060
-------------------------	------

## PSTN Trunk - SIP Trunk Security Profile

1. Navigate to **System->Security-> SIP Trunk Security Profile**.
2. Click **Add New**.
3. **Name: Non-Secure SIP Trunk Profile**.
4. **Incoming Transport Type: TCP+UDP**
5. **Outgoing Transport Type: UDP**
6. Make sure the **Incoming Port** is set to **5060**

### Cisco CUCM: PSTN Trunk – SIP Trunk Security Profile (1/2)



The screenshot shows the Cisco CUCM configuration interface. The navigation menu on the left is expanded to show the path: **System** > **Security** > **SIP Trunk Security Profile**. The **System** and **Security** items are highlighted with red boxes. The **SIP Trunk Security Profile** item is also highlighted with a red box. The main configuration area shows the details for a profile named "Non Secure SIP Trunk P". The configuration includes fields for Name, Incoming Transport Type (set to TCP+UDP), Outgoing Transport Type (set to UDP), and Incoming Port (set to 600). The "Alternate Name" field is empty.

### Cisco CUCM: PSTN Trunk – SIP Trunk Security Profile (2/2)

#### SIP Trunk Security Profile Configuration

Save Delete Copy Reset Apply Config Add New

**Status**  
Status: Ready

**SIP Trunk Security Profile Information**

Name*	Non Secure SIP Trunk Profile
Description	Non Secure SIP Trunk Profile authenticated by null String
Device Security Mode	Non Secure
Incoming Transport Type*	TCP+UDP
Outgoing Transport Type	UDP

Enable Digest Authentication  
Nonce Validity Time (mins)\* 600  
Secure Certificate Subject or Subject Alternate Name

Incoming Port*	5060
----------------	------

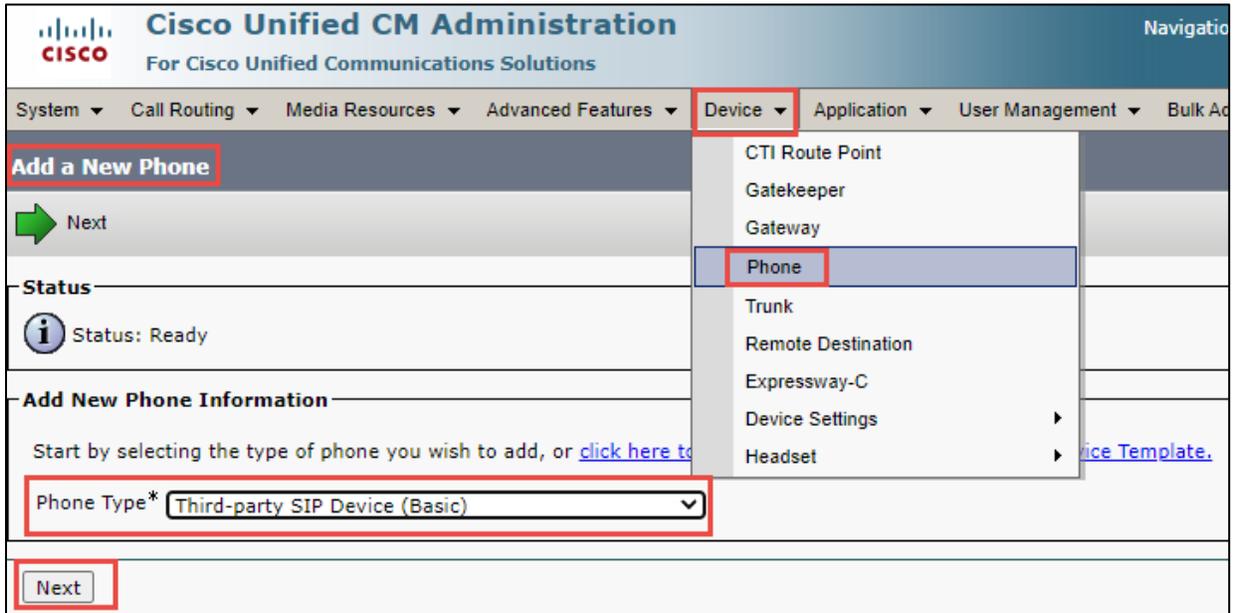
Enable Application level authorization  
 Accept presence subscription  
 Accept out-of-dialog refer\*\*  
 Accept unsolicited notification  
 Accept replaces header  
 Transmit security status  
 Allow charging header  
SIP V.150 Outbound SDP Offer Filtering\* Use Default Filter

## Crestron Mercury devices Configured as Third Party SIP Device (Basic)

The Crestron Mercury devices are configured as a Third Party SIP Device (Basic) in the Cisco CUCM Phone Configuration

1. Navigate to **Device->Phone**.
2. Click **Add New**.
3. **Phone Type** as **Third-party SIP Device (Basic)**.
4. Click **Next**
5. **MAC Address:** Enter MAC Address of the Crestron Mercury - **00177F8B67B8**.
6. **Device Pool:** **G711\_pool**.
7. **Phone Button Template:** as **Third-party SIP Device (Basic)**.
8. **Common Phone Template:** as **Standard Common Phone Profile**.
9. **Owner:** click the **User** radio button.
10. **Owner User ID:** select the End User configured earlier from the drop down. **2648** is selected for the Crestron Mercury, **2645** is selected for the Crestron Mercury X.
11. **Device Security Profile** as **Third-party SIP Device Basic - Standard SIP Non-Secure Profile**.
12. **SIP Profile** as configured earlier from the drop down menu - **Crestron Standard SIP Profile**.
13. **Digest User ID:** select the End User configured earlier from the drop down. **2648** is selected for the Crestron Mercury and **2645** is for the Crestron Mercury X
14. Click **Save**

### Cisco CUCM: Third Party SIP Device (Basic) (1/3)



The screenshot displays the Cisco Unified CM Administration interface. The top navigation bar includes 'System', 'Call Routing', 'Media Resources', 'Advanced Features', 'Device', 'Application', 'User Management', and 'Bulk Ad'. The 'Device' menu is expanded, showing options like 'CTI Route Point', 'Gatekeeper', 'Gateway', 'Phone', 'Trunk', 'Remote Destination', 'Expressway-C', 'Device Settings', and 'Headset'. The 'Phone' option is selected. Below the navigation, the 'Add a New Phone' section is visible, with a 'Next' button. The 'Status' section shows 'Status: Ready'. The 'Add New Phone Information' section contains a 'Phone Type\*' dropdown menu set to 'Third-party SIP Device (Basic)'. A 'Next' button is located at the bottom of the form.

Cisco CUCM: Third Party SIP Device (Basic) (2/3)

<b>Phone Type</b>	
<b>Product Type:</b>	Third-party SIP Device (Basic)
<b>Device Protocol:</b>	SIP
<b>Real-time Device Status</b>	
<b>Registration:</b>	Registered with Cisco Unified Communications Manager 10.80.17.2
<b>IPv4 Address:</b>	192.168.58.104
<b>Active Load ID:</b>	None
<b>Download Status:</b>	None
<b>Device Information</b>	
<input checked="" type="checkbox"/>	Device is Active
<input type="checkbox"/>	Device is not trusted
<b>MAC Address*</b>	00177F8B67B8 (SEP00177F8B67B8)
<b>Description</b>	SEP00177F8B67B8
<b>Device Pool*</b>	G711_Pool <a href="#">View Details</a>
<b>Common Device Configuration</b>	< None > <a href="#">View Details</a>
<b>Phone Button Template*</b>	Third-party SIP Device (Basic) <a href="#">View Details</a>
<b>Common Phone Profile*</b>	Standard Common Phone Profile <a href="#">View Details</a>
<b>Calling Search Space</b>	< None >
<b>AAR Calling Search Space</b>	< None >
<b>Media Resource Group List</b>	< None >
<b>Location*</b>	Hub_None
<b>AAR Group</b>	< None >
<b>Device Mobility Mode*</b>	Default <a href="#">View Current Device Mobility Settings</a>
<b>Owner</b>	<input checked="" type="radio"/> User <input type="radio"/> Anonymous (Public/Shared Space)
<b>Owner User ID*</b>	2648
<b>Mobility User ID</b>	< None >
<b>Use Trusted Relay Point*</b>	Default
<b>Always Use Prime Line*</b>	Default
<b>Always Use Prime Line for Voice Message*</b>	Default
<b>Geolocation</b>	< None >
<input type="checkbox"/>	Ignore Presentation Indicators (internal calls only)
<input checked="" type="checkbox"/>	Logged Into Hunt Group
<input type="checkbox"/>	Remote Device

### Cisco CUCM: Third Party SIP Device (Basic) (3/3)

<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*	Third-party SIP Device Basic - Standard SIP Non-Se
Rerouting Calling Search Space	< None >
SUBSCRIBE Calling Search Space	< None >
SIP Profile*	Crestron Standard SIP Profile <a href="#">View Details</a>
Digest User	2648
<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 >

## Directory Number

Assign a Directory Number to the Crestron Mercury devices.

1. From the Crestron Mercury Phone Configuration, (**Device->Phone**).
2. Click on **Add a new DN**.
3. **Directory Number: 2648** is used for the Crestron Mercury and DN **2645** is used for Crestron Mercury X.
4. The 10 Digit DID is entered for the **Display (Caller ID)**, **ASCII Display (Caller ID)** and the **External Phone Number Mask**.

### Cisco CUCM: Crestron Mercury Directory Number (1/4)

**Phone Configuration**

Save ✖ Delete 📄 Copy 🔄 Reset 🖋 Apply Config ➕ Add New

**Status**

📘 Status: Ready

**Association**

Modify Button Items

1	778 785	Line [1] - Add a new DN
---	------------	-------------------------

**Phone Type**

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

---

**Real-time Device Status**

**Registration:** Unregistered  
**IPv4 Address:** 192.168.58.104  
**Active Load ID:** None  
**Download Status:** None

---

**Device Information**

Device is Active  
 Device is not trusted

MAC Address*	00177F8B67B8
Description	SEP00177F8B67B8
Device Pool*	G711_Pool
Common Device Configuration	< None >
Phone Button Template*	Third-party SIP Device (Basic)

### Cisco CUCM: Crestron Mercury Directory Number (2/4)

#### Directory Number Configuration

Save  Delete  Reset  Apply Config  Add New

---

##### Directory Number Information

Directory Number\*   Urgent Priority

Route Partition

Description

Alerting Name

ASCII Alerting Name

External Call Control Profile

Associated Devices

▼ ▲

Dissociate Devices

---

##### Directory Number Settings

Voice Mail Profile  (Choose <None> to use s

Calling Search Space

BLF Presence Group\*

User Hold MOH Audio Source

Network Hold MOH Audio Source

Calling Line ID Presentation When Diverted

Reject Anonymous Calls

---

##### External Presentation Information

Anonymous External Presentation

External Presentation Number

External Presentation Name

---

##### Enterprise Alternate Number

---

##### +E.164 Alternate Number

### Cisco CUCM: Crestron Mercury Directory Number (3/4)

Primary	URI	Partition	Advertise Globally via ILS
<input type="checkbox"/>	<input type="text"/>	< None >	<input checked="" type="checkbox"/>
<input type="button" value="Add Row"/>			

**- PSTN Failover for Enterprise Alternate Number, +E.164 Alternate Number, and URI Dialing -**  
 Advertised Failover Number: < None >

**- AAR Settings -**

AAR	Voice Mail	AAR Destination Mask	AAR Group
<input type="checkbox"/>	<input type="checkbox"/>	<input type="text"/>	< None >

Retain this destination in the call forwarding history

**- Call Forward and Call Pickup Settings -**

	Voice Mail	Destination	Calling Search Space
Calling Search Space Activation Policy			Use System Default
Forward All	<input type="checkbox"/>	<input type="text"/>	< None >
Secondary Calling Search Space for Forward All			< None >
Forward Busy Internal	<input type="checkbox"/>	<input type="text"/>	< None >
Forward Busy External	<input type="checkbox"/>	<input type="text"/>	< None >
Forward No Answer Internal	<input type="checkbox"/>	<input type="text"/>	< None >
Forward No Answer External	<input type="checkbox"/>	<input type="text"/>	< None >
Forward No Coverage Internal	<input type="checkbox"/>	<input type="text"/>	< None >
Forward No Coverage External	<input type="checkbox"/>	<input type="text"/>	< None >
Forward on CTI Failure	<input type="checkbox"/>	<input type="text"/>	< None >
Forward Unregistered Internal	<input type="checkbox"/>	<input type="text"/>	< None >
Forward Unregistered External	<input type="checkbox"/>	<input type="text"/>	< None >
No Answer Ring Duration (seconds)	<input type="text"/>		
Call Pickup Group		< None >	

**- Park Monitoring -**

	Voice Mail	Destination	Calling Search Space
Park Monitoring Forward No Retrieve Destination External	<input type="checkbox"/>	<input type="text"/>	< None >
Park Monitoring Forward No Retrieve Destination Internal	<input type="checkbox"/>	<input type="text"/>	< None >

### Cisco CUCM: Crestron Mercury Directory Number (4/4)

Park Monitoring Reversion Timer:  A blank value will use value set in Park Monitoring Reversion Timer service parameter

**- MLPP Alternate Party And Confidential Access Level Settings -**

Target (Destination):

MLPP Calling Search Space: < None >

MLPP No Answer Ring Duration (seconds):

Confidential Access Mode: < None >

Confidential Access Level: < None >

Call Control Agent Profile: < None >

**- Line Settings for All Devices -**

Hold Reversion Ring Duration (seconds):  Setting the Hold Reversion Ring Duration to zero will disable the feature

Hold Reversion Notification Interval (seconds):  Setting the Hold Reversion Notification Interval to zero will disable the feature

Party Entrance Tone\*: Default

**- Line 1 on Device SEP00177F8B67B8 -**

Display (Caller ID): 972-2648

ASCII Display (Caller ID): 972-2648

External Phone Number Mask: 972-2648

Monitoring Calling Search Space: < None >

**- Multiple Call/Call Waiting Settings on Device SEP00177F8B67B8 -**

Note: The range to select the Max Number of calls is: 1-2

Maximum Number of Calls\*: 2

Busy Trigger\*: 2 (Less than or equal to Max. Calls)

**- Forwarded Call Information Display on Device SEP00177F8B67B8 -**

Caller Name

Caller Number

Redirected Number

Dialed Number

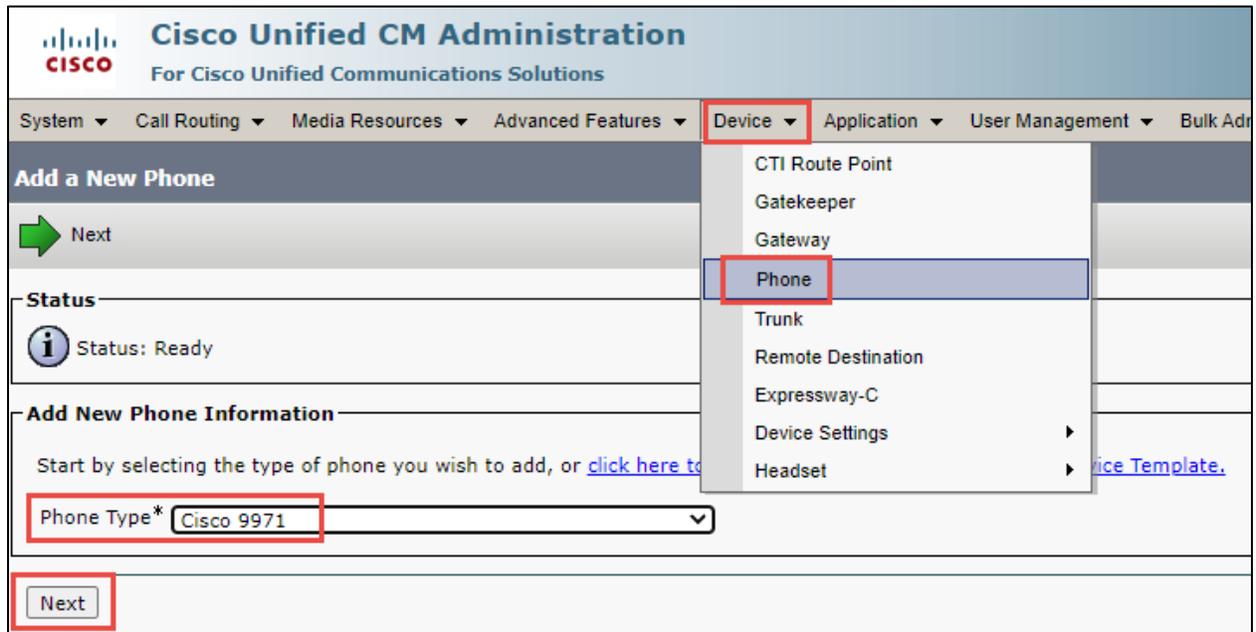
**- Users Associated with Line -**

## Cisco 9971 SIP PBX Phone

The Cisco 9971 PBX phone is SIP PBX phone.

1. Navigate to **Device->Phone**
2. Click **Add New**
3. Select **Phone Type** as **Cisco 9971**
4. Click **Next**
5. **MAC Address:** Enter MAC Address of the Cisco 9971 SIP Phone - **1C17D337D08D**.
6. **Device Pool** – **G711\_pool**.
7. **Phone Button Template** as **Standard 9971 SIP**.
8. **Media Resource Group List:** **Crestron**, (created below).
9. **Owner:** select the **Anonymous** radio button.
10. **Device Security Profile** from the drop down – **Cisco 9971 - Standard SIP Non-Secure Profile**.
11. **SIP Profile** from the drop down select **Crestron Standard SIP Profile**.
12. Click **Save**

### Cisco CUCM: Cisco 9971 SIP Phone (1/7)



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', and 'Bulk Addressing'. The 'Device' menu is expanded, showing options like 'CTI Route Point', 'Gatekeeper', 'Gateway', 'Phone', 'Trunk', 'Remote Destination', 'Expressway-C', 'Device Settings', and 'Headset'. The 'Phone' option is selected. Below the navigation bar, the 'Add a New Phone' section is visible. It includes a 'Next' button with a green arrow, a 'Status' section showing 'Status: Ready', and an 'Add New Phone Information' section. The 'Phone Type\*' dropdown menu is set to 'Cisco 9971'. A 'Next' button is located at the bottom of the form.

Cisco CUCM: Cisco 9971 SIP Phone (2/7)

Phone Load Name	<input type="text"/>
Use Trusted Relay Point*	Default <input type="button" value="v"/>
BLF Audible Alert Setting (Phone Idle)*	Default <input type="button" value="v"/>
BLF Audible Alert Setting (Phone Busy)*	Default <input type="button" value="v"/>
Always Use Prime Line*	Default <input type="button" value="v"/>
Always Use Prime Line for Voice Message*	Default <input type="button" value="v"/>
Geolocation	< None > <input type="button" value="v"/>
Feature Control Policy	< None > <input type="button" value="v"/>
<input type="checkbox"/> Ignore Presentation Indicators (internal calls only) <input checked="" type="checkbox"/> Allow Control of Device from CTI <input checked="" type="checkbox"/> Logged Into Hunt Group <input type="checkbox"/> Remote Device <input type="checkbox"/> Protected Device**** <input type="checkbox"/> Require off-premise location	
<b>Number Presentation Transformation</b>	
<b>Caller ID For Calls From This Phone</b>	
Calling Party Transformation CSS	< None > <input type="button" value="v"/>
<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 type="button" value="v"/>
<input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS (Device Mobility Related Information)	
<b>Protocol Specific Information</b>	
Packet Capture Mode*	None <input type="button" value="v"/>
Packet Capture Duration	<input type="text" value="0"/>
BLF Presence Group*	Standard Presence group <input type="button" value="v"/>
SIP Dial Rules	< None > <input type="button" value="v"/>
MTP Preferred Originating Codec*	711ulaw <input type="button" value="v"/>
Device Security Profile*	Cisco 9971 - Standard SIP Non-Secure Profile <input type="button" value="v"/>
Rerouting Calling Search Space	< None > <input type="button" value="v"/>
SUBSCRIBE Calling Search Space	< None > <input type="button" value="v"/>
SIP Profile*	Crestron Standard SIP Profile <input type="button" value="v"/> <a href="#">View Details</a>
Digest User	< None > <input type="button" value="v"/>
<input type="checkbox"/> Media Termination Point Required	

Cisco CUCM: Cisco 9971 SIP Phone (3/7)

<input type="checkbox"/> Unattended Port	
<input type="checkbox"/> Require DTMF Reception	
<b>Certification Authority Proxy Function (CAPF) Information</b>	
Certificate Operation*	<input type="text" value="No Pending Operation"/>
Authentication Mode*	<input type="text" value="By Null String"/>
Authentication String	<input type="text"/>
<input type="button" value="Generate String"/>	
Key Order*	<input type="text" value="RSA Only"/>
RSA Key Size (Bits)*	<input type="text" value="2048"/>
EC Key Size (Bits)	<input type="text"/>
Operation Completes By	<input type="text" value="2021"/> <input type="text" value="10"/> <input type="text" value="08"/> <input type="text" value="12"/> (YYYY:MM:DD:HH)
Certificate Operation Status: None	
Note: Security Profile Contains Addition CAPF Settings.	
<b>Expansion Module Information</b>	
Module 1	<input type="text" value="&lt; None &gt;"/>
Module 1 Load Name	<input type="text"/>
Module 2	<input type="text" value="&lt; None &gt;"/>
Module 2 Load Name	<input type="text"/>
Module 3	<input type="text" value="&lt; None &gt;"/>
Module 3 Load Name	<input type="text"/>
<b>External Data Locations Information (Leave blank to use default)</b>	
Information	<input type="text"/>
Directory	<input type="text"/>
Messages	<input type="text"/>
Services	<input type="text"/>
Authentication Server	<input type="text"/>
Proxy Server	<input type="text"/>
Idle	<input type="text"/>
Idle Timer (seconds)	<input type="text"/>
Secure Authentication URL	<input type="text"/>
Secure Directory URL	<input type="text"/>
Secure Idle URL	<input type="text"/>
Secure Information URL	<input type="text"/>

**Cisco CUCM: Cisco 9971 SIP Phone (4/7)**

Secure Messages URL	<input type="text"/>
Secure Services URL	<input type="text"/>
<b>Extension Information</b>	
<input type="checkbox"/> Enable Extension Mobility	
Log Out Profile	-- Use Current Device Settings --
Log in Time	< None >
Log out Time	< None >
<b>MLPP and Confidential Access Level Information</b>	
MLPP Domain	< None >
MLPP Indication*	Default
MLPP Preemption*	Default
Confidential Access Mode	< None >
Confidential Access Level	< None >
<b>Do Not Disturb</b>	
<input type="checkbox"/> Do Not Disturb	
DND Option*	Use Common Phone Profile Setting
DND Incoming Call Alert	< None >
<b>Secure Shell Information</b>	
Secure Shell User	<input type="text"/>
Secure Shell Password	<input type="text"/>
<b>Product Specific Configuration Layout</b>	
?	
<b>Parameter Value</b>	
<input type="checkbox"/> Disable Speakerphone	
<input type="checkbox"/> Disable Speakerphone and Headset	
PC Port *	Enabled
Back USB Port*	Enabled
Side USB Port*	Enabled
Cisco Camera*	Disabled
Console Access*	Disabled
Video Capabilities*	Disabled
Enable/Disable USB Classes	Mass Storage Human Interface Device

Cisco CUCM: Cisco 9971 SIP Phone (5/7)

Enable/Disable USB Classes	Mass Storage Human Interface Device Audio Class	<input type="checkbox"/>
SDIO *	Disabled	<input type="checkbox"/>
Bluetooth *	Enabled	<input type="checkbox"/>
Wifi *	Enabled	<input type="checkbox"/>
Bluetooth Profiles*	Handsfree Human Interface Device	<input type="checkbox"/>
Settings Access*	Enabled	<input type="checkbox"/>
Gratuitous ARP*	Disabled	<input type="checkbox"/>
PC Voice VLAN Access*	Enabled	<input type="checkbox"/>
Web Access*	Disabled	<input type="checkbox"/>
Show All Calls on Primary Line*	Disabled	<input type="checkbox"/>
Days Display Not Active	Sunday Monday Tuesday	<input type="checkbox"/>
Display On Time	07:30	<input type="checkbox"/>
Display On Duration	10:30	<input type="checkbox"/>
Display Idle Timeout	01:00	<input type="checkbox"/>
HTTPS Server*	http and https Enabled	<input type="checkbox"/>
Enable Power Save Plus	Sunday Monday Tuesday	<input type="checkbox"/>
Phone On Time	00:00	<input type="checkbox"/>
Phone Off Time	24:00	<input type="checkbox"/>
Phone Off Idle Timeout*	60	<input type="checkbox"/>
<input type="checkbox"/> Enable Audible Alert		<input type="checkbox"/>
EnergyWise Domain		<input type="checkbox"/>
EnergyWise Endpoint Security Secret		<input type="checkbox"/>
<input type="checkbox"/> Allow EnergyWise Overrides		<input type="checkbox"/>
Span to PC Port*	Disabled	<input type="checkbox"/>
Logging Display*	Disabled	<input type="checkbox"/>
Load Server		<input type="checkbox"/>
IPv6 Load Server		<input type="checkbox"/>
Recording Tone*	Disabled	<input type="checkbox"/>
Recording Tone Local Volume*	100	
Recording Tone Remote Volume*	50	

**Cisco CUCM: Cisco 9971 SIP Phone (6/7)**

Recording Tone Duration	<input type="text"/>	
Display On When Incoming Call*	Enabled	<input type="checkbox"/>
RTCP*	Disabled	<input type="checkbox"/>
Log Server	<input type="text"/>	<input type="checkbox"/>
IPv6 Log Server	<input type="text"/>	<input type="checkbox"/>
Remote Log*	Disabled	<input type="checkbox"/>
Log Profile	Default Preset Telephony	<input type="checkbox"/>
Advertise G.722 and iSAC Codecs *	Use System Default	
Wideband Headset UI Control*	Enabled	
Wideband Headset*	Enabled	
Peer Firmware Sharing*	Enabled	<input type="checkbox"/>
Cisco Discovery Protocol (CDP): Switch Port*	Enabled	<input type="checkbox"/>
Cisco Discovery Protocol (CDP): PC Port*	Enabled	<input type="checkbox"/>
Link Layer Discovery Protocol - Media Endpoint Discover (LLDP-MED): Switch Port*	Enabled	<input type="checkbox"/>
Link Layer Discovery Protocol (LLDP): PC Port*	Enabled	<input type="checkbox"/>
LLDP Asset ID	<input type="text"/>	
LLDP Power Priority*	Unknown	
802.1x Authentication*	User Controlled	<input type="checkbox"/>
FIPS Mode*	Disabled	<input type="checkbox"/>
Detect Unified CM Connection Failure*	Normal	<input type="checkbox"/>
Switch Port Remote Configuration*	Disabled	<input type="checkbox"/>
PC Port Remote Configuration*	Disabled	<input type="checkbox"/>
Automatic Port Synchronization*	Disabled	<input type="checkbox"/>
Power Negotiation*	Enabled	<input type="checkbox"/>
Restrict Data Rates*	Disabled	<input type="checkbox"/>
SSH Access*	Disabled	<input type="checkbox"/>
Incoming Call Toast Timer*	5	<input type="checkbox"/>
Provide Dial Tone from Release Button*	Disabled	<input type="checkbox"/>

**Cisco CUCM: Cisco 9971 SIP Phone (7/7)**

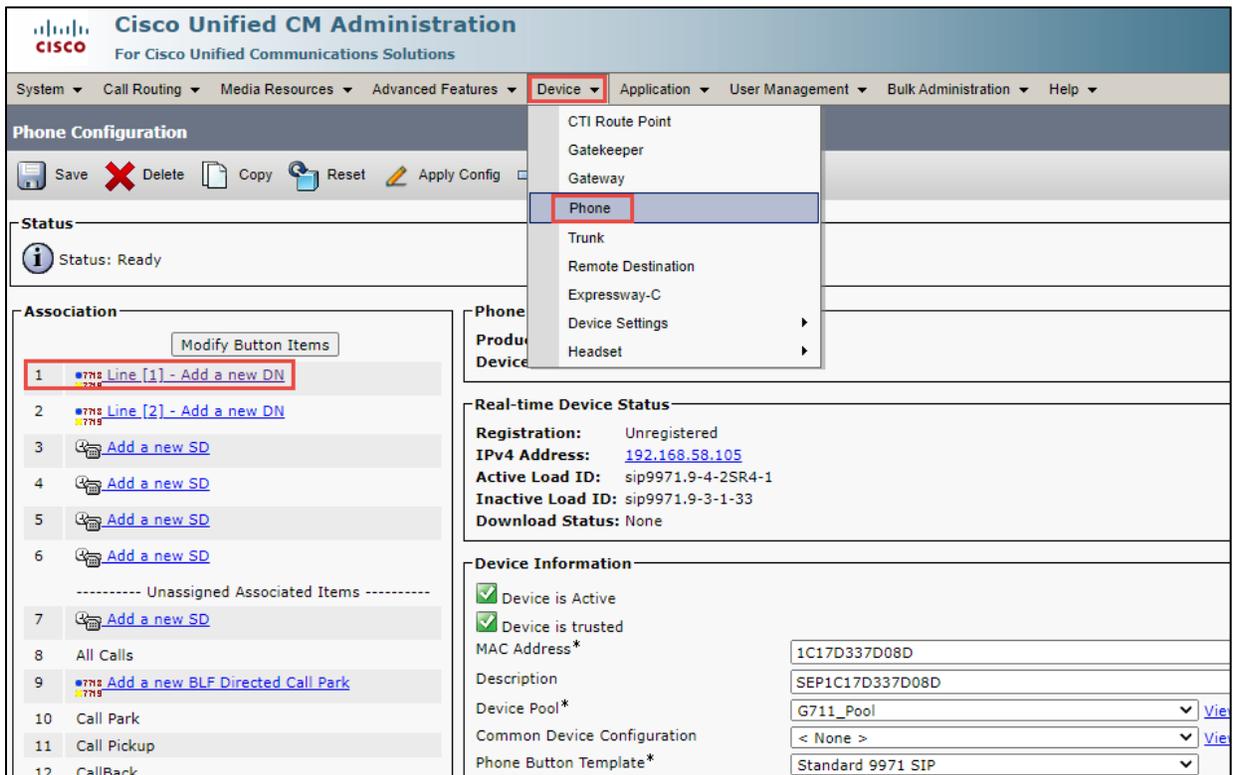
Hide Video By Default*	Disabled	<input type="checkbox"/>
Background Image		<input type="checkbox"/>
Simplified New Call UI*	Disabled	<input type="checkbox"/>
Enable VXC VPN for MAC		<input type="checkbox"/>
VXC VPN Option*	Dual Tunnel	<input type="checkbox"/>
VXC Challenge*	Challenge	<input type="checkbox"/>
VXC-M Servers		<input type="checkbox"/>
Revert to All Calls*	Disabled	<input type="checkbox"/>
RTCP for Video*	Enabled	<input type="checkbox"/>
Record Call Log from Shared Line*	Disabled	<input type="checkbox"/>
Show Remote Private Calls*	Disabled	<input type="checkbox"/>
Record Call Log For Remote Private Calls*	Enabled	<input type="checkbox"/>
Show Call History for Selected Line Only.*	Disabled	<input type="checkbox"/>
Actionable Incoming Call Alert*	Disabled	<input type="checkbox"/>
DF bit*	0	<input type="checkbox"/>
Default Line Filter		<input type="checkbox"/>
Separate Audio and Video Mute*	Disabled	<input type="checkbox"/>
Softkey Control*	Feature Control Policy	<input type="checkbox"/>
Start Video Port		<input type="checkbox"/>
Stop Video Port		<input type="checkbox"/>
Lowest Alerting Line State Priority*	Disabled	<input type="checkbox"/>
TLS Resumption Timer*	3600	<input type="checkbox"/>
Audio EQ*	Default : Default	<input type="checkbox"/>

## Directory Number

Assign a Directory Number to the Cisco PBX phone.

1. From the Cisco PBX Phone Configuration, (**Device->Phone**).
2. Click on **Add a new DN**.
3. **Directory Number: 2640** is configured for the Cisco 9971 SIP Phone
4. The 10 Digit DID is entered for the **Display (Caller ID)**, **ASCII Display (Caller ID)** and the **External Phone Number Mask**.

### Cisco CUCM: Cisco 9971 SIP Phone Directory Number (1/5)



The screenshot displays 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 'Device' menu is expanded, showing options like 'CTI Route Point', 'Gatekeeper', 'Gateway', 'Phone', 'Trunk', 'Remote Destination', 'Expressway-C', 'Device Settings', and 'Headset'. The 'Phone' option is selected.

The main content area is divided into several sections:

- Phone Configuration:** Includes buttons for 'Save', 'Delete', 'Copy', 'Reset', and 'Apply Config'.
- Status:** Shows 'Status: Ready'.
- Association:** A list of lines (1-12) with 'Add a new DN' links. Line 1 is selected, and 'Add a new DN' is highlighted.
- Real-time Device Status:**
  - Registration: Unregistered
  - IPv4 Address: [192.168.58.105](#)
  - Active Load ID: sip9971.9-4-2SR4-1
  - Inactive Load ID: sip9971.9-3-1-33
  - Download Status: None
- Device Information:**
  - Device is Active:
  - Device is trusted:
  - MAC Address\*: 1C17D337D08D
  - Description: SEP1C17D337D08D
  - Device Pool\*: G711\_Pool
  - Common Device Configuration: < None >
  - Phone Button Template\*: Standard 9971 SIP

### Cisco CUCM: Cisco 9971 SIP Phone Directory Number (2/5)

#### Directory Number Configuration

Save  Delete  Reset  Apply Config  Add New

---

##### Directory Number Information

Directory Number\*   Urgent Priority

Route Partition

Description

Alerting Name

ASCII Alerting Name

External Call Control Profile

Allow Control of Device from CTI

Associated Devices

▼ ▲

Dissociate Devices

---

##### Directory Number Settings

Voice Mail Profile  (Choose <None> to u

Calling Search Space

BLF Presence Group\*

User Hold MOH Audio Source

Network Hold MOH Audio Source

Auto Answer\*

Calling Line ID Presentation When Diverted

Reject Anonymous Calls

---

##### External Presentation Information

Anonymous External Presentation

External Presentation Number

External Presentation Name

---

##### Enterprise Alternate Number

---

##### +E.164 Alternate Number

### Cisco CUCM: Cisco 9971 SIP Phone Directory Number (3/5)

Primary	URI	Partition	Advertise Globally via ILS
<input type="checkbox"/>	<input type="text"/>	< None >	<input checked="" type="checkbox"/>
<input type="button" value="Add Row"/>			
<b>- PSTN Failover for Enterprise Alternate Number, +E.164 Alternate Number, and URI Dialing -</b>			
Advertised Failover Number <input type="text" value="&lt; None &gt;"/>			
<b>- AAR Settings -</b>			
AAR	<input type="checkbox"/> or	Voice Mail <input type="text"/>	AAR Destination Mask <input type="text" value="&lt; None &gt;"/>
<input checked="" type="checkbox"/> Retain this destination in the call forwarding history			
<b>- Call Forward and Call Pickup Settings -</b>			
Calling Search Space Activation Policy			Calling Search Space <input type="text" value="Use System Default"/>
Forward All	<input type="checkbox"/> or	<input type="text"/>	<input type="text" value="&lt; None &gt;"/>
Secondary Calling Search Space for Forward All			<input type="text" value="&lt; None &gt;"/>
Forward Busy Internal	<input type="checkbox"/> or	<input type="text"/>	<input type="text" value="&lt; None &gt;"/>
Forward Busy External	<input type="checkbox"/> or	<input type="text"/>	<input type="text" value="&lt; None &gt;"/>
Forward No Answer Internal	<input type="checkbox"/> or	<input type="text"/>	<input type="text" value="&lt; None &gt;"/>
Forward No Answer External	<input type="checkbox"/> or	<input type="text"/>	<input type="text" value="&lt; None &gt;"/>
Forward No Coverage Internal	<input type="checkbox"/> or	<input type="text"/>	<input type="text" value="&lt; None &gt;"/>
Forward No Coverage External	<input type="checkbox"/> or	<input type="text"/>	<input type="text" value="&lt; None &gt;"/>
Forward on CTI Failure	<input type="checkbox"/> or	<input type="text"/>	<input type="text" value="&lt; None &gt;"/>
Forward Unregistered Internal	<input type="checkbox"/> or	<input type="text"/>	<input type="text" value="&lt; None &gt;"/>
Forward Unregistered External	<input type="checkbox"/> or	<input type="text"/>	<input type="text" value="&lt; None &gt;"/>
No Answer Ring Duration (seconds)	<input type="text"/>		
Call Pickup Group	<input type="text" value="&lt; None &gt;"/>		
<b>- Park Monitoring -</b>			
Park Monitoring Forward No Retrieve Destination External	<input type="checkbox"/> or	<input type="text"/>	<input type="text" value="&lt; None &gt;"/> A blank value means to call the park
Park Monitoring Forward No Retrieve Destination Internal	<input type="checkbox"/> or	<input type="text"/>	<input type="text" value="&lt; None &gt;"/> A blank value means to call the park

### Cisco CUCM: Cisco 9971 SIP Phone Directory Number (4/5)

Park Monitoring Reversion Timer	<input type="text"/>	A blank value will use value set in Park Monitoring Reversion Timer service parameter
<b>- MLPP Alternate Party And Confidential Access Level Settings -</b>		
Target (Destination)	<input type="text"/>	
MLPP Calling Search Space	<input type="text" value="&lt; None &gt;"/>	
MLPP No Answer Ring Duration (seconds)	<input type="text"/>	
Confidential Access Mode	<input type="text" value="&lt; None &gt;"/>	
Confidential Access Level	<input type="text" value="&lt; None &gt;"/>	
Call Control Agent Profile	<input type="text" value="&lt; None &gt;"/>	
<b>- Line Settings for All Devices -</b>		
Hold Reversion Ring Duration (seconds)	<input type="text"/>	Setting the Hold Reversion Ring Duration to zero will disable the feature
Hold Reversion Notification Interval (seconds)	<input type="text"/>	Setting the Hold Reversion Notification Interval to zero will disable the feature
Party Entrance Tone*	<input type="text" value="Default"/>	

### Cisco CUCM: Cisco 9971 SIP Phone Directory Number (5/5)

Line 1 on Device SEP1C17D337D08D	
Display (Caller ID)	972-2640-2640 <small>Display text for a line appearance is intended for the phone.</small>
ASCII Display (Caller ID)	972-2640-2640
Line Text Label	
External Phone Number Mask	972-2640-2640
Visual Message Waiting Indicator Policy*	Use System Policy
Audible Message Waiting Indicator Policy*	Default
Ring Setting (Phone Idle)*	Use System Default
Ring Setting (Phone Active)	Use System Default <small>Applies to this line when any line on the phone has a ring.</small>
Call Pickup Group Audio Alert Setting(Phone Idle)	Use System Default
Call Pickup Group Audio Alert Setting(Phone Active)	Use System Default
Recording Option*	Call Recording Disabled
Recording Profile	< None >
Recording Media Source*	Gateway Preferred
Monitoring Calling Search Space	< None >
<input checked="" type="checkbox"/> Log Missed Calls	
Multiple Call/Call Waiting Settings on Device SEP1C17D337D08D	
<small>Note: The range to select the Max Number of calls is: 1-200</small>	
Maximum Number of Calls*	4
Busy Trigger*	2 <small>(Less than or equal to Max. Calls)</small>
Forwarded Call Information Display on Device SEP1C17D337D08D	
<input type="checkbox"/> Caller Name	
<input type="checkbox"/> Caller Number	
<input type="checkbox"/> Redirected Number	
<input type="checkbox"/> Dialed Number	
Users Associated with Line	
<input type="button" value="Associate End Users"/>	
<input type="button" value="Save"/> <input type="button" value="Delete"/> <input type="button" value="Reset"/> <input type="button" value="Apply Config"/> <input type="button" value="Add New"/>	

## Media Resource Group and Media Resource Group List

A Media Resource Group and Media Resource Group List are required to include Music on Hold, (MOH) servers Conference Bridges and Media Termination Points that may be required for the Cisco CUCM MOH features.

### Media Resource Group

Media Resource Group “**Crestron**” is configured for the MOH features.

1. Navigate to **Media Resources** -> **Media Resource Group**.
2. Select **Add New**.
3. Provide a **Name: Crestron**.
4. Move the Media Resources from the **Available Media Resources** box to the **Selected Media Resources** box. (These are assumed to have been added earlier and are available for use /registered with this Cisco CUCM).
  - a. **ANN\_2 (ANN)**
  - b. **CFB\_2 (CFB)**
  - c. **IVR\_2 (IVR)**
  - d. **MOH\_2 (MOH)**
  - e. **MTP\_2 (MTP)**
  - f. **SRTP-MTP (MTP)**
  - g. **XCoder (XCODE)**
  - h. **Crestron (CFB)**
5. Click **Save**.

### Cisco CUCM: Media Resource Group

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

System ▾ Call Routing ▾ **Media Resources ▾** Advanced Features ▾ Device ▾ Application ▾ User M...

**Media Resource Group**

Save **X** Delete

**Status**  
Status: Ready

**Media Resource Group**  
Media Resource Group: C

**Media Resource Group**

Name\*   
Description

**Media Resource Group**

- Annunciator
- Interactive Voice Response
- Conference Bridge
- Media Termination Point
- Music On Hold Audio Source
- Fixed MOH Audio Source
- Music On Hold Server
- Video On Hold Server
- Transcoder
- Media Resource Group**
- Media Resource Group List
- MOH Audio File Management
- Mobile Voice Access
- Announcement

**Devices for this Group**

Available Media Resources\*\*

- ANN\_3
- CFB\_3
- IVR\_3
- MOH\_3
- MTP\_3

Selected Media Resources\*

- ANN\_2 (ANN)
- CFB\_2 (CFB)
- IVR\_2 (IVR)
- MOH\_2 (MOH)
- MTP\_2 (MTP)

Use Multi-cast for MOH Audio (If at least one multi-cast MOH resource is available)

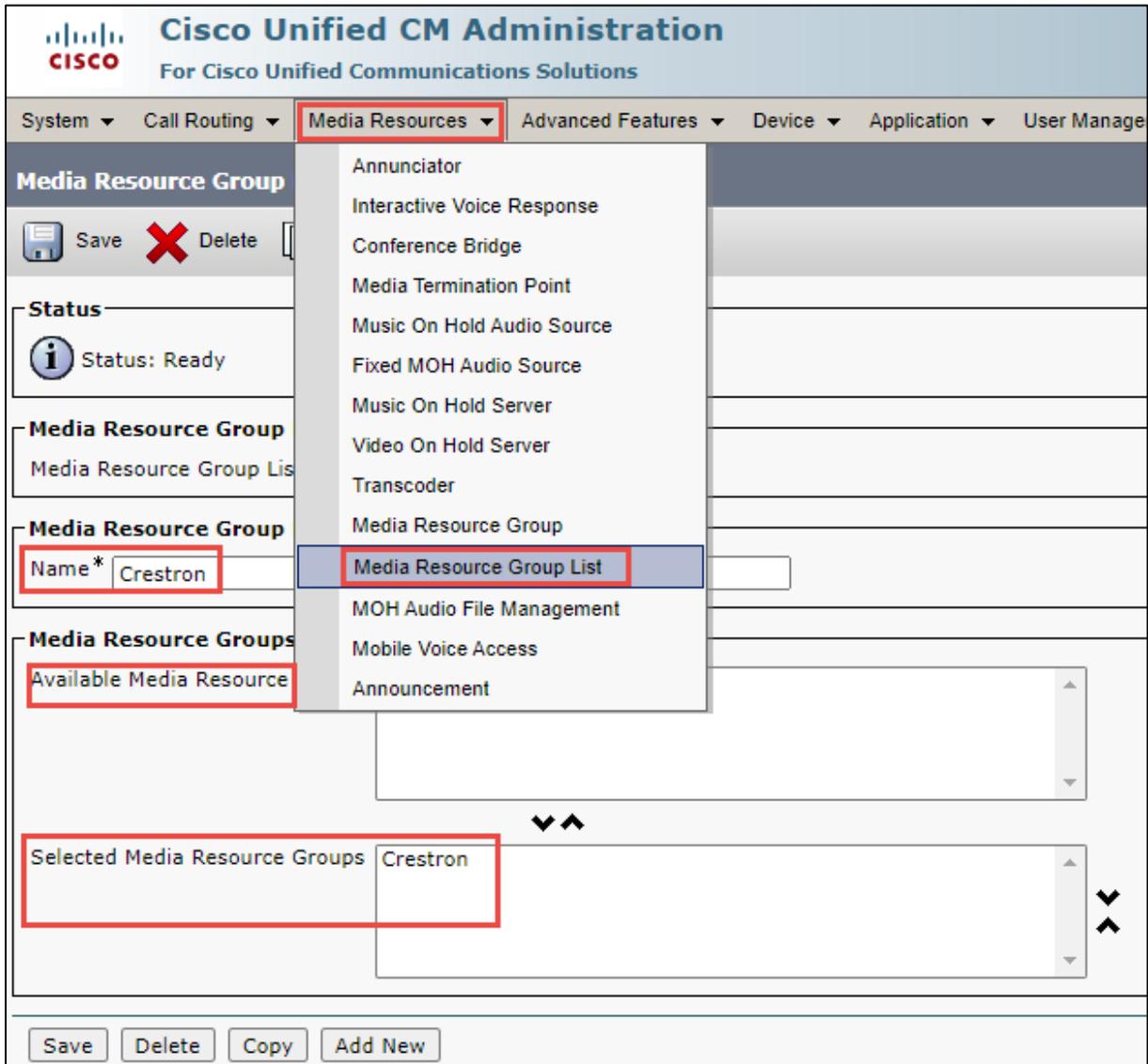
Save Delete Copy Add New

## Media Resource Group List

Media Resource Group List “**Crestron**” is configured for the MOH features.

1. Navigate to **Media Resources** -> **Media Resource Group List**.
2. Select **Add New**.
3. Provide a **Name: Crestron**
4. Move the **Crestron** Media Resource Group from the **Available Media Resource Groups** box to the **Selected Media Resource Groups** box.
5. Click **Save**.

### Cisco CUCM: Media Resource Group



The screenshot displays the Cisco Unified CM Administration web interface. The top navigation bar includes 'System', 'Call Routing', 'Media Resources', 'Advanced Features', 'Device', 'Application', and 'User Management'. The 'Media Resources' menu is expanded, showing a list of options: Annunciator, Interactive Voice Response, Conference Bridge, Media Termination Point, Music On Hold Audio Source, Fixed MOH Audio Source, Music On Hold Server, Video On Hold Server, Transcoder, Media Resource Group, **Media Resource Group List**, MOH Audio File Management, Mobile Voice Access, and Announcement. The 'Media Resource Group List' option is highlighted. Below the navigation bar, the 'Media Resource Group' configuration page is visible. It includes a 'Save' button, a 'Delete' button with a red 'X', and a 'Status' section showing 'Status: Ready'. The 'Media Resource Group' name is 'Media Resource Group List'. The 'Name\*' field contains 'Crestron'. The 'Available Media Resource Groups' list is empty. The 'Selected Media Resource Groups' list contains 'Crestron'. At the bottom, there are buttons for 'Save', 'Delete', 'Copy', and 'Add New'.

## Trunks

Two trunks were configured.

- Between the Cisco CUCM and the PSTN Gateway for calls to and from the PSTN.
- Between the Cisco CUCM and Cisco Unity Connection for Voicemail.

### PSTN Gateway <-> Cisco CUCM Trunk

For the connection between the Cisco CUCM and the PSTN Gateway a Trunk is created.

1. Navigate to **Device** ->**Trunk**.
2. Click **Add New**.
3. **Trunk Type** as **SIP Trunk** , **Device Protocol** as **SIP** and **Trunk Service Type** as **None(Default)**
4. Click **Next**.
5. **Device Name** – **PSTN\_GW**.
6. **Device Pool** - **G711\_Pool**.
7. **Media Resource Group List**, select **Crestron**.
8. Ensure that the **Media Termination Point Required** is unchecked.
9. **Significant Digits**: set to **4**
10. Select the **Redirecting Diversion Header Delivery – Inbound** check box.
11. Select the **Redirecting Diversion Header Delivery – Outbound** check box.
12. **SIP Information** - Destination Address “**10.64.1.72**” and port “**5060**” of the PSTN Gateway.
13. Select the **Non Secure SIP Trunk Profile** as the SIP Trunk Security Profile.
14. Select the configured **Standard SIP Profile** SIP Profile.
15. Click **Save**.

#### Cisco CUCM: PSTN Gateway Trunk (1/5)

The screenshot displays the Cisco Unified CM Administration interface for configuring a Trunk. The breadcrumb navigation shows: System > Call Routing > Media Resources > Advanced Features > Device > Application > User Management. The 'Device' menu is open, with 'Trunk' selected. The 'Trunk Configuration' page shows a 'Next' button at the top left. The 'Status' section indicates 'Status: Ready'. The 'Trunk Information' section contains the following fields:

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

A 'Next' button is located at the bottom left of the configuration area.

### Cisco CUCM: PSTN Gateway Trunk (2/5)

SIP Trunk Status	
<b>Service Status:</b>	Full Service
<b>Duration:</b>	Time In Full Service: 1 day 5 hours 12 minutes
Device Information	
Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type:	None(Default)
Device Name*	PSTN_GW
Description:	PSTN_GW
Device Pool*	G711_Pool
Common Device Configuration:	< None >
Call Classification*	Use System Default
Media Resource Group List:	Crestron
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
<input type="checkbox"/> Media Termination Point Required	
<input checked="" type="checkbox"/> Retry Video Call as Audio	
<input type="checkbox"/> Path Replacement Support	
<input type="checkbox"/> Transmit UTF-8 for Calling Party Name	
<input type="checkbox"/> Transmit UTF-8 Names in QSIG APDU	
<input type="checkbox"/> Unattended Port	
<input type="checkbox"/> SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security	
Consider Traffic on This Trunk Secure*	When using both sRTP and TLS
Route Class Signaling Enabled*	Default
Use Trusted Relay Point*	Default
<input checked="" type="checkbox"/> PSTN Access	
<input type="checkbox"/> Run On All Active Unified CM Nodes	
Intercompany Media Engine (IME)	
E.164 Transformation Profile	< None >

### Cisco CUCM: PSTN Gateway Trunk (3/5)

**- MLPP and Confidential Access Level Information -**

MLPP Domain   
 Confidential Access Mode   
 Confidential Access Level

**- Call Routing Information -**

Remote-Party-Id  
 Asserted-Identity  
 Asserted-Type\*   
 SIP Privacy\*   
 Trust Received Identity\*

**- 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"/>

**- 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"/>

### Cisco CUCM: PSTN Gateway Trunk (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  
 Use original calling line's Calling Line ID Presentation for diverted calls

**- Presentation Information -**

Anonymous Presentation  
 Presentation Number   
 Presentation Name   
 Send Presentation Name and Number only in the FROM header and not in the other identity headers

**- SIP Information -**

**- Destination -**

Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port	Status
1*	10.64.1.72		5060	up

### Cisco CUCM: PSTN Gateway Trunk (5/5)

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*	No Preference	▼

---

**Normalization Script**

Normalization Script

Enable Trace

	Parameter Name	Parameter Value
1	<input type="text"/>	<input type="text"/>

---

**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

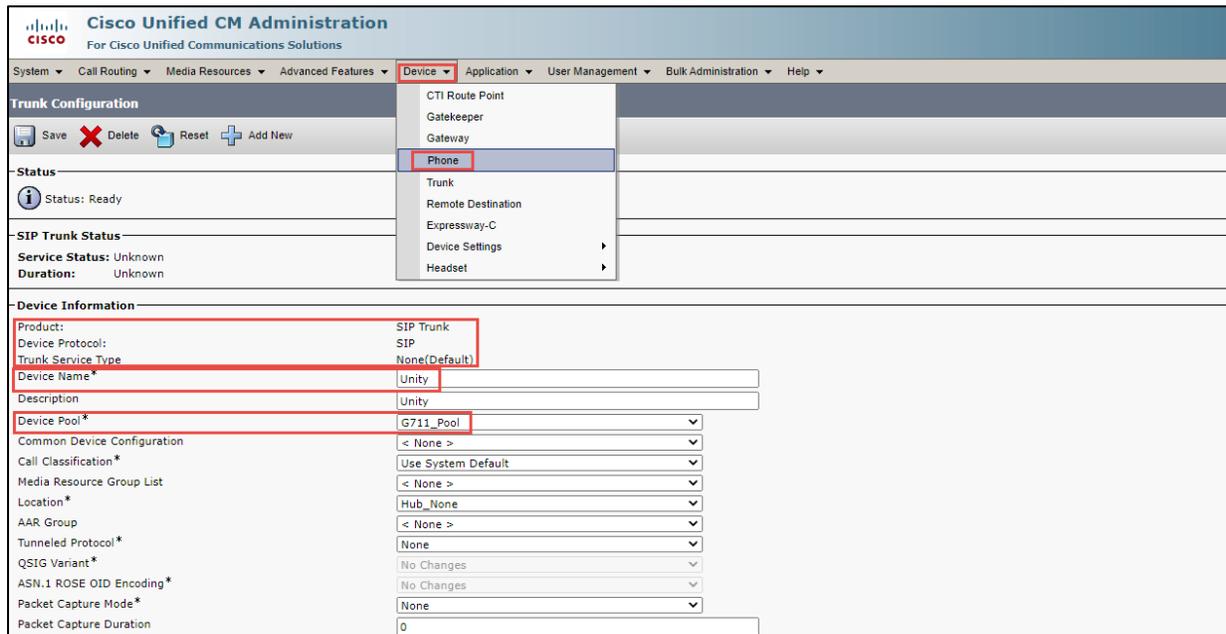
---

## Cisco CUCM <--> Cisco Unity Connection Trunk

For the connection between the Cisco CUCM and the Cisco Unity Connection, a Trunk is created.

1. Navigate to **Device ->Trunk**.
2. Click **Add New**.
3. **Trunk Type** as **SIP Trunk** , **Device Protocol** as **SIP** and **Trunk Service Type** as **None(Default)**
4. Click **Next**.
5. **Device Name** – **Unity**.
6. **Device Pool** - **G711\_Pool**.
7. **SIP Information** - Destination Address “**10.64.1.72**” and port “**5060**” of the PSTN Gateway.
8. Select the **Non Secure SIP Trunk Profile** as the SIP Trunk Security Profile.
9. Select the configured **Standard SIP Profile** SIP Profile.
10. Click **Save**.

### Cisco CUCM: Cisco Unity Connection Trunk (1/4)



The screenshot displays the Cisco Unified CM Administration interface for configuring a SIP Trunk. The 'Device' menu is open, and 'Phone' is selected. The configuration form is as follows:

Cisco Unified CM Administration	
For Cisco Unified Communications Solutions	
System	Call Routing
Media Resources	Advanced Features
Device	Application
User Management	Bulk Administration
Help	
<b>Trunk Configuration</b>	
Save	Delete
Reset	Add New
<b>-Status-</b>	
Status:	Ready
<b>-SIP Trunk Status-</b>	
Service Status:	Unknown
Duration:	Unknown
<b>-Device Information-</b>	
Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type:	None(Default)
Device Name*	Unity
Description	Unity
Device Pool*	G711_Pool
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

### Cisco CUCM: Cisco Unity Connection Trunk (2/4)

<input type="checkbox"/> Media Termination Point Required	
<input checked="" type="checkbox"/> Retry Video Call as Audio	
<input type="checkbox"/> Path Replacement Support	
<input type="checkbox"/> Transmit UTF-8 for Calling Party Name	
<input type="checkbox"/> Transmit UTF-8 Names in QSIG APDU	
<input type="checkbox"/> Unattended Port	
<input type="checkbox"/> SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to	
Consider Traffic on This Trunk Secure* <span style="float: right;">When using both sRTP and TLS</span>	
Route Class Signaling Enabled* <span style="float: right;">Default</span>	
Use Trusted Relay Point* <span style="float: right;">Default</span>	
<input type="checkbox"/> PSTN Access	
<input type="checkbox"/> Run On All Active Unified CM Nodes	
<b>Intercompany Media Engine (IME)</b>	
E.164 Transformation Profile <span style="float: right;">&lt; None &gt;</span>	
<b>MLPP and Confidential Access Level Information</b>	
MLPP Domain <span style="float: right;">&lt; None &gt;</span>	
Confidential Access Mode <span style="float: right;">&lt; None &gt;</span>	
Confidential Access Level <span style="float: right;">&lt; None &gt;</span>	
<b>Call Routing Information</b>	
<input type="checkbox"/> Remote-Party-Id	
<input checked="" type="checkbox"/> Asserted-Identity	
Asserted-Type* <span style="float: right;">Default</span>	
SIP Privacy* <span style="float: right;">Default</span>	
Trust Received Identity* <span style="float: right;">Trust All (Default)</span>	
<b>Inbound Calls</b>	
Significant Digits* <span style="float: right;">All</span>	
Connected Line ID Presentation* <span style="float: right;">Default</span>	
Connected Name Presentation* <span style="float: right;">Default</span>	
Calling Search Space <span style="float: right;">&lt; None &gt;</span>	
AAR Calling Search Space <span style="float: right;">&lt; None &gt;</span>	
Prefix DN <span style="float: right;"><input type="text"/></span>	
<input checked="" type="checkbox"/> Redirecting Diversion Header Delivery - Inbound	

### Cisco CUCM: Cisco Unity Connection Trunk (3/4)

Redirecting Diversion Header Delivery - Inbound

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

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

**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 >	<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  
 Redirecting Party Transformation CSS: < None >  
 Use Device Pool Redirecting Party Transformation CSS  
 Use original calling line's Calling Line ID Presentation for diverted calls

### Cisco CUCM: Cisco Unity Connection Trunk (4/4)

**Presentation Information**  
 Anonymous Presentation  
 Presentation Number:   
 Presentation Name:   
 Send Presentation Name and Number only in the FROM header and not in the other identity headers

**SIP Information**  
**Destination**  
 Destination Address is an SRV  

Destination Address	Destination Address IPv6	Destination Port	Status	Status Reason	Duration
1* 10.80.17.6		5060	N/A	N/A	N/A

 MTP Preferred Originating Codec\*: 711ulaw  
 BLF Presence Group\*: Standard Presence group  
 SIP Trunk Security Profile\*: Non Secure SIP Trunk Profile  
 Routing Calling Search Space: < None >  
 Out-Of-Dialog Refer Calling Search Space: < None >  
 SUBSCRIBE Calling Search Space: < None >  
 SIP Profile\*: Standard SIP Profile [View Details](#)  
 DTMF Signaling Method\*: RFC 2833

**Normalization Script**  
 Normalization Script: < None >  
 Enable Trace  

Parameter Name	Parameter Value
1	

**Recording Information**  
 None  
 This trunk connects to a recording-enabled gateway  
 This trunk connects to other clusters with recording-enabled gateways

**Geolocation Configuration**  
 Geolocation: < None >  
 Geolocation Filter: < None >  
 Send Geolocation Information

## Route Patterns

Route patterns were configured for the following:

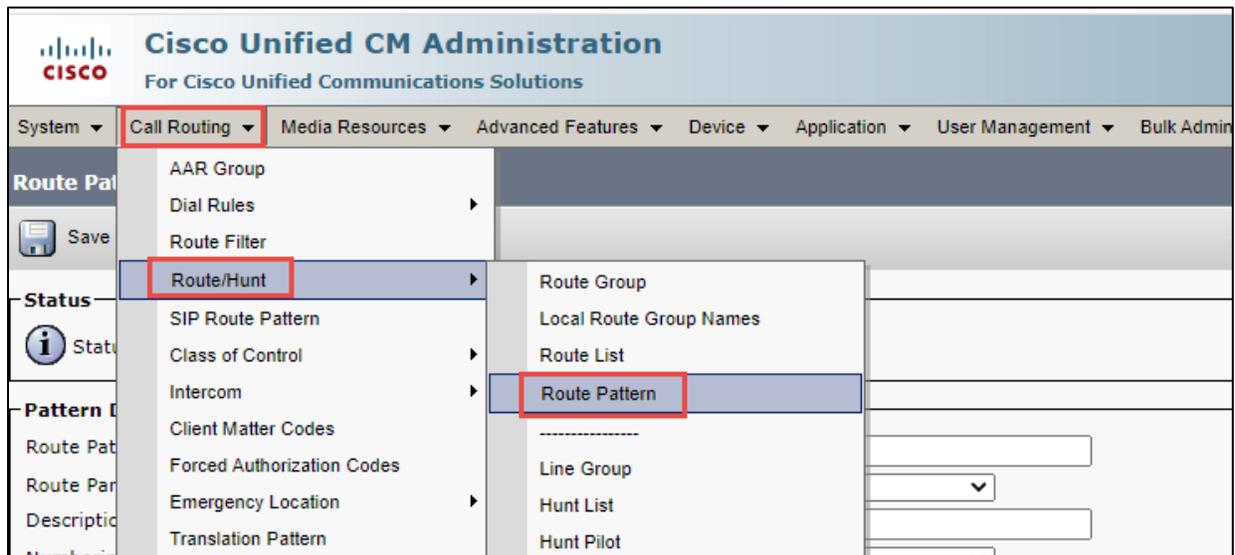
- To route calls from the Cisco CUCM to the PSTN.
- To restrict caller id on outgoing calls.
- To access the voicemail system.

### PSTN Access - 7.@

The route pattern 7.@ is used to enable outbound dialing from the phones to PSTN using the access code "7", before dialing the phone number.

1. Navigate to **Call Routing -> Route/Hunt-> Route Pattern**
2. **Add New**
3. **Route Pattern – 7.@**
4. **Numbering Plan – NANP**
5. **Gateway/Route List – PSTN\_GW**
6. **Call Classification – OffNet**
7. **Calling Line ID Presentation – Default**
8. **Calling Name Presentation – Default**
9. **Calling Party Number Type – Cisco CallManager**
10. **Calling Party Numbering Plan – Cisco CallManager**
11. **Discard Digits - PreDot**

#### Cisco CUCM: PSTN Route Pattern (1/3)



### Cisco CUCM: PSTN Route Pattern (2/3)

Pattern Definition	
Route Pattern*	7.@
Route Partition	< None >
Description	
Numbering Plan*	NANP
Route Filter	< None >
MLPP Precedence*	Default
<input type="checkbox"/> Apply Call Blocking Percentage	
Resource Priority Namespace Network Domain	< None >
Route Class*	Default
Gateway/Route List*	PSTN_GW <a href="#">(Edit)</a>
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern <span>No Error</span>
Call Classification*	OffNet
External Call Control Profile	< None >
<input type="checkbox"/> Allow Device Override	
<input checked="" type="checkbox"/> Provide Outside Dial Tone	
<input type="checkbox"/> Allow Overlap Sending	
<input type="checkbox"/> Urgent Priority	
<input type="checkbox"/> Require Forced Authorization Code	
Authorization Level*	0
<input type="checkbox"/> Require Client Matter Code	

Calling Party Transformations	
<input checked="" 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

### Cisco CUCM: PSTN Route Pattern (3/3)

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

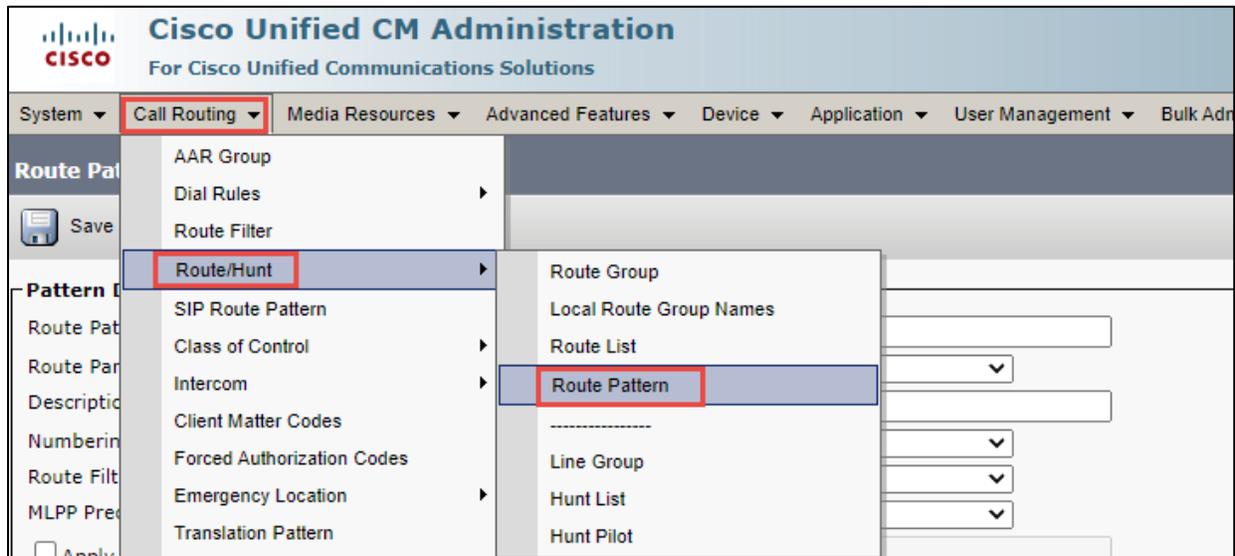
ISDN Network-Specific Facilities Information Element		
Network Service Protocol	-- Not Selected --	
Carrier Identification Code		
Network Service	Service Parameter Name	Service Parameter Value
-- Not Selected --	< Not Exist >	

## Restrict Outbound Caller ID - 767.@

The route pattern of **767.@** is used to restrict caller id on outbound calls to the PSTN using the access code "**767**", before dialing the phone number.

1. Navigate to **Call Routing -> Route/Hunt-> Route Pattern**
2. **Add New**
3. **Route Pattern – 767.@**
4. **Numbering Plan – NANP**
5. **Gateway/Route List – PSTN\_GW**
6. **Call Classification – OffNet**
7. **Calling Line ID Presentation – Restricted**
8. **Calling Name Presentation – Restricted**
9. **Discard Digits - PreDot**

### Cisco CUCM: Caller ID Restricted PSTN Route Pattern (1/3)



### Cisco CUCM: Caller ID Restricted PSTN Route Pattern (2/3)

**Route Pattern Configuration**

Save

**Pattern Definition**

Route Pattern\* 767.@

Route Partition < None >

Description PSTN Restricted Call

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\_GW [\(Edit\)](#)

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 CUCM: Caller ID Restricted PSTN Route Pattern (3/3)

**Calling Party Transformations**

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

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

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 >	

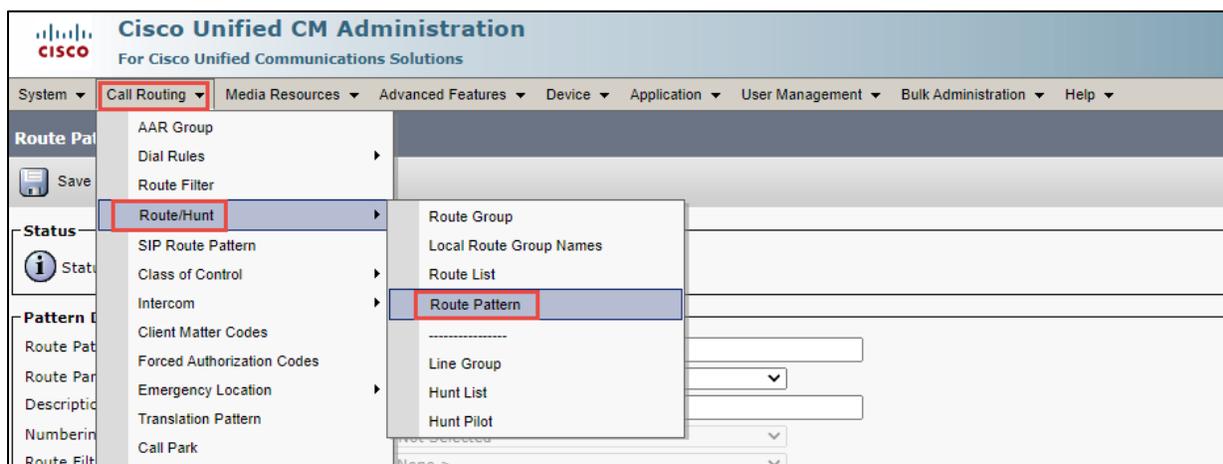
Save

## Voicemail Access - 5555

Route pattern **5555** is used to route the voice mail pilot number 5555 to the Cisco Unity Connection server.

1. Navigate to **Call Routing -> Route/Hunt-> Route Pattern**
2. **Add New**
3. **Route Pattern – 5555**
4. **Gateway/Route List – Unity**

### Cisco CUCM: Cisco Unity Connection Route Pattern (1/3)



### Cisco CUCM: Cisco Unity Connection Route Pattern (2/3)

**Status**

 Status: Ready

---

**Pattern Definition**

Route Pattern\*

Route Partition

Description

Numbering Plan

Route Filter

MLPP Precedence\*

Apply Call Blocking Percentage

Resource Priority Namespace Network Domain

Route Class\*

Gateway/Route List\*  [\(Edit\)](#)

Route Option

Route this pattern

Block this pattern

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

### Cisco CUCM: Cisco Unity Connection Route Pattern (3/3)

**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="&lt; Not Exist &gt;"/>	<input type="text"/>