Configuration Note

AudioCodes Professional Services – Interoperability Lab

Configuring MediaPack™ 1288 Analog Gateway in Cisco Unified Communications Manager Ver. 10.0.1

Version 7.2

audiocodes

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Document Revision Record

LTRT	Description
29301	Initial document release.
29302	Added notice about new limitation related to implementation of MP-1288 as SIP Endpoint.

Documentation Feedback

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1 Introduction

This Configuration Note describes how to set up the AudioCodes MediaPack 1288 Analog Gateway to communicate with the Cisco Unified Communications Manager (CUCM).

1.1 Intended Audience

The document is intended for engineers, or AudioCodes and Cisco Partners who are responsible for installing and configuring Cisco's CUCM and AudioCodes MediaPack 1288 Analog Gateway for enabling VoIP calls.

1.2 About AudioCodes MediaPack 1288 Product

AudioCodes MediaPack 1288 (MP-1288), is a cost-effective best-of-breed, high density analog media voice-over-IP (VoIP) gateway. The device provides superior voice technology for connecting legacy telephones, fax machines and modems with IP-based telephony networks, as well as for integration with IP PBX systems. It is designed and tested to be fully interoperable with leading softswitches, unified communications (UC) servers and SIP proxies.

The device also supports session border controller (SBC) functionality.

The device is designed for carrier environments including 1+1 power supplies and 1+1 Ethernet redundancy, maintaining high voice quality to deliver reliable enterprise VoIP communications. Advanced call routing mechanisms, network voice quality monitoring and survivability capabilities (including PSTN fallback) result in minimum communications downtime.

The device can be deployed for the following applications:

- Enterprise campus deployments
- PSTN emulation for service providers
- Large-scale analog integration with Microsoft Skype for Business environments or other cloud-based or hybrid PBX deployments

2 Component Information

2.1 AudioCodes MP-1288 Version

Table 1: AudioCodes MP-1288 Version

SBC Vendor	AudioCodes
Models	MediaPack 1288
Software Version	SIP_7.20A.156.023
Protocol	SIP/UDP or SIP/TCP
Additional Notes	None

2.2 Cisco CUCM Version

Table 2: Cisco Version

Vendor/Service Provider	Cisco
SSW Model/Service	CUCM
Software Version	10.0.1.11900-2
Protocol	SIP
Additional Notes	None

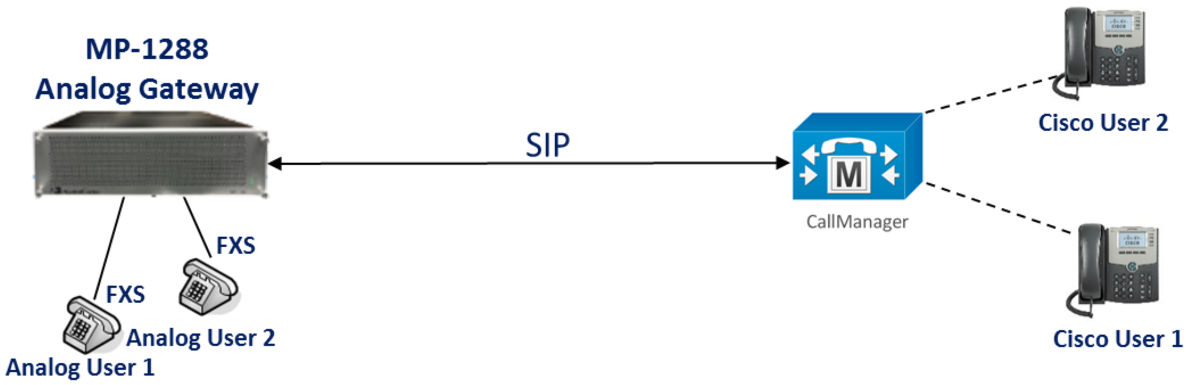
2.3 Interoperability Test Topology

The interoperability testing between AudioCodes MediaPack 1288 Analog Gateway and Cisco CUCM was done using the following topology setup:

- Enterprise Analog PBX (based on AudioCodes MediaPack 1288 Analog Gateway).
- Enterprise wishes to offer its employees enterprise-voice capabilities by connecting the Analog PBX to the Cisco CUCM.

The figure below illustrates this interoperability test topology:

Figure 1: Interoperability Test Topology between MP-1288 and Cisco CUCM



2.3.1 Environment Setup

The interoperability test topology includes the following environment setup:

Table 3: Environment Setup

Area	Setup
Signaling Transcoding	Both MP-1288 and Cisco CUCM can operate with SIP-over-UDP or SIP-over-TCP transport types
Codecs Transcoding	Both MP-1288 and Cisco CUCM support G.711A-law, G.711U-law, and G.729 coder (other coders can be configured)
Media Transcoding	Both MP-1288 and Cisco CUCM operate with RTP media type

2.3.2 Known Limitations

The following limitations were observed in the interoperability tests for the AudioCodes MP-1288 interworking with Cisco's CUCM.

- When MP-1288 implemented as CUCM 3rd party SIP **Device**, this device type in CUCM is unencrypted and allows up to eight DID's to be configured and associated with one phone device. To differentiate phones which are represented by one MP-1288 (due to this Cisco CUCM limitation), each phone should be configured with a different signaling port. And in this case, the MP-1288 is required to configure dedicated SIP Interface for representing **eight** FXS ports.
- When MP-1288 implemented as CUCM 3rd party SIP **Endpoint** (and this is a mandatory configuration when encryption is required), this device type in CUCM allows only 2 DID's to be configured and associated with one phone device. To differentiate phones which are represented by one MP-1288 (due to this Cisco CUCM limitation), each phone should be configured with a different signaling port. In this case, the MP-1288 is required to configure dedicated SIP Interface for representing only **two** FXS ports. Due to this limitation, only 144 FXS ports can be used on the MP-1288.

3 Configuring Cisco CUCM Administration

This section describes how to configure the Cisco Unified CM Administration interface.

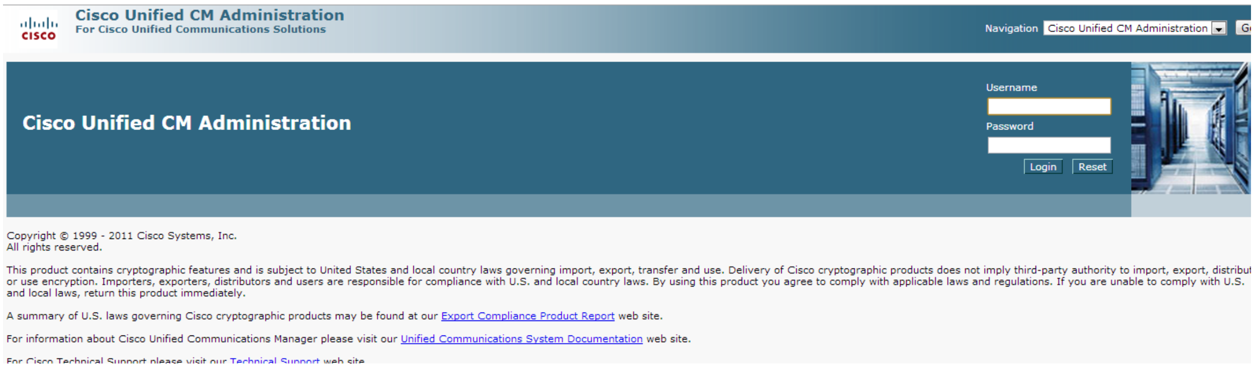
3.1 Log in to Cisco Unified CM Administration

The procedure below describes how to log in to the Cisco CM Unified Administration interface.

To log in to the Cisco Unified CM Administration interface:

1. Log in to the Cisco Unified CM Administration by entering the IP address of the Cisco Unified Communications Manager (CUCM) in the Web browser address field.

Figure 2: Cisco Unified CM Administration



2. In the 'Username' field, enter the user name.
3. In the 'Password' field, enter the password.
4. Click **Login**.

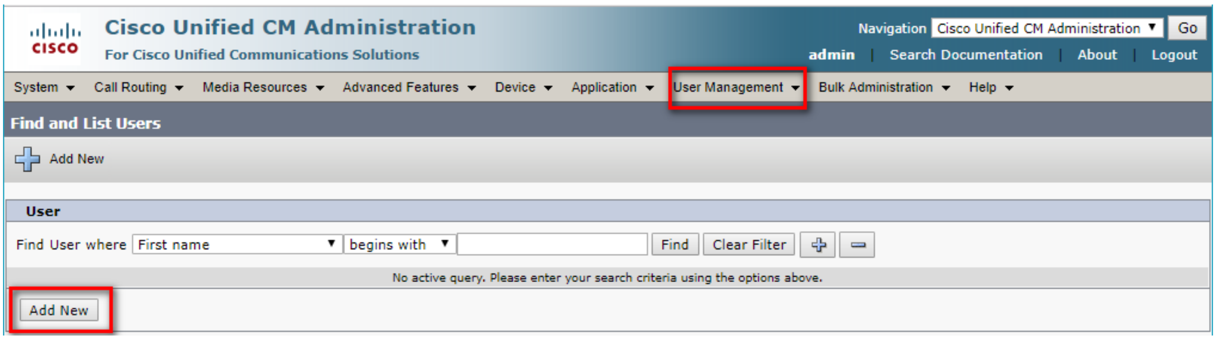
3.2 Add an End User

The procedure below describes how to add an end user in the Cisco CM Unified Administration. In this configuration, the end user is the MP-1288 device.

To add an end user:

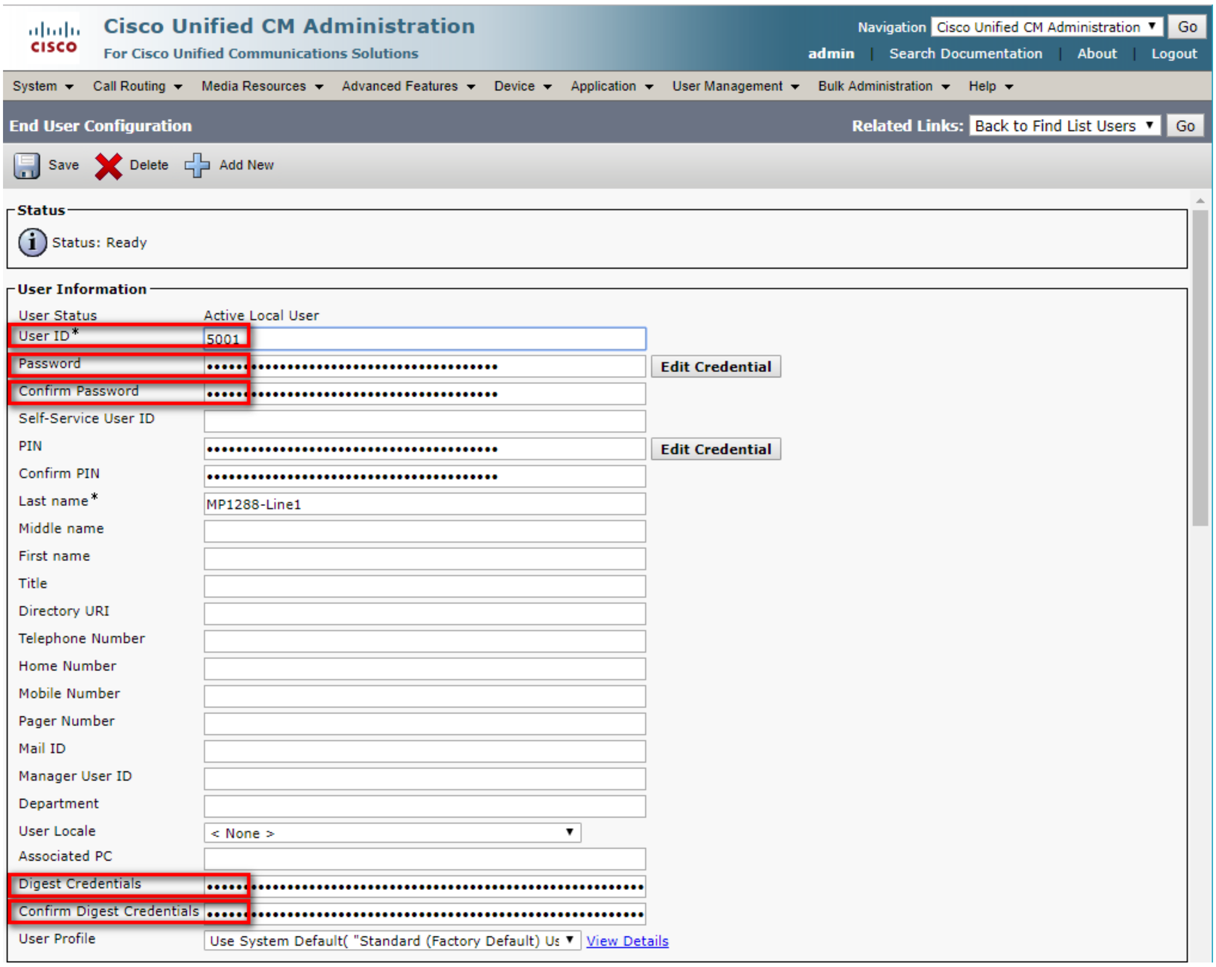
1. Select **User Management** > **End User**, and then click **Add New** to add a new End User.

Figure 3: Add an End User



The following is a screen capture of a typical end user:

Figure 4: Typical End User Configuration



2. Enter the unique end user identification name. You can enter any character, including alphanumeric and special characters. The User ID is the username that should be configured on the MP-1288 Authentication page (see Section 4.9 on page 24). In the example above, the User ID '5001' is configured.
3. In the 'Last name' field, enter the last name. You can enter any character, including alphanumeric and special characters.
4. In the 'Digest Credentials' field, enter Digest Credentials. Cisco Unified Communications Manager uses the digest credentials that you specify here to validate the credentials that the phone offers during digest authentication (e.g., Registration). The Digest Credentials is the password that should be configured on the MP-1288 Authentication page (see Section 4.9 on page 24).
5. Click **Save**.

> Due to the Cisco CUCM limitation (explained in section 2.3.2 above), you need to configure an End User for 2 or up to 8 FXS ports (depending on implementation) on the MP-1288.

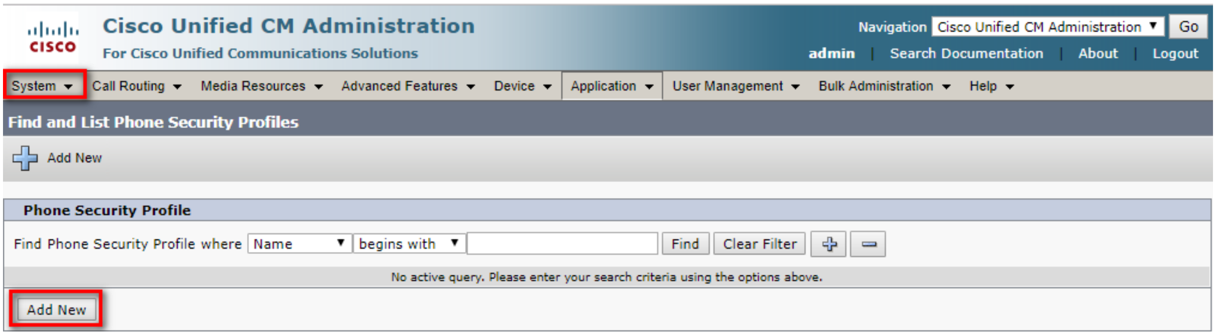
3.3 Configure Phone Security Profile for MP-1288

The procedure below describes how to configure the phone security profile for the MP-1288 device which will communicate with the CUCM.

To add a phone security profile for the MP-1288:

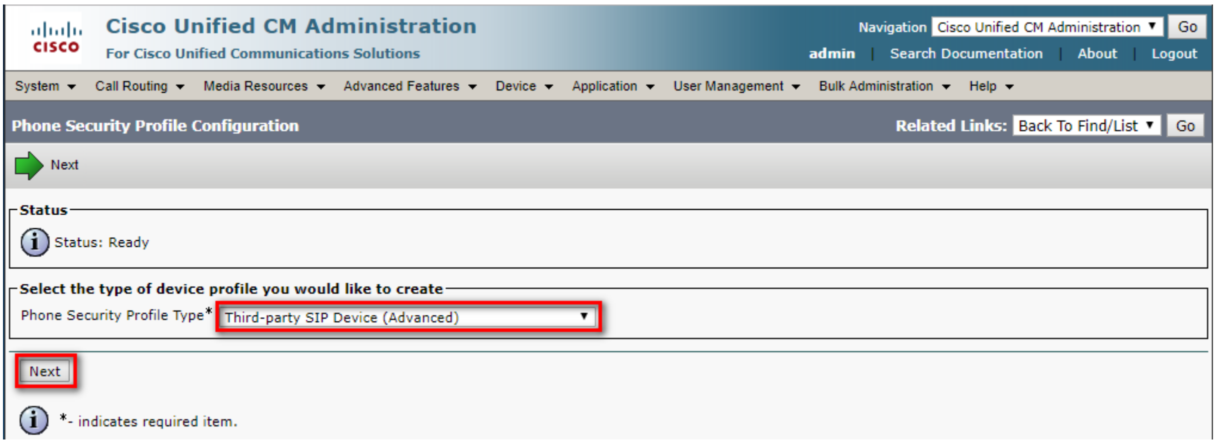
1. Open the Cisco Unified Communications Solutions page.
2. Select **System** > **Security** > **Phone Security Profile**; the 'Find and List Phone Security Profiles' screen is displayed:

Figure 5: Phone Security Profile-Add New



3. Click **Add New**; the 'Phone Security Profile Configuration' screen is displayed:

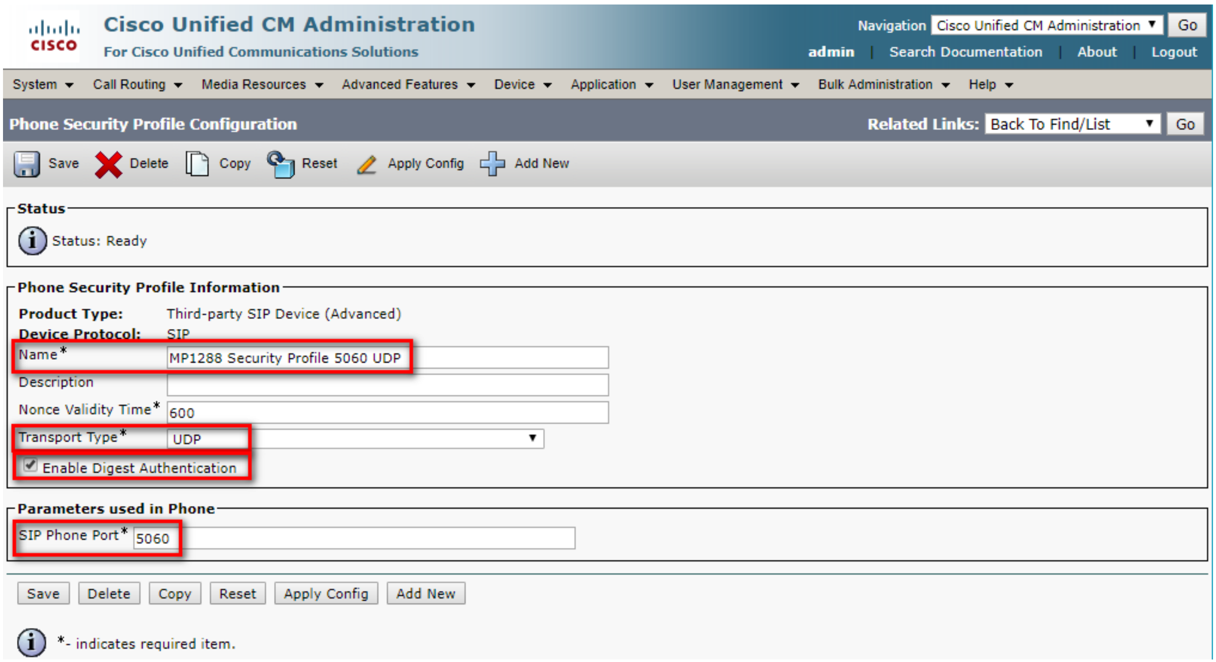
Figure 6: Phone Security Profile Configuration-Device Type



4. From the 'Phone Security Profile Type' drop-down list, select **Third-party SIP Device (Advanced)**, and then click **Next**.

   The Phone Security Profile Information pane is displayed:

Figure 7: Phone Security Profile Configuration - Information



5. In the 'Name' field, enter the name of the Security Profile, i.e., 'MP-1288 Security profile 5060 UDP'.
6. From the 'Transport Type' field, select the appropriate Transport Type, i.e., **UDP**.
7. Select the **Enable Digest Authentication** check box.
8. In the 'SIP Phone Port' field, enter the required port for signaling, i.e., 5060.
9. Click the **Save** button.

In order to differentiate phones which are represented by **one** MP-1288 (explained in section 2.3.2 above), each phone should be configured with a different signaling port.

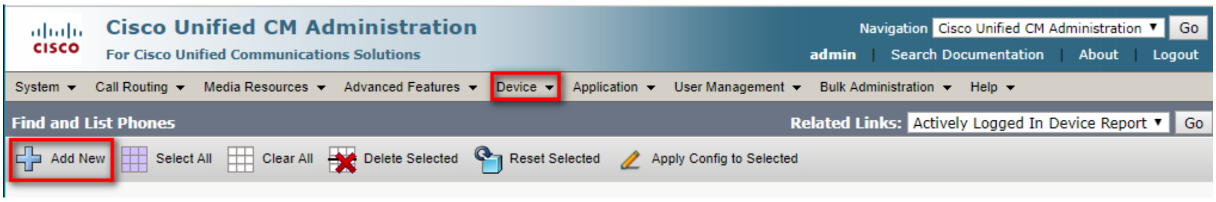
3.4 Configure MP-1288 as Third-Party SIP Device (Advanced)

The procedure below describes how to add the MP-1288 as a third-party SIP device on the CUCM.

To add a third-party SIP device (advanced):

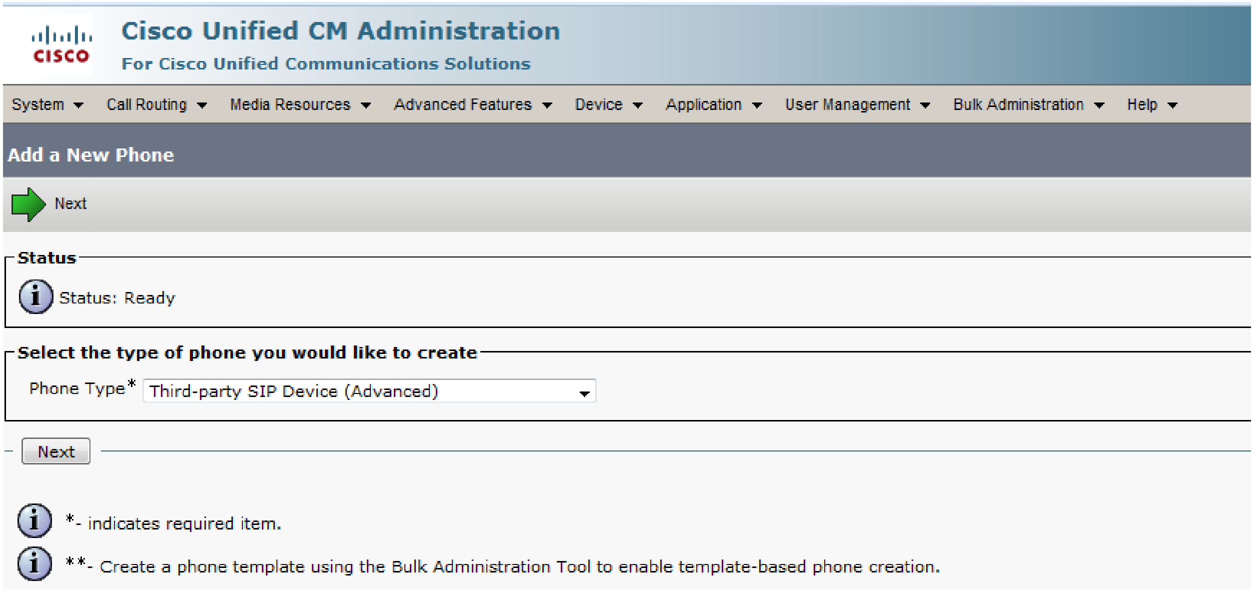
1. Open the Cisco Unified Communications Solutions page.
2. Select **Device** > **Phone**; the 'Find and List Phones' screen is displayed:

Figure 8: Add Third-Party SIP Device



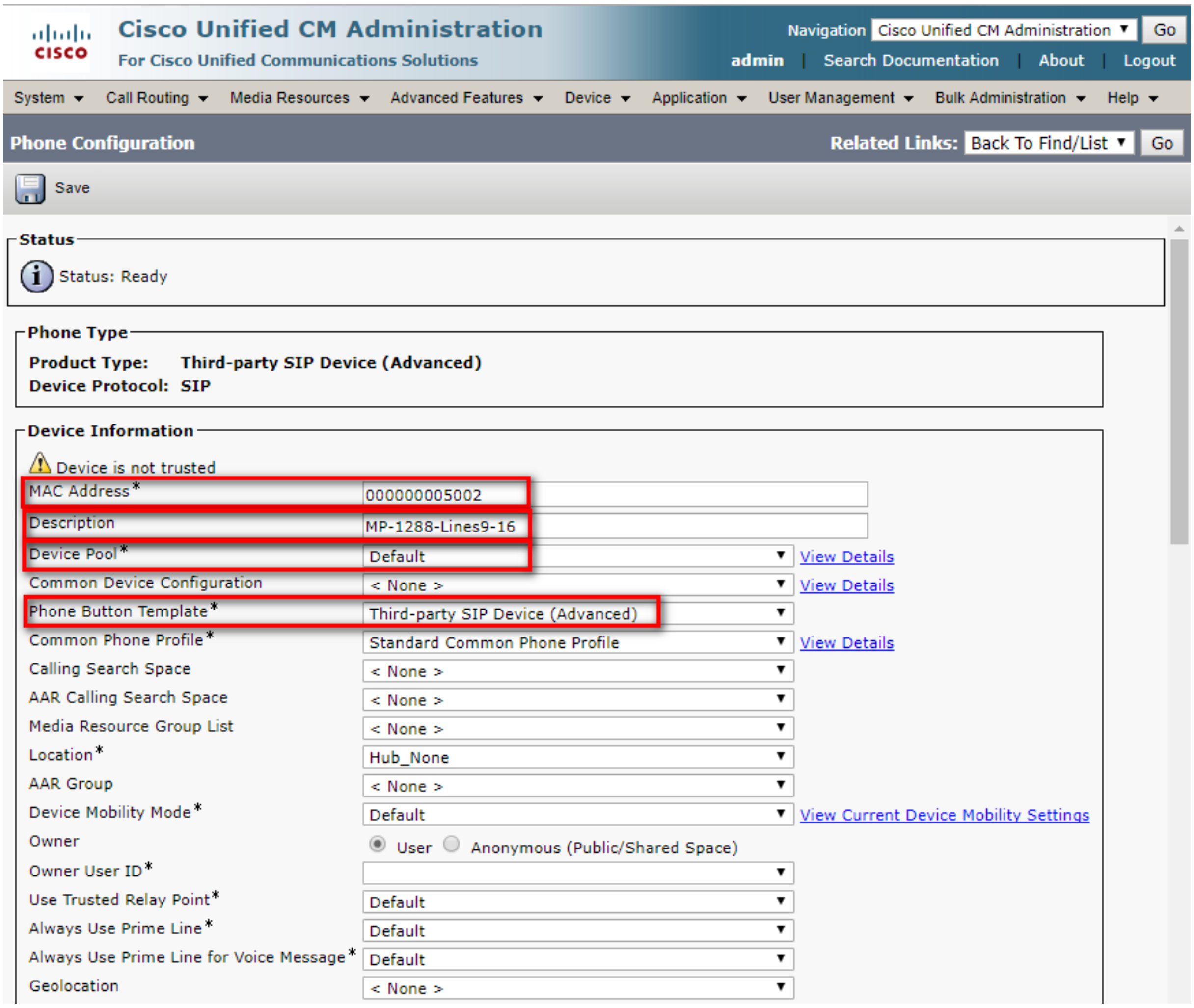
3. Click the **Add New** button to add a new third-party device; the 'Add a New Phone' screen is displayed:

Figure 9: Add a New Phone



4. From the 'Phone Type' drop-down list, select **Third-party SIP Device (Advanced)**, and then click **Next**; the 'Phone Configuration' screen is displayed:

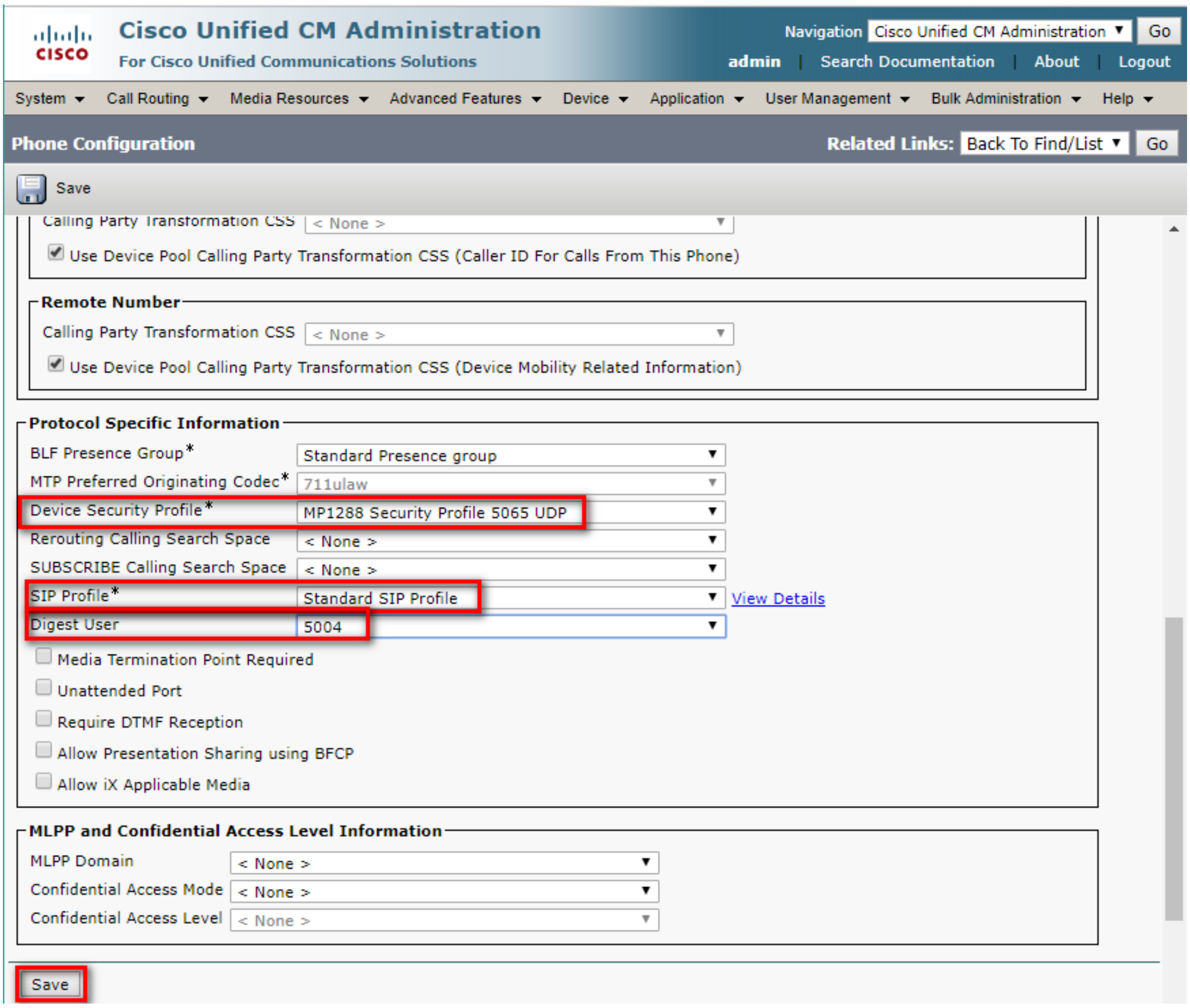
Figure 10: Phone Configuration (1)



5. In the 'MAC Address' field, enter a 12-digit string.
6. In the 'Description' field, enter a short description, i.e., MP-1288-Lines 9-16.
7. From the 'Device Pool' drop-down list, select **Default**.

8. From the 'Phone Button Template' drop-down list, select **Third-party SIP Device (Advanced)**.

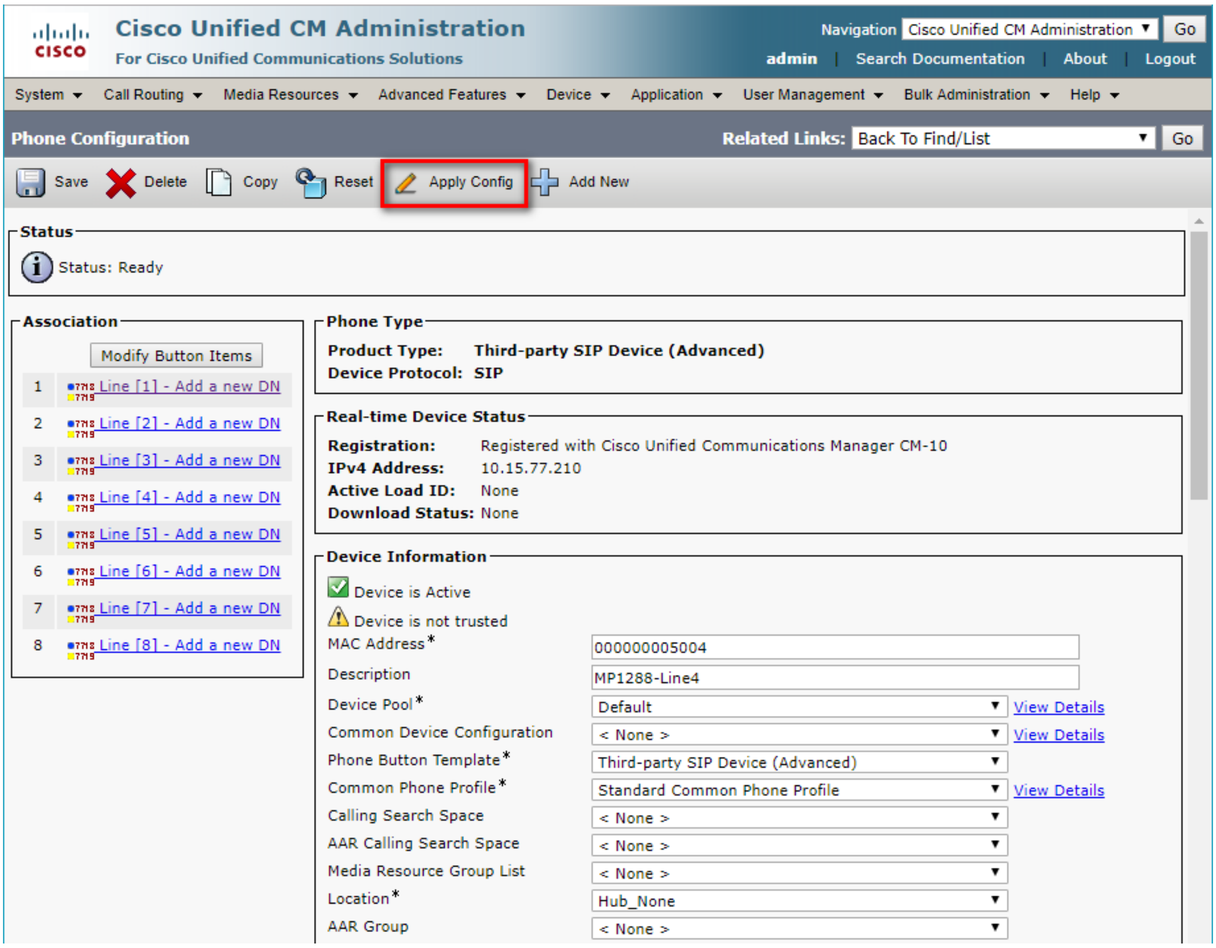
Figure 11: Phone Configuration (2)



9. From the 'Device Security Profile' drop-down list, select **MP-1288 Security Profile 5065 UDP** (the profile is configured in Section 3.3 on page 6).
10. From the 'SIP Profile' drop-down list, select **Standard SIP Profile**.
11. From the 'Digest User' drop-down list, select **5004** (the user is configured in Section 3.2 on page 4).

12. Click **Save**; the device information is displayed:

Figure 12: Apply Config



13. You can configure up to eight phone line connections between the CUCM and the MP-1288 device (for implementation as CUCM 3rd party SIP Device).
14. Click **Apply Config**.

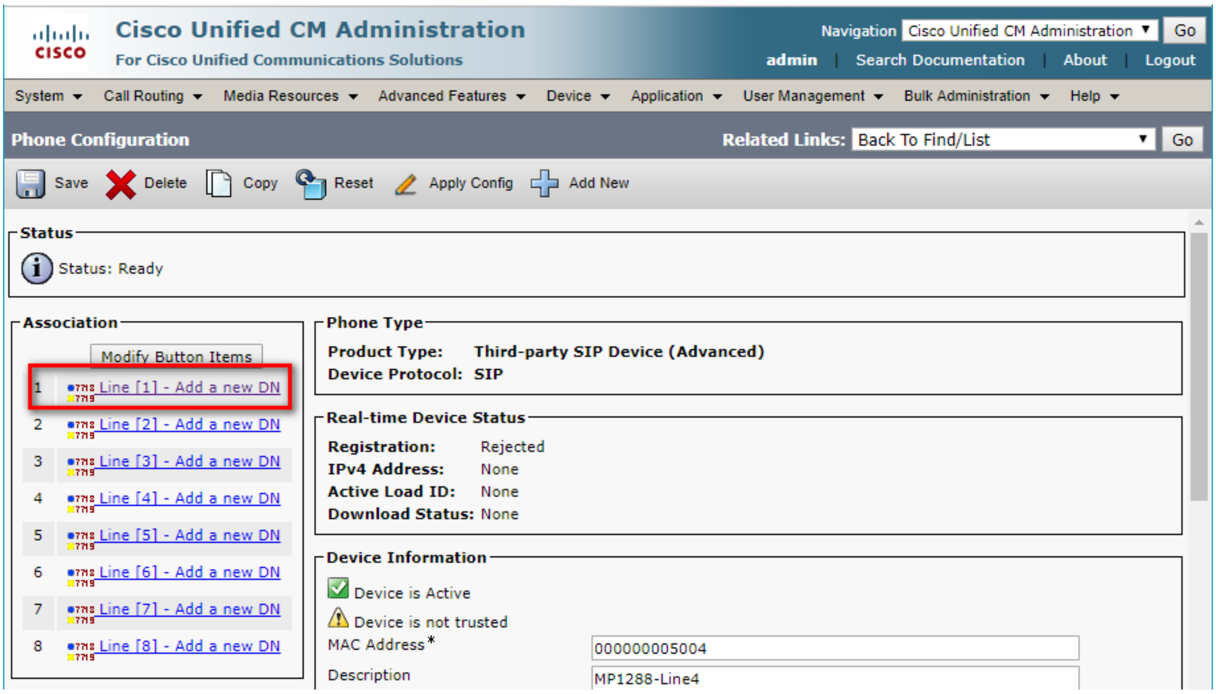
3.5 Configure Directory Number

The procedure below describes how to configure the directory numbers (extension numbers) for communicating between the CUCM and the MP-1288 device.

To add new directory numbers to the Phone device:

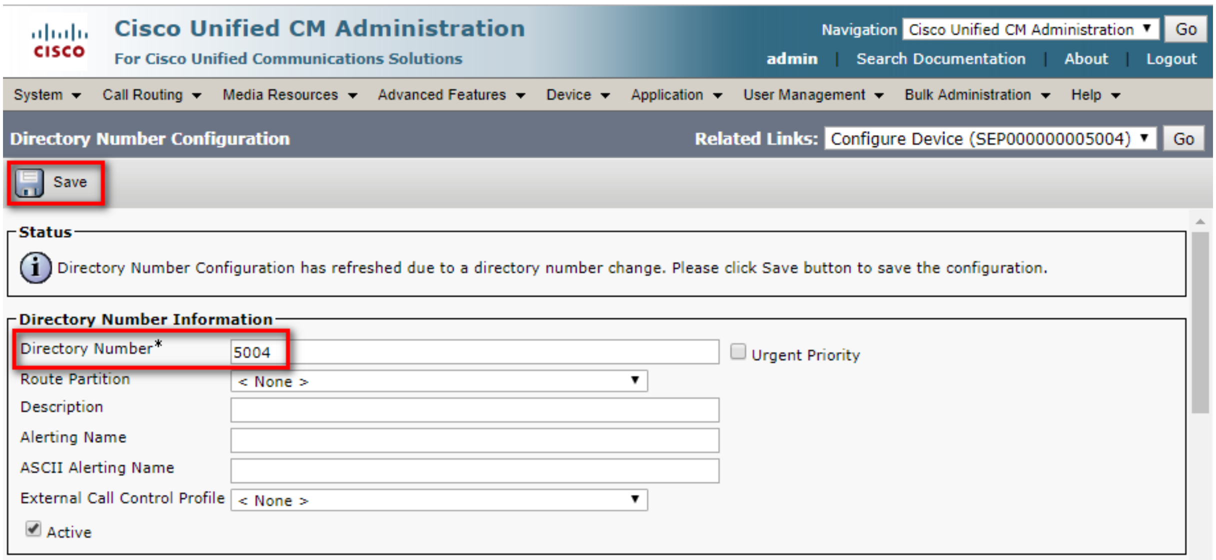
1. Select the '**Add a new DN**' link in the Association part of the Phone Configuration:

Figure 13: Add New Directory Number



2. On the **Directory Number Configuration** page, in the '**Directory Number**' field, enter the extension number that you wish to configure.

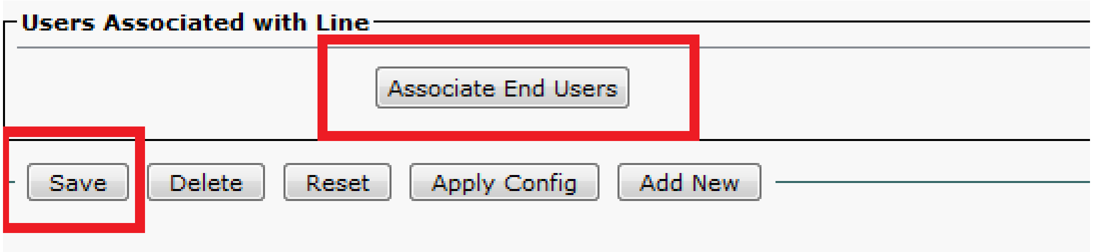
Figure 14: Configure Directory Numbers



3. Click **Save**.

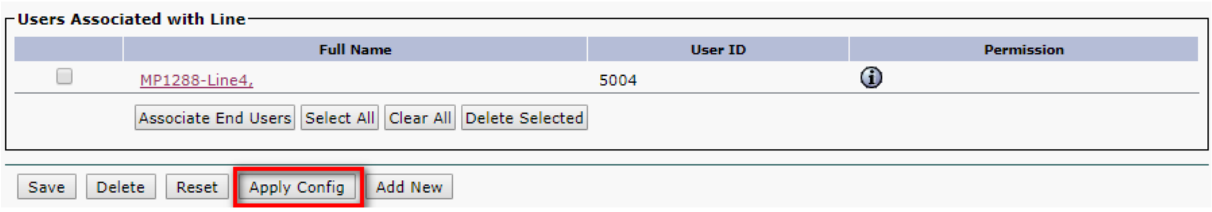
4. Scroll down to the '**Users Associated with Line**' pane.

Figure 15: Associate Line with End User



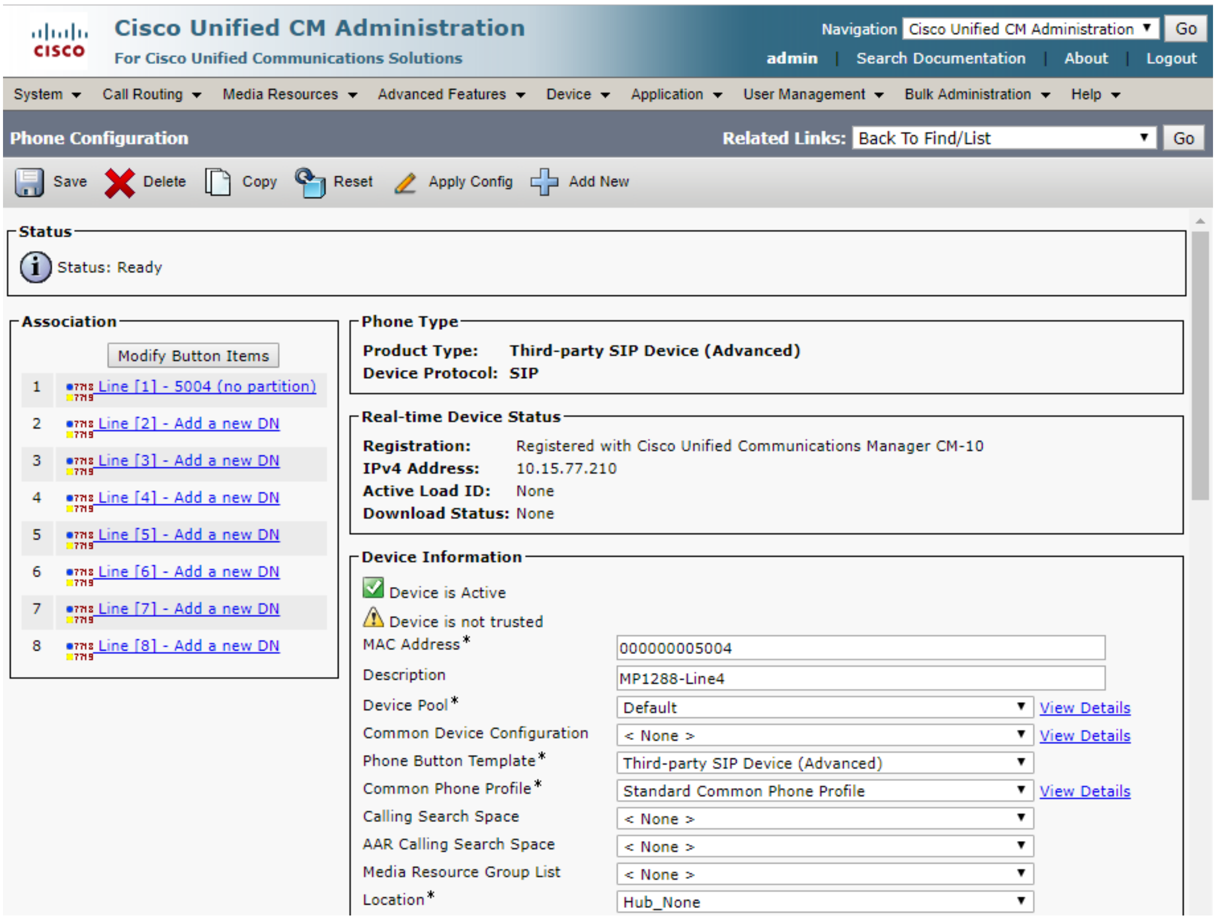
5. Click **Associate End Users** to associate this line with the end user, created in Section 3.2 on page 4.
6. Click **Save**.
7. Click **Apply Config**.

Figure 16: Apply Configuration



The configuration is displayed in the following screen:

Figure 17: Phone Configuration



8. Repeat the above steps for each extension number that you wish to configure.

- You can configure up to eight directory numbers (phone numbers for the endpoints) for implementation of MP-1288 as CUCM 3rd party SIP Device or up to two telephony numbers for implementation of MP-1288 as CUCM 3rd party SIP Endpoint.
  NOTE: only 144 FXS MP-1288 ports can be used for implementation as SIP Endpoint.
- Each phone number extension that you configure in this section should also be configured in the Endpoint Phone Number Table on the MP-1288 (see section 4.5 Configure Endpoint Phone Numbers, on page 21).

4 Configuring AudioCodes MP-1288

This section provides step-by-step procedures on how to configure the AudioCodes MP-1288 Analog Gateway to communicate with the Cisco CUCM.

4.1 Configure SIP Signaling Interfaces

The procedure below describes how to configure SIP Interfaces. As was mentioned above, due to the Cisco CUCM limitation of the analog ports (two or up to eight ports depending on implementation) can be associated with one phone device, each phone should be configured with a different signaling port. In the MP-1288 signaling ports are configured in the SIP Interface Table. So, for the interoperability between MP-1288 and Cisco CUCM, the SIP Interface with a different port must be configured for each of the analog ports (two or up to eight ports depending on implementation).

To configure SIP Interfaces:

1. Open the SIP Interfaces table (**Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **SIP Interfaces**).
2. Add a SIP Interface. You can use the default SIP Interface (Index 0), but modify it as shown below:

Parameter	Value
Index	**0**
Name	**SIPInterface_5060**
Network Interface	**Voice**
Application Type	**GW**
UDP Port	**5060**
TCP Port	**0**
TLS Port	**0**

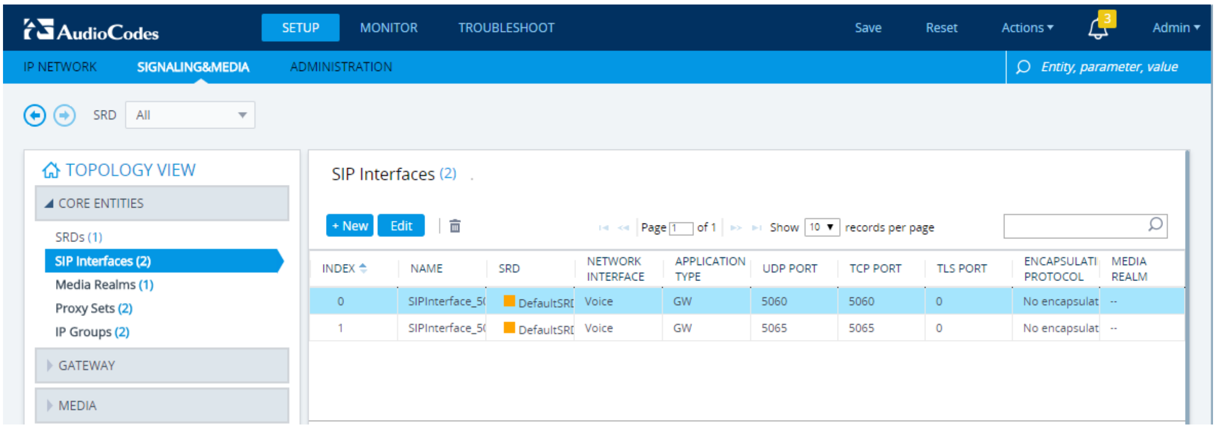
3. Configure a SIP Interface for Port 5065:

Parameter	Value
Index	**1**
Name	**SIPInterface_5065**
Network Interface	**WAN_IF**
Application Type	**Voice**
Application Type	**GW**
UDP Port	**5065**
TCP Port	**0**
TLS Port	**0**

4. Repeat the above steps for each of the eight analog ports that you wish to configure.

The configured SIP Interfaces are shown in the figure below:

Figure 18: Configured SIP Interfaces in SIP Interface Table



4.2 Configure Proxy Sets

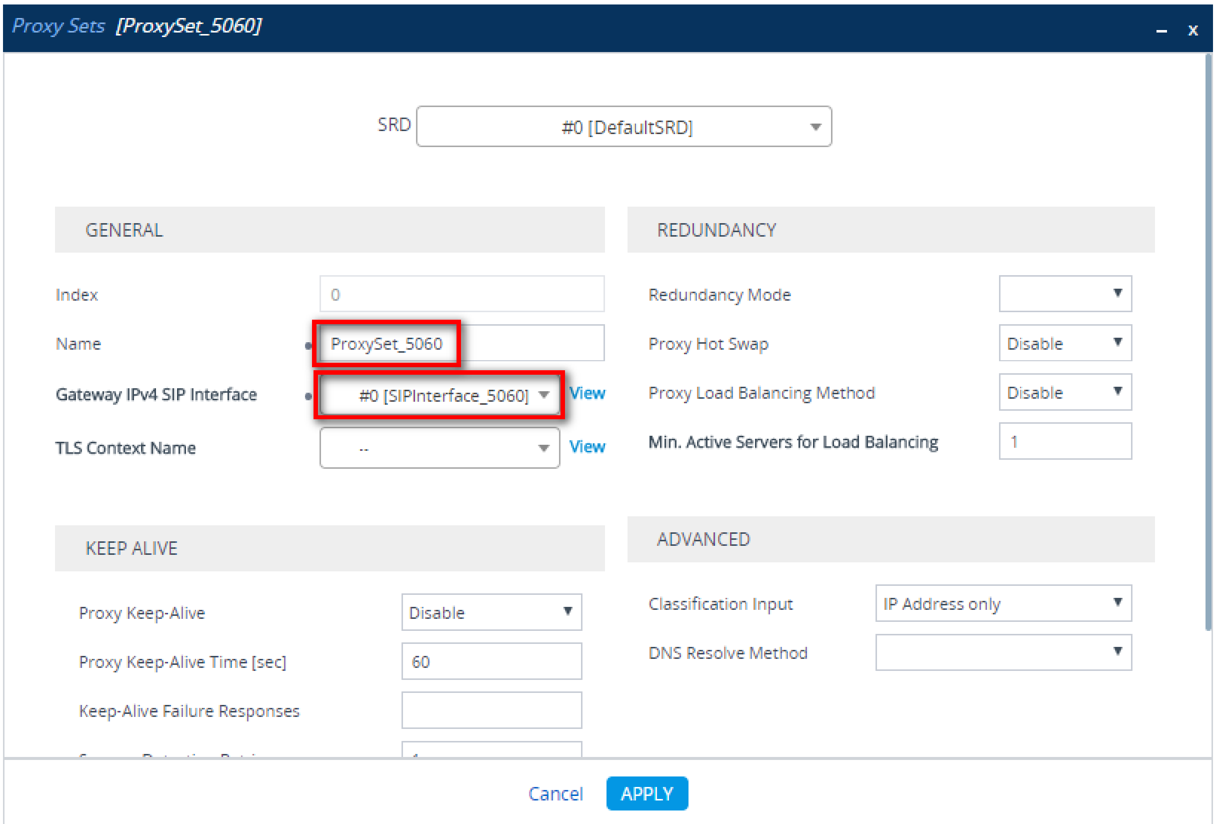
The procedure below describes how to configure Proxy Sets. The Proxy Set defines the destination address (IP address or FQDN) of the Cisco CUCM server. The Proxy Sets will be later applying to the VoIP network by assigning them to IP Groups.

To configure Proxy Sets:

1. Open the Proxy Sets table (**Setup** menu > **Signaling & Media** tab > **Core Entities** folder >**Proxy Sets**).
2. Add a Proxy Set for the Cisco CUCM. You can use the default Proxy Set (Index 0), but modify it as shown below:

Parameter	Value
Index	**0**
Name	**ProxySet_5060**
SBC IPv4 SIP Interface	**SIPInterface_5060**

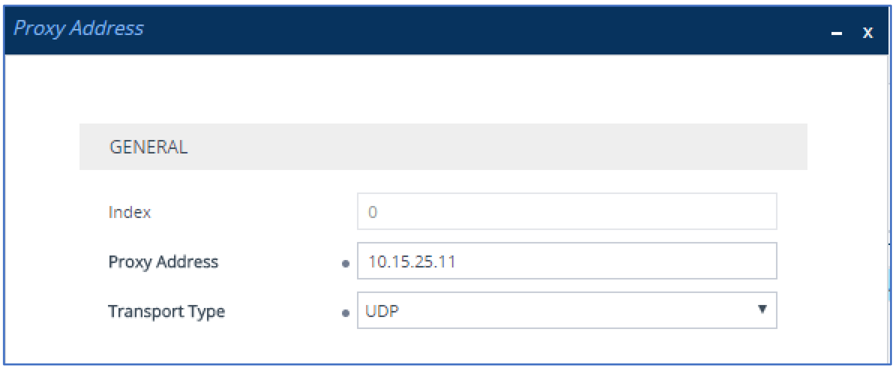
Figure 19: Configuring Proxy Set for port 5060 toward Cisco CUCM



a. Select the Index row of the Proxy Set that you added, and then click the **Proxy Address** link located below the table on the Proxy Sets page; the Proxy Address table opens.

b. Click **New**; the following screen appears:

Figure 20: Configuring Proxy Address for Port 5060 Towards Cisco CUCM



c. Configure the address of the Proxy Set according to the parameters described in the table below.

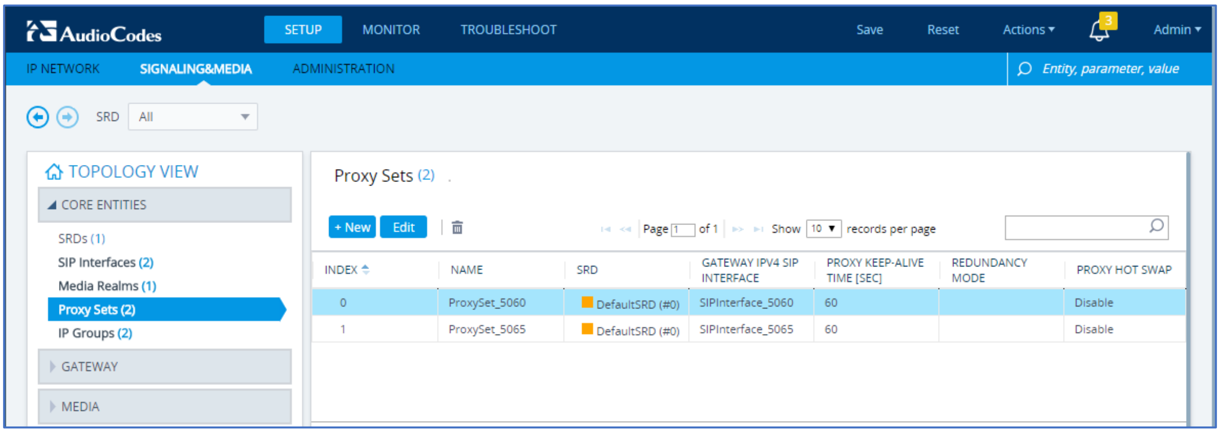
d. Click **Apply**.

Parameter	Value
Index	**0**
Proxy Address	**10.15.25.11** (Cisco CUCM IP address)
Transport Type	**UDP**

3. Repeat the above steps for each signaling port (different port for each analog port) that you wish to configure.

The configured Proxy Sets are shown in the figure below:

Figure 21: Configured Proxy Sets in Proxy Sets Table



4.3 Configure IP Groups

The procedure below describes how to configure IP Groups. The IP Group represents Cisco CUCM. It is associated with a Proxy Set.

To configure IP Groups:

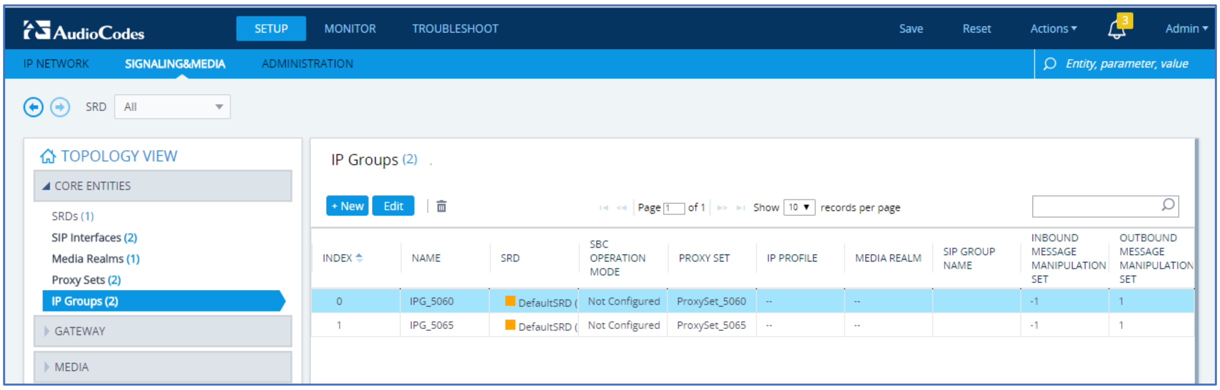
1. Open the IP Groups table (**Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **IP Groups**).
2. Add an IP Group for port 5060 toward Cisco CUCM. You can use the default IP Group (Index 0), but modify it as shown below:

Parameter	Value
Index	**0**
Name	**IPG_5060**
Type	**Server**
Proxy Set	**ProxySet_5060**

3. Repeat the above steps for each signaling port (different for each of the analog ports) that you wish to configure.

The configured IP Groups are shown in the figure below:

Figure 22: Configured IP Groups in IP Group Table



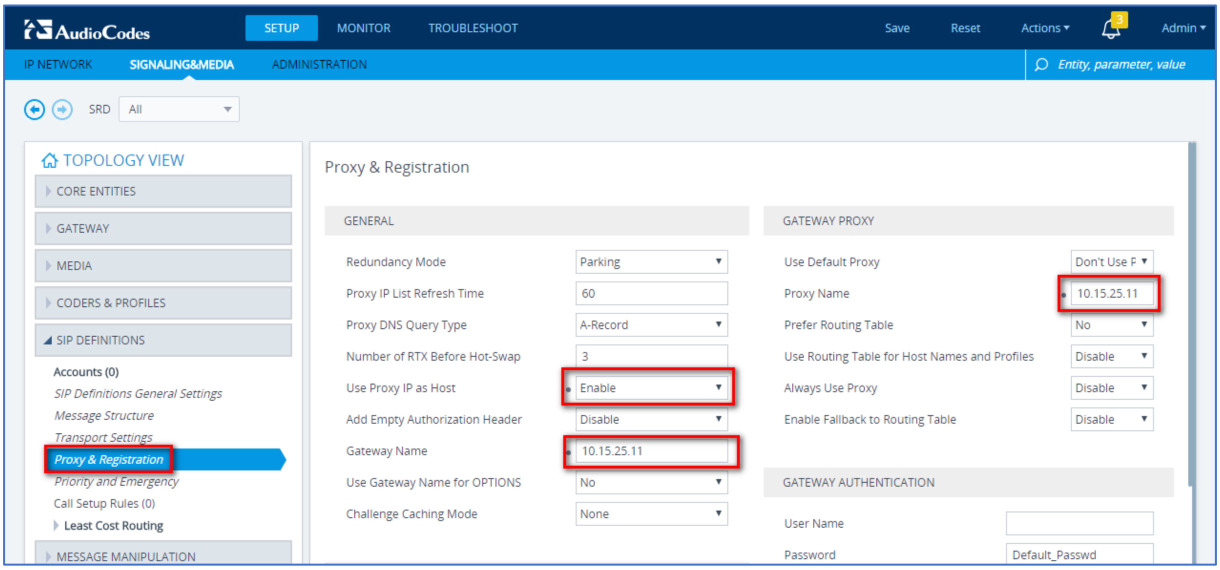
4.4 Configure Proxy Server and Registration

The procedure below describes how to configure the Proxy Server (Cisco CUCM) and registration parameters.

To configure the Proxy and Registration parameters:

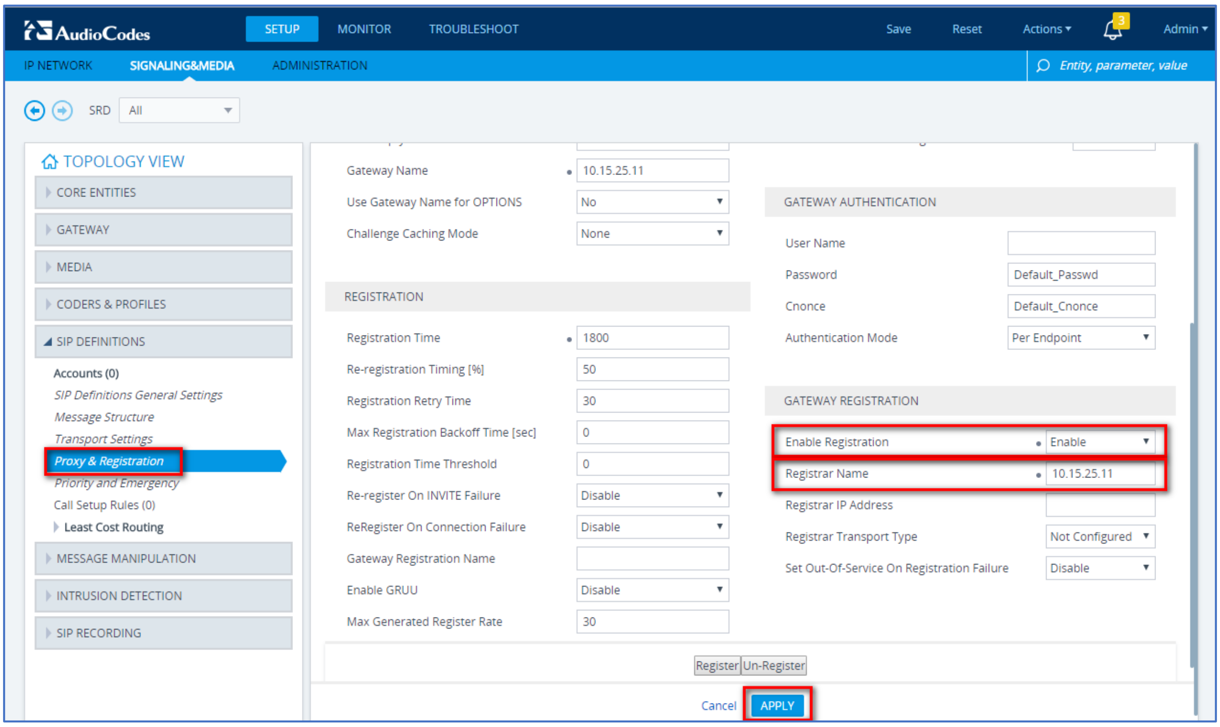
1. Open the Proxy & Registration page (**Setup menu > Signaling & Media tab > SIP Definitions folder > Proxy & Registration**).

Figure 23: Proxy



2. From the 'Use Proxy IP as Host' drop-down list, select **Enable**.
3. In the 'Gateway Name' field, enter the CUCM IP address.
4. In the 'Proxy Name' field, enter the CUCM IP address.

Figure 24: Registration



5. From the 'Enable Registration' drop-down list, select **Enable**.
6. In the 'Registrar Name' field, enter the CUCM IP address.
7. Click the **Apply** button.

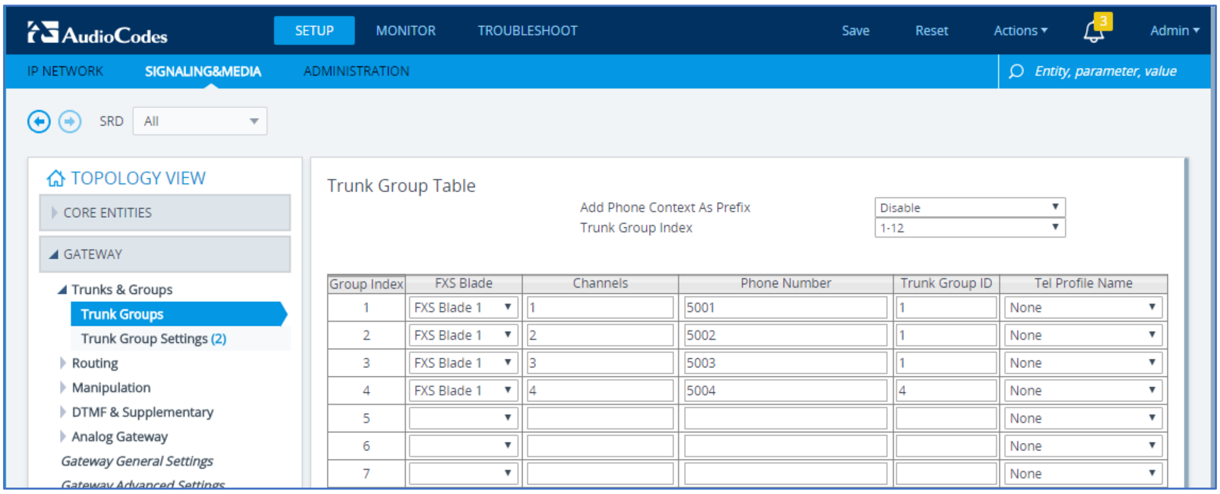
4.5 Configure Endpoint Phone Numbers

The procedure below describes the configuration of the MP-1288 channels, which includes assigning them to Trunk Groups. A Trunk Group is a logical group of physical trunks and channels. A Trunk Group can include multiple trunks and ranges of channels. To enable and activate the channels of the device, Trunk Groups need to be configured and assigned with telephone numbers. Channels that are not configured in this table are disabled.

To configure a Trunk Group:

1. Open the Trunk Group table (**Setup** menu > **Signaling & Media** tab > **Gateway** folder > **Trunks & Groups** > **Trunk Groups**).

Figure 25: Endpoint Phone Number Table



2. In the 'Phone Number' fields, enter the directory numbers that you configured on the Cisco lines (see Section 3.5 on page 12).
3. In the 'Trunk Group ID' fields, enter "**1**" for first eight numbers and increment it for each of the eight Trunk Groups.
4. Click **Apply**.

4.6 Configure Trunk Group Settings

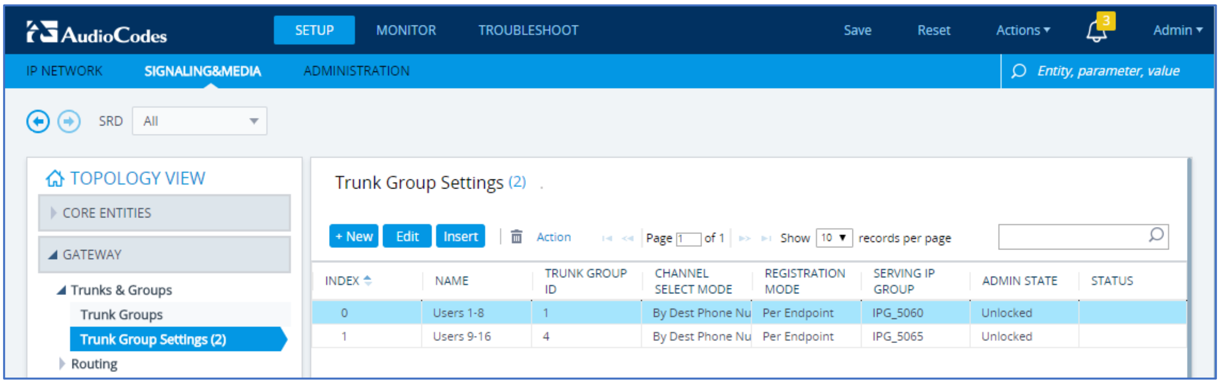
The procedure below describes how to configure the Trunk Group Settings Table. The main configuration includes the following:

- Channel select method, which defines how the device allocates incoming IP-to-Tel calls to the channels of a Trunk Group.
- Registration method for registering Trunk Groups to remote IP servers (*Serving IP Group*).

To configure Trunk Group settings:

1. Open the Trunk Group Settings table (**Setup** menu > **Signaling & Media** tab > **Gateway** folder > **Trunks & Groups** > **Trunk Group Settings**).

Figure 26: Trunk Group Settings



2. Configure the entry as shown in the screen above. For each Trunk Group configure:
   a. ‘Channel Select Mode’ as **By Dest Phone Number**
   b. ‘Registration Mode’ as **Per Endpoint**
   c. ‘Serving IP Group’ the IP Group, configured in Section 3.5 on page 19 above.

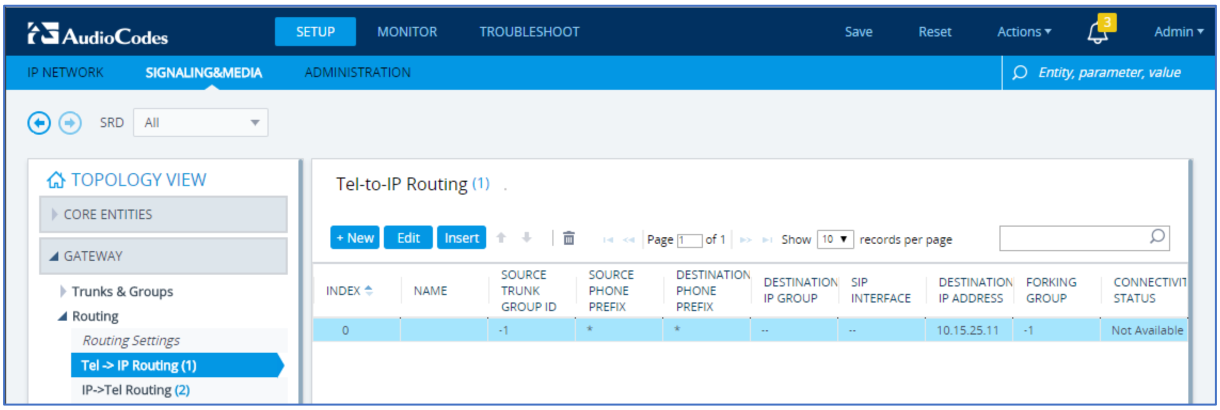
4.7 Configure Tel-to-IP Routing

The procedure below describes how to configure routing rules that are used to route calls from the Tel side to an IP destination (Cisco CUCM).

To configure Tel-to-IP routing:

1. Open the Tel-to-IP Routing table (**Setup** menu > **Signaling & Media** tab > **Gateway** folder > **Routing** > **Tel->IP Routing**).
2. Click **New.**

Figure 27: Tel-to-IP Routing Table



3. Configure the entry as shown in the screen above (to send all messages from Tel to Cisco CUCM).
4. Click **Apply**.

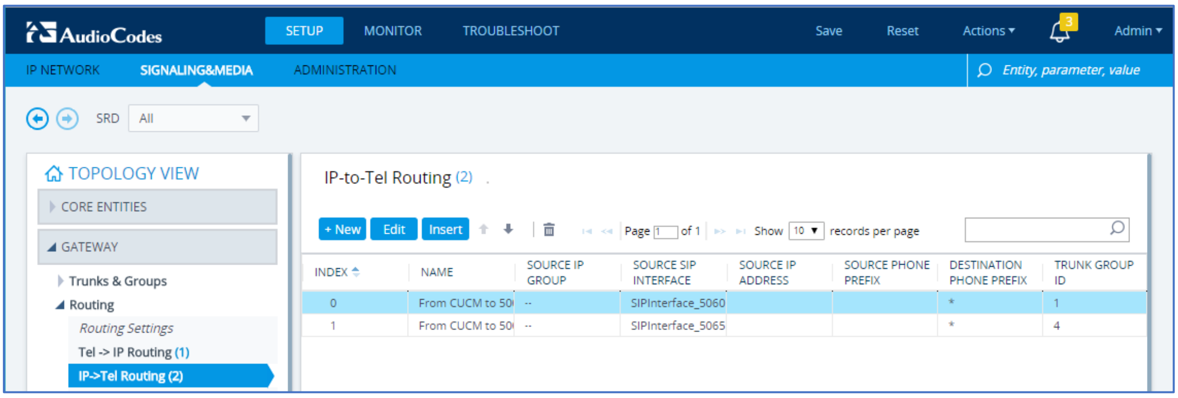
4.8 Configure IP-to-Tel Routing

The procedure below describes how to configure routing rules are used to route incoming IP calls from Cisco CUCM to Trunk Groups.

To configure IP-to-Tel routing:

1. Open the IP-to-Tel Routing table (**Setup** menu > **Signaling & Media** tab > **Gateway** folder > **Routing** > **IP->Tel Routing**).
2. Click **New.**

Figure 28: IP-to-Tel Routing Table



3. Configure the entry as shown in the screen above (this sends all messages from a specific SIP Interface to the appropriate Trunk Group).
4. Click **Apply**.

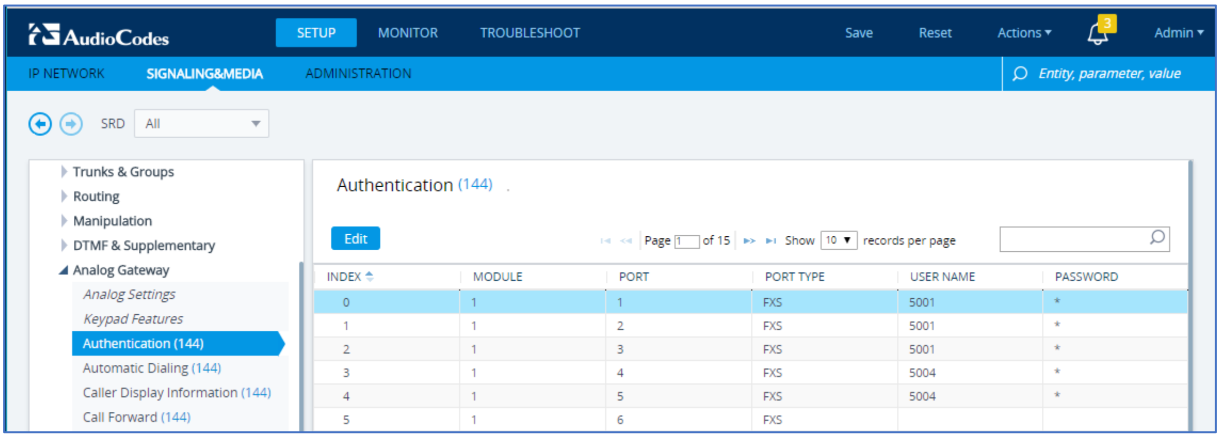
4.9 Configure End User Authentication

The procedure below describes how to configure the end user authentication. The Authentication table lets you configure an authentication user name and password per FXS port

To configure authentication credentials per port:

1. Open the Authentication page (**Setup** menu > **Signaling & Media** tab > **Gateway** folder > **Analog Gateway** > **Authentication**).

Figure 29: Authentication Table



2. Configure the username and password according to the CUCM end user credentials (see section 3.3 Configure Phone Security Profile for MP-1288, on page 6).
3. Click **Apply**.

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