

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

Version 7.2



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## Notice

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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

AudioCodes continually strives to produce high quality documentation. If you have any comments (suggestions or errors) regarding this document, please fill out the Documentation Feedback form on our Web site at <https://online.audiocodes.com/documentation-feedback>.

# 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**

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

### 2.2 Cisco CUCM Version

**Table 2: Cisco Version**

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

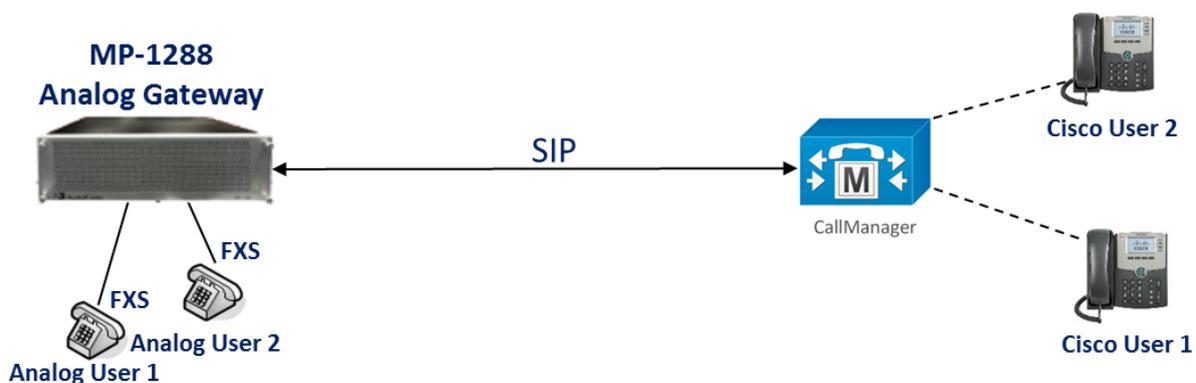
### 2.3 Interoperability Test Topology

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
<b>Signaling Transcoding</b>	Both MP-1288 and Cisco CUCM can operate with SIP-over-UDP or SIP-over-TCP transport types
<b>Codecs Transcoding</b>	Both MP-1288 and Cisco CUCM support G.711A-law, G.711U-law, and G.729 coder (other coders can be configured)
<b>Media Transcoding</b>	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 3<sup>rd</sup> 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**.

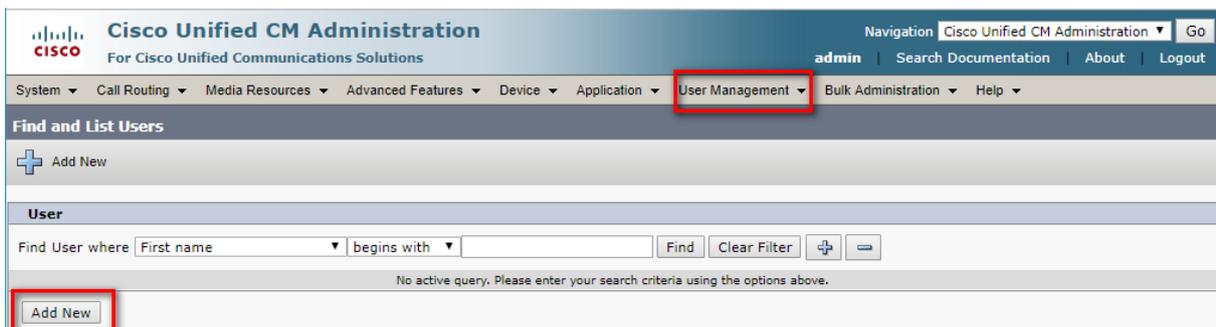
### 3.2 Add an End User

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**

The screenshot shows the Cisco Unified CM Administration interface for configuring an end user. The page title is "End User Configuration" and the user status is "Ready". The "User Information" section is visible, with several fields highlighted in red:

- User ID\*: 5001
- Password: [Redacted]
- Confirm Password: [Redacted]
- Digest Credentials: [Redacted]
- Confirm Digest Credentials: [Redacted]

Other fields include: Self-Service User ID, PIN, Confirm PIN, Last name\* (MP1288-Line1), Middle name, First name, Title, Directory URI, Telephone Number, Home Number, Mobile Number, Pager Number, Mail ID, Manager User ID, Department, User Locale (< None >), Associated PC, and User Profile (Use System Default('Standard (Factory Default) Us)).

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

The screenshot shows the Cisco Unified CM Administration interface. The top navigation bar includes the Cisco logo, the title 'Cisco Unified CM Administration', and the user role 'admin'. Below the navigation bar, there is a menu with 'System' highlighted in red. The main content area is titled 'Find and List Phone Security Profiles' and contains a search bar with a dropdown for 'Name' and a 'begins with' dropdown. There are buttons for 'Find', 'Clear Filter', and 'Add New'. The 'Add New' button is highlighted in red.

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

Figure 6: Phone Security Profile Configuration-Device Type

The screenshot shows the Cisco Unified CM Administration interface for the 'Phone Security Profile Configuration' screen. The top navigation bar is the same as in Figure 5. Below the navigation bar, there is a 'Status' section showing 'Status: Ready'. The main content area is titled 'Phone Security Profile Configuration' and contains a 'Next' button highlighted in green. Below this, there is a section titled 'Select the type of device profile you would like to create' with a dropdown menu. The dropdown menu has 'Third-party SIP Device (Advanced)' selected, which is highlighted in red. Below this, there is a 'Next' button highlighted in red.

- 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**

The screenshot shows the Cisco Unified CM Administration interface. The main heading is "Phone Security Profile Configuration". Below the heading, there are navigation tabs: System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The "Device" tab is selected. The "Phone Security Profile Configuration" pane is active, showing the following information:

- Status:** Ready
- Phone Security Profile Information:**
  - Product Type: Third-party SIP Device (Advanced)
  - Device Protocol: SIP
  - Name\*: MP1288 Security Profile 5060 UDP
  - Description: (empty)
  - Nonce Validity Time\*: 600
  - Transport Type\*: UDP
  - Enable Digest Authentication
- Parameters used in Phone:**
  - SIP Phone Port\*: 5060

At the bottom of the pane, there are buttons for Save, Delete, Copy, Reset, Apply Config, and Add New. A note indicates that an asterisk (\*) denotes a required item.

- In the 'Name' field, enter the name of the Security Profile, i.e., 'MP-1288 Security profile 5060 UDP'.
- From the 'Transport Type' field, select the appropriate Transport Type, i.e., **UDP**.
- Select the **Enable Digest Authentication** check box.
- In the 'SIP Phone Port' field, enter the required port for signaling, i.e., 5060.
- 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):

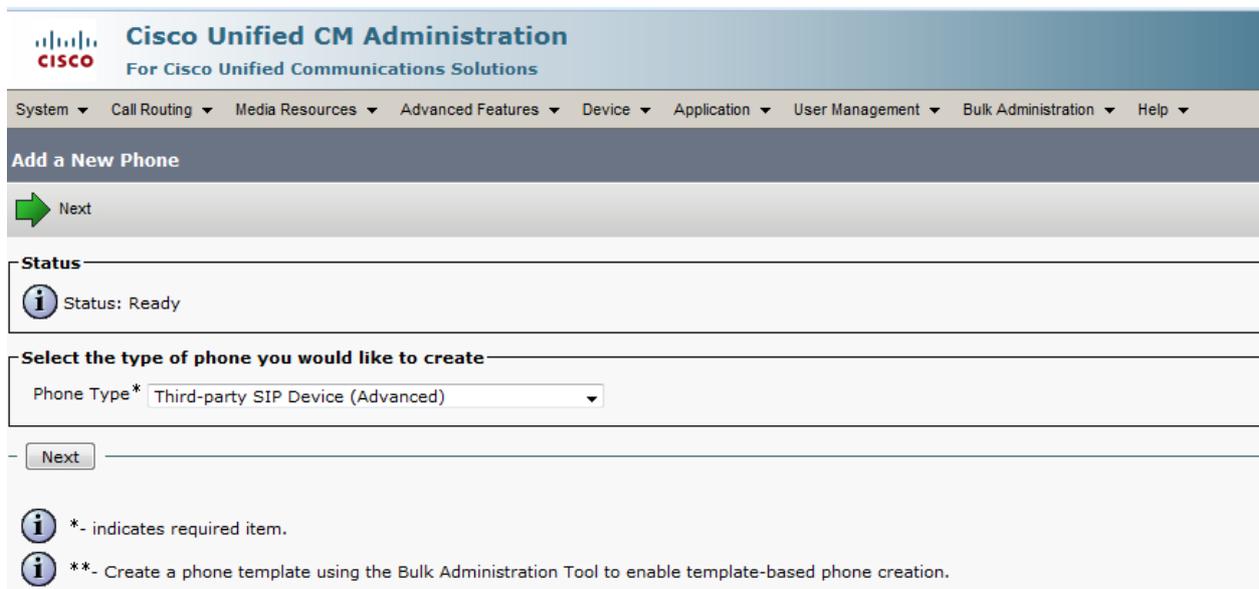
1. Open the Cisco Unified Communications Solutions page.
2. Select **Device > Phone**; the 'Find and List Phones' screen is displayed:

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

The screenshot shows the 'Add a New Phone' configuration page. At the top, there is a green arrow labeled 'Next'. Below this is a 'Status' section with an information icon and the text 'Status: Ready'. The main section is titled 'Select the type of phone you would like to create'. It contains a dropdown menu for 'Phone Type\*' which is currently set to 'Third-party SIP Device (Advanced)'. Below the dropdown is another 'Next' button. At the bottom, there are two information icons with explanatory text: '\*- indicates required item.' and '\*\*- Create a phone template using the Bulk Administration Tool to enable template-based phone creation.'

- 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)

The screenshot shows the 'Phone Configuration' page in Cisco Unified CM Administration. The 'Phone Type' is 'Third-party SIP Device (Advanced)'. The 'Device Information' section contains the following fields:

Field	Value
MAC Address*	000000005002
Description	MP-1288-Lines9-16
Device Pool*	Default
Common Device Configuration	< None >
Phone Button Template*	Third-party SIP Device (Advanced)
Common Phone Profile*	Standard Common Phone Profile
Calling Search Space	< None >
AAR Calling Search Space	< None >
Media Resource Group List	< None >
Location*	Hub_None
AAR Group	< None >
Device Mobility Mode*	Default
Owner	<input checked="" type="radio"/> User <input type="radio"/> Anonymous (Public/Shared Space)
Owner User ID*	
Use Trusted Relay Point*	Default
Always Use Prime Line*	Default
Always Use Prime Line for Voice Message*	Default
Geolocation	< None >

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

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

**Figure 11: Phone Configuration (2)**

The screenshot displays the 'Phone Configuration' page in Cisco Unified CM Administration. The page is titled 'Phone Configuration' and includes a 'Save' button at the top left. The configuration is organized into several sections:

- Calling Party Transformation CSS:** A dropdown menu set to '< None >'. Below it is a checked checkbox for 'Use Device Pool Calling Party Transformation CSS (Caller ID For Calls From This Phone)'.
- Remote Number:** A dropdown menu set to '< None >'. Below it is a checked checkbox for 'Use Device Pool Calling Party Transformation CSS (Device Mobility Related Information)'.
- Protocol Specific Information:** This section contains several dropdown menus and checkboxes:
  - BLF Presence Group\*: Standard Presence group
  - MTP Preferred Originating Codec\*: 711ulaw
  - Device Security Profile\*: MP1288 Security Profile 5065 UDP
  - Rerouting Calling Search Space: < None >
  - SUBSCRIBE Calling Search Space: < None >
  - SIP Profile\*: Standard SIP Profile (with a 'View Details' link)
  - Digest User: 5004
  - Media Termination Point Required:
  - Unattended Port:
  - Require DTMF Reception:
  - Allow Presentation Sharing using BFCP:
  - Allow iX Applicable Media:
- MLPP and Confidential Access Level Information:** This section contains three dropdown menus:
  - MLPP Domain: < None >
  - Confidential Access Mode: < None >
  - Confidential Access Level: < None >

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

- 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).
- From the 'SIP Profile' drop-down list, select **Standard SIP Profile**.
- From the 'Digest User' drop-down list, select **5004** (the user is configured in Section 3.2 on page 4).

- Click **Save**; the device information is displayed:

**Figure 12: Apply Config**

The screenshot shows the Cisco Unified CM Administration web interface. At the top, the navigation bar includes 'Cisco Unified CM Administration' and 'For Cisco Unified Communications Solutions'. The user is logged in as 'admin'. The main menu includes 'System', 'Call Routing', 'Media Resources', 'Advanced Features', 'Device', 'Application', 'User Management', 'Bulk Administration', and 'Help'. The current page is 'Phone Configuration', with a 'Related Links' section containing 'Back To Find/List'. A toolbar at the top of the page contains icons for 'Save', 'Delete', 'Copy', 'Reset', 'Apply Config' (highlighted with a red box), and 'Add New'. Below the toolbar, the 'Status' section shows 'Status: Ready'. The 'Association' section lists eight lines, each with a 'Line [X] - Add a new DN' link. The 'Phone Type' section shows 'Product Type: Third-party SIP Device (Advanced)' and 'Device Protocol: SIP'. The 'Real-time Device Status' section shows 'Registration: Registered with Cisco Unified Communications Manager CM-10', 'IPv4 Address: 10.15.77.210', 'Active Load ID: None', and 'Download Status: None'. The 'Device Information' section includes a 'Device is Active' checkbox (checked) and a 'Device is not trusted' warning. Below this, various fields are listed with their values: MAC Address\* (000000005004), Description (MP1288-Line4), Device Pool\* (Default), Common Device Configuration (< None >), Phone Button Template\* (Third-party SIP Device (Advanced)), Common Phone Profile\* (Standard Common Phone Profile), Calling Search Space (< None >), AAR Calling Search Space (< None >), Media Resource Group List (< None >), Location\* (Hub\_None), and AAR Group (< None >).

- You can configure up to eight phone line connections between the CUCM and the MP-1288 device (for implementation as CUCM 3<sup>rd</sup> party SIP Device).
- 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

The screenshot shows the Cisco Unified CM Administration interface. The top navigation bar includes 'Cisco Unified CM Administration' and 'For Cisco Unified Communications Solutions'. The main navigation menu includes 'System', 'Call Routing', 'Media Resources', 'Advanced Features', 'Device', 'Application', 'User Management', 'Bulk Administration', and 'Help'. The current page is 'Phone Configuration' for a device, with a 'Related Links' section containing 'Back To Find/List'. The 'Status' section shows 'Status: Ready'. The 'Association' section is highlighted with a red box, showing a list of lines (1-8) with 'Add a new DN' links. The 'Phone Type' section shows 'Product Type: Third-party SIP Device (Advanced)' and 'Device Protocol: SIP'. The 'Real-time Device Status' section shows 'Registration: Rejected', 'IPv4 Address: None', 'Active Load ID: None', and 'Download Status: None'. The 'Device Information' section shows 'Device is Active' (checked), 'Device is not trusted' (warning icon), 'MAC Address\*: 000000005004', and 'Description: MP1288-Line4'.

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

The screenshot shows the Cisco Unified CM Administration interface for 'Directory Number Configuration'. The top navigation bar is the same as in Figure 13. The main navigation menu includes 'System', 'Call Routing', 'Media Resources', 'Advanced Features', 'Device', 'Application', 'User Management', 'Bulk Administration', and 'Help'. The current page is 'Directory Number Configuration' for a device, with a 'Related Links' section containing 'Configure Device (SEP000000005004)'. The 'Status' section shows 'Directory Number Configuration has refreshed due to a directory number change. Please click Save button to save the configuration.' The 'Directory Number Information' section is highlighted with a red box, showing the 'Directory Number\*' field with the value '5004' entered. Other fields include 'Route Partition' (set to '< None >'), 'Description', 'Alerting Name', 'ASCII Alerting Name', 'External Call Control Profile' (set to '< None >'), and 'Active' (checked).

3. Click **Save**.

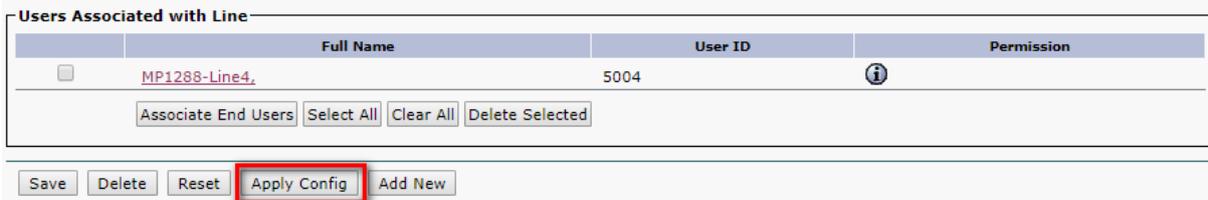
4. Scroll down to the 'Users Associated with Line' pane.

Figure 15: Associate Line with End User



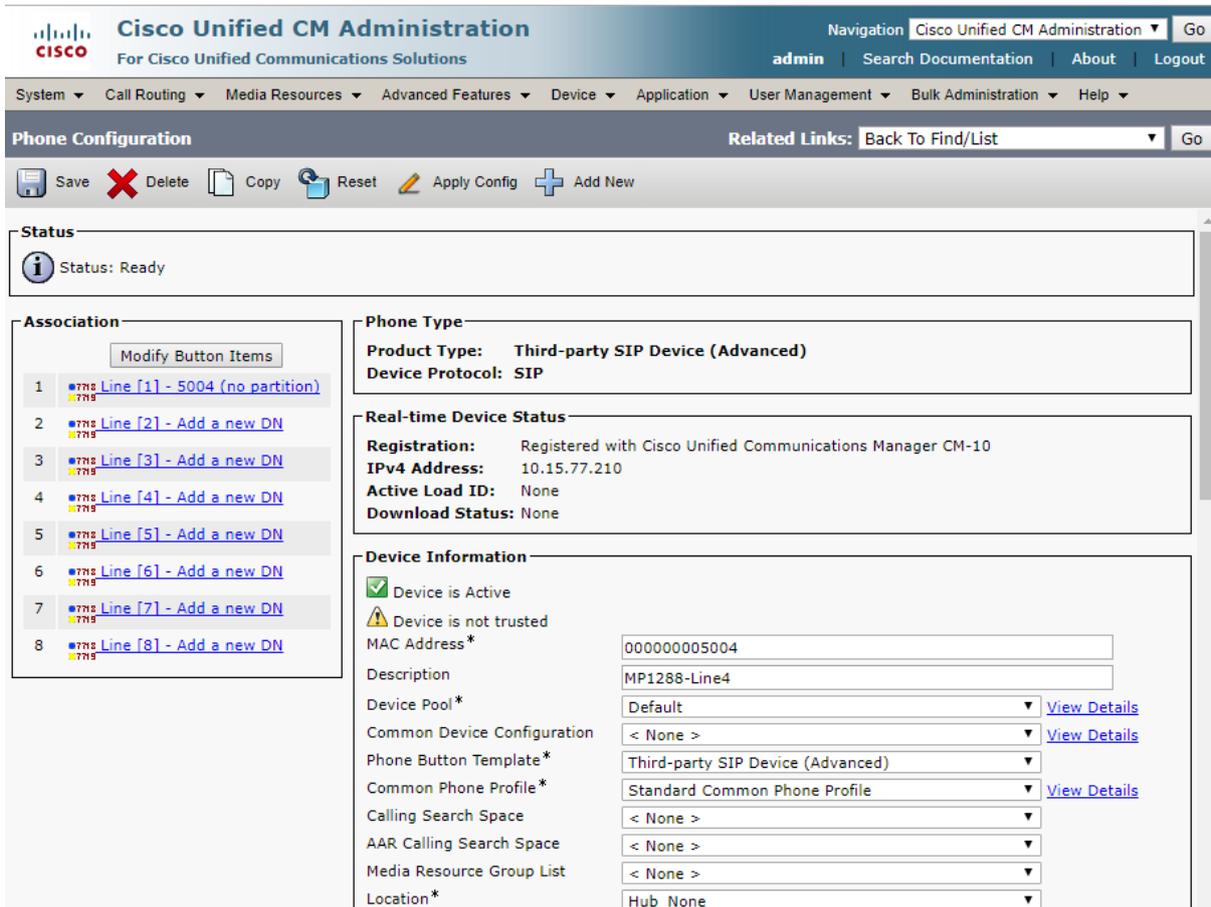
5. Click **Associate End Users** to associate this line with the end user, created in Section 3.2 on page 4.
6. Click **Save**.
7. Click **Apply Config**.

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 3<sup>rd</sup> party SIP Device or up to two telephony numbers for implementation of MP-1288 as CUCM 3<sup>rd</sup> 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	<b>0</b>
Name	<b>SIPInterface_5060</b>
Network Interface	<b>Voice</b>
Application Type	<b>GW</b>
UDP Port	<b>5060</b>
TCP Port	<b>0</b>
TLS Port	<b>0</b>

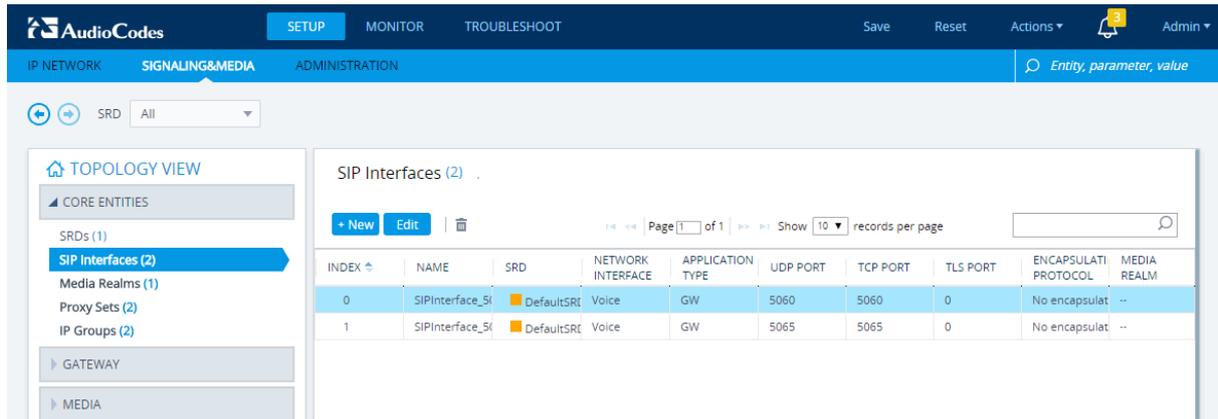
3. Configure a SIP Interface for Port 5065:

Parameter	Value
Index	<b>1</b>
Name	<b>SIPInterface_5065</b>
Network Interface	<b>WAN_IF</b>
Application Type	<b>Voice</b>
Application Type	<b>GW</b>
UDP Port	<b>5065</b>
TCP Port	<b>0</b>
TLS Port	<b>0</b>

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**



The screenshot shows the AudioCodes management console interface. The top navigation bar includes 'AudioCodes', 'SETUP', 'MONITOR', and 'TROUBLESHOOT'. Below this, there are tabs for 'IP NETWORK', 'SIGNALING&MEDIA', and 'ADMINISTRATION'. The main content area displays the 'SIP Interfaces (2)' table. The table has columns for INDEX, NAME, SRD, NETWORK INTERFACE, APPLICATION TYPE, UDP PORT, TCP PORT, TLS PORT, ENCAPSULATION PROTOCOL, and MEDIA REALM. Two rows of data are visible, representing the configured SIP interfaces.

INDEX	NAME	SRD	NETWORK INTERFACE	APPLICATION TYPE	UDP PORT	TCP PORT	TLS PORT	ENCAPSULATION PROTOCOL	MEDIA REALM
0	SIPInterface_Si	DefaultSR1	Voice	GW	5060	5060	0	No encapsulat	--
1	SIPInterface_Si	DefaultSR1	Voice	GW	5065	5065	0	No encapsulat	--

## 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

SRD

**GENERAL**

Index

Name

Gateway IPv4 SIP Interface  [View](#)

TLS Context Name  [View](#)

**REDUNDANCY**

Redundancy Mode

Proxy Hot Swap

Proxy Load Balancing Method

Min. Active Servers for Load Balancing

**KEEP ALIVE**

Proxy Keep-Alive

Proxy Keep-Alive Time [sec]

Keep-Alive Failure Responses

**ADVANCED**

Classification Input

DNS Resolve Method

[Cancel](#) [APPLY](#)

- 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	<b>0</b>
Proxy Address	<b>10.15.25.11</b> (Cisco CUCM IP address)
Transport Type	<b>UDP</b>

- 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**

INDEX	NAME	SRD	GATEWAY IPV4 SIP INTERFACE	PROXY KEEP-ALIVE TIME [SEC]	REDUNDANCY MODE	PROXY HOT SWAP
0	ProxySet_5060	DefaultSRD (#0)	SIPInterface_5060	60		Disable
1	ProxySet_5065	DefaultSRD (#0)	SIPInterface_5065	60		Disable

## 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**

INDEX	NAME	SRD	SBC OPERATION MODE	PROXY SET	IP PROFILE	MEDIA REALM	SIP GROUP NAME	INBOUND MESSAGE MANIPULATION SET	OUTBOUND MESSAGE MANIPULATION SET
0	IPG_5060	DefaultSRD	Not Configured	ProxySet_5060	--	--		-1	1
1	IPG_5065	DefaultSRD	Not Configured	ProxySet_5065	--	--		-1	1

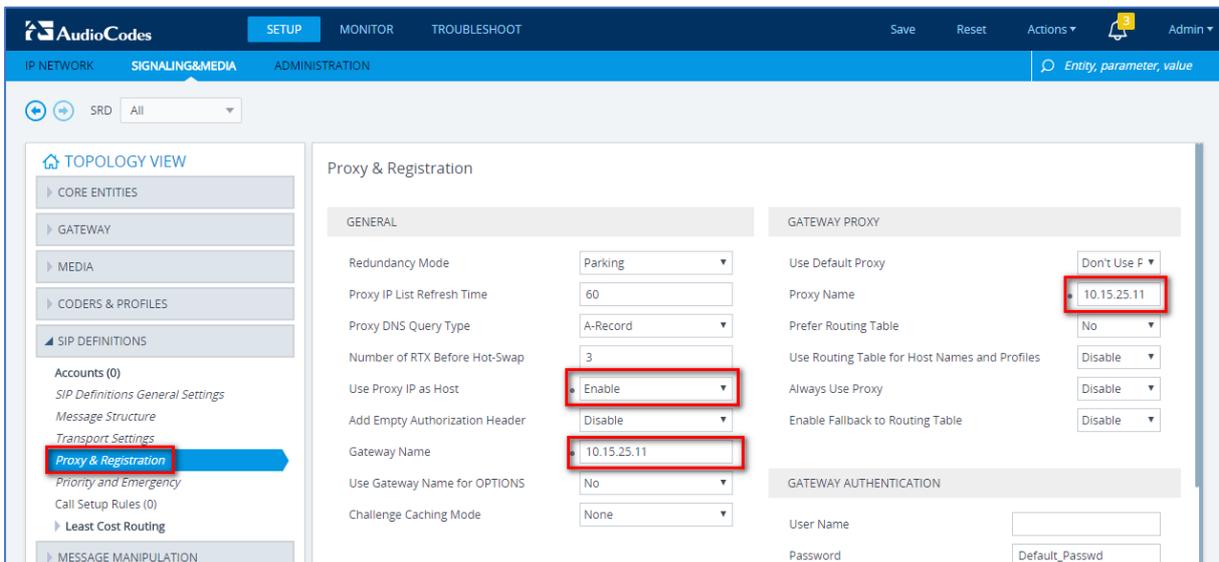
## 4.4 Configure Proxy Server and Registration

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

To configure the Proxy and Registration parameters:

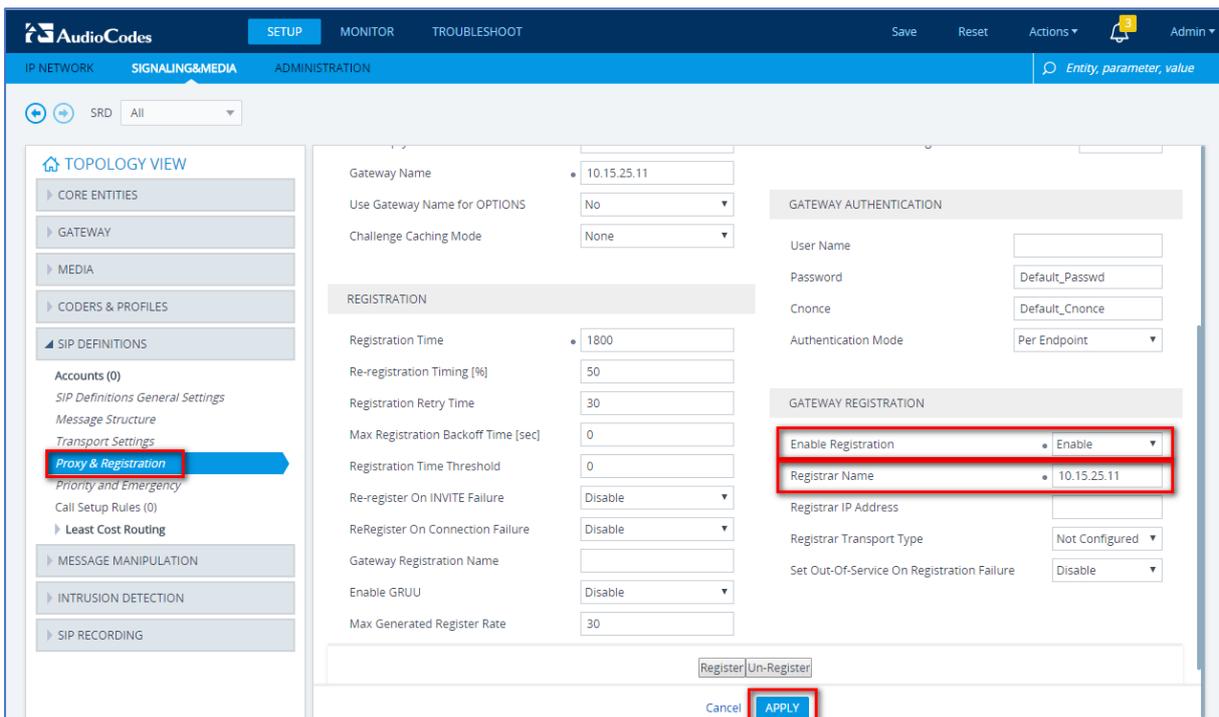
1. Open the Proxy & Registration page (Setup menu > Signaling & Media tab > SIP Definitions folder > Proxy & Registration).

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

The screenshot shows the 'Trunk Group Table' configuration page. The table is as follows:

Group Index	FXS Blade	Channels	Phone Number	Trunk Group ID	Tel Profile Name
1	FXS Blade 1	1	5001	1	None
2	FXS Blade 1	2	5002	1	None
3	FXS Blade 1	3	5003	1	None
4	FXS Blade 1	4	5004	4	None
5					None
6					None
7					None

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**

The screenshot shows the AudioCodes management interface. The top navigation bar includes 'IP NETWORK', 'SIGNALING&MEDIA', and 'ADMINISTRATION'. The left sidebar shows a 'TOPOLOGY VIEW' with 'GATEWAY' expanded to 'Trunks & Groups' and 'Trunk Group Settings (2)' selected. The main content area displays a table titled 'Trunk Group Settings (2)' with the following data:

INDEX	NAME	TRUNK GROUP ID	CHANNEL SELECT MODE	REGISTRATION MODE	SERVING IP GROUP	ADMIN STATE	STATUS
0	Users 1-8	1	By Dest Phone Nu	Per Endpoint	IPG_5060	Unlocked	
1	Users 9-16	4	By Dest Phone Nu	Per Endpoint	IPG_5065	Unlocked	

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

The screenshot shows the AudioCodes management console interface. The top navigation bar includes 'IP NETWORK', 'SIGNALING&MEDIA', and 'ADMINISTRATION'. The 'SIGNALING&MEDIA' tab is active. On the left, a 'TOPOLOGY VIEW' sidebar shows a tree structure with 'CORE ENTITIES', 'GATEWAY', 'Trunks & Groups', and 'Routing'. Under 'Routing', 'Tel->IP Routing (1)' is selected. The main area displays the 'Tel-to-IP Routing (1)' table with the following data:

INDEX	NAME	SOURCE TRUNK GROUP ID	SOURCE PHONE PREFIX	DESTINATION PHONE PREFIX	DESTINATION IP GROUP	SIP INTERFACE	DESTINATION IP ADDRESS	FORKING GROUP	CONNECTIVITY STATUS
0		-1	*	*	--	--	10.15.25.11	-1	Not Available

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

The screenshot shows the AudioCodes management console interface. The top navigation bar includes 'SETUP', 'MONITOR', and 'TROUBLESHOOT'. The main menu is divided into 'IP NETWORK', 'SIGNALING&MEDIA', and 'ADMINISTRATION'. The left sidebar shows a tree view with 'CORE ENTITIES', 'GATEWAY', and 'Routing' expanded to 'IP->Tel Routing (2)'. The main content area displays the 'IP-to-Tel Routing (2)' table with the following data:

INDEX	NAME	SOURCE IP GROUP	SOURCE SIP INTERFACE	SOURCE IP ADDRESS	SOURCE PHONE PREFIX	DESTINATION PHONE PREFIX	TRUNK GROUP ID
0	From CUCM to 50	--	SIPInterface_5060			*	1
1	From CUCM to 50	--	SIPInterface_5065			*	4

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

The screenshot shows the AudioCodes management console interface. The top navigation bar includes 'SETUP', 'MONITOR', and 'TROUBLESHOOT'. The main menu is divided into 'IP NETWORK', 'SIGNALING&MEDIA', and 'ADMINISTRATION'. The left sidebar shows a tree view with 'Trunks & Groups', 'Routing', 'Manipulation', 'DTMF & Supplementary', and 'Analog Gateway' expanded to 'Authentication (144)'. The main content area displays the 'Authentication (144)' table with the following data:

INDEX	MODULE	PORT	PORT TYPE	USER NAME	PASSWORD
0	1	1	FXS	5001	*
1	1	2	FXS	5001	*
2	1	3	FXS	5001	*
3	1	4	FXS	5004	*
4	1	5	FXS	5004	*
5	1	6	FXS		

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**.

**International Headquarters**

1 Hayarden Street,  
Airport City  
Lod 7019900, Israel  
Tel: +972-3-976-4000  
Fax: +972-3-976-4040

**AudioCodes Inc.**

80 Kingsbridge Rd  
Piscataway, NJ 08854, USA  
Tel: +1-732-469-0880  
Fax: +1-732-469-2298

Contact us: <https://www.audiocodes.com/corporate/offices-worldwide>

Website: <https://www.audiocodes.com>

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