

Audiocodes MP-114

Integration with Cisco CUCMv12.5

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Cisco Unified Communications Manager

Cisco Unified Communications Manager (Unified CM) provides reliable, secure, scalable, and manageable call control and session management.

Consolidate your communications infrastructure and enable your people and teams to communicate simply with the Cisco Unified Communications Manager. The solution features IP telephony, high-definition video, unified messaging, Instant Message and Presence.

In addition to the Cisco IP Phones that run SIP, Unified Communications Manager supports a variety of third-party SIP endpoints. You can configure the following third-party SIP endpoints in Cisco Unified Communications Manager Administration:

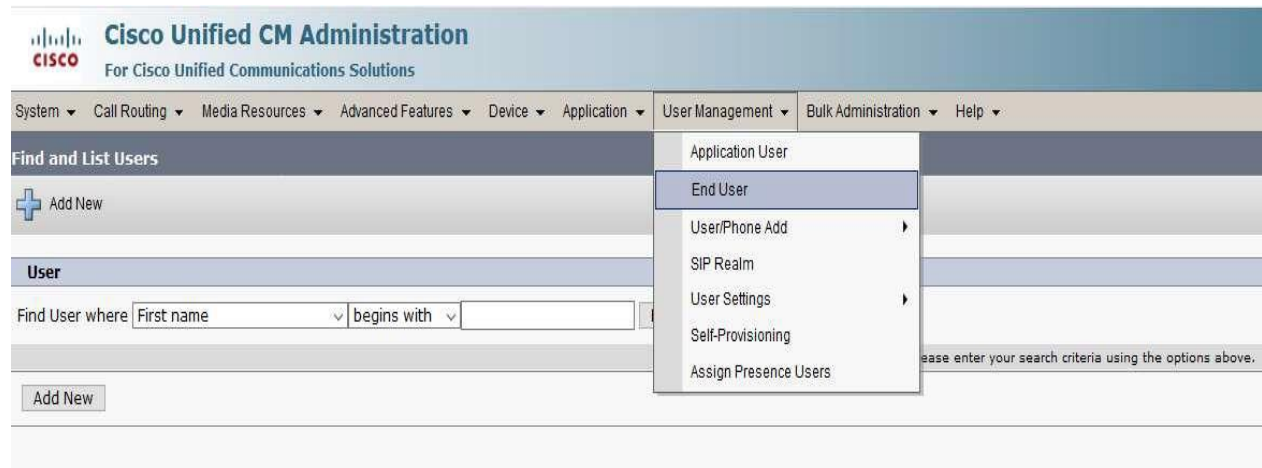
Third-Party SIP Device (Advanced)

This eight-line SIP device is an RFC3261-compliant phone that is running SIP from third-party companies.

Configure an End User

To enable digest authentication, configure an end user that is a digest user. Cisco Unified Communications Manager uses the digest credentials that you specify in the End User Configuration window to validate the SIP user agent response during a challenge to the SIP trunk.

Select User Management -> End User-> Add New



Create an Enabled Local User:

Username: [Site_mp114](#)

Last Name: [site](#)

Digest Credentials: *****

each Directory Number will require a username and digest credentials and have the DN associated with the original user

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

End User Configuration

Save

User Information

User Status: Enabled Local User

User ID*: Site_mp114

Password:

Confirm Password:

Self-Service User ID:

PIN:

Confirm PIN:

Last name*: site

Middle name:

First name:

Display name:

Title:

Directory URI:

Telephone Number:

Home Number:

Mobile Number:

Pager Number:

Mail ID:

Manager User ID:

Department:

User Locale: < None >

Associated PC/Site Code:

Digest Credentials: ●●●●

Confirm Digest Credentials: ●●●●

User Profile: Use System Default("Standard (Factory Default) Us" [View Details](#))

User Rank*: 1-Default User Rank

Assign this user to the phone that will be created in the next step.

Add a Third-Party SIP Endpoint

Configure a third-party SIP endpoint (Advanced).

Select Device-> [Add New](#)

Select Phone Type-> [Third-party-SIP Device \(Advanced\)](#)

Cisco Unified CM Administration Navigation: Cisco Unified CM Administration Go

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Add a New Phone Related Links: [Back To Find/List](#) Go

Next

Status

Status: Ready

Add New Phone Information

Start by selecting the type of phone you wish to add, or [click here to add a new phone using a Universal Device Template.](#)

Phone Type* Third-party SIP Device (Advanced)

Next

* indicates required item.



** Create a phone template using the Bulk Administration Tool to enable template-based phone creation.

The device will be inserted.

MAC Address: [MAC Address of the MP114](#)

Description: [Site_mp114](#)

User: [User you created \(site_mp114\)](#)

Status	
 Status: Ready	
Phone Type	
Product Type: Third-party SIP Device (Advanced)	
Device Protocol: SIP	
Device Information	
 Device is not trusted	
MAC Address*	123412341234
Description	site_mp114
Device Pool*	dp_corporate View Details
Common Device Configuration	< None > View Details
Phone Button Template*	Third-party SIP Device (Advanced)
Common Phone Profile*	Standard Common Phone Profile View Details
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*	MP114 Find

Associate Device to End User

Associate the third-party endpoint with an end user.

Also ensure that the Digest User is in the phone configuration.

Protocol Specific Information	
BLF Presence Group*	Standard Presence group
MTP Preferred Originating Codec*	711ulaw
Device Security Profile*	Audiocodes SIP Device Advanced - Standard SIP No View Details
Rerouting Calling Search Space	< None >
SUBSCRIBE Calling Search Space	< None >
SIP Profile*	< None > View Details
Digest User	MP114 Find
<input type="checkbox"/> Media Termination Point Required	
<input type="checkbox"/> Unattended Port	
<input type="checkbox"/> Require DTMF Reception	
<input type="checkbox"/> Allow Presentation Sharing using BFCP	
<input type="checkbox"/> Allow iX Applicable Media	

Audiocodes MP114 Gateway

The MP114-FXO Audiocodes Gateway is a stand-alone analog VoIP Gateway that provides superior voice technology for connecting legacy telephones, fax machines and PBX systems with IP-based telephony networks, as well as for integration with new IP-based PBX architecture.

Administration

Log into the AudioCodes gateway.

Use the Administrative username and password provided for the gateway.

In the example below the login is default.

Login Screen

The base web page login from an AudioCodes MP114

AudioCodes MP-114 FXS

Web Login

Username
Admin

Password
Admin

Remember Me **Login**

Remember Me **Login**

Splash Screen

The screen will change to the AudioCodes default page.

Note: The registration status may not be green until after configuration.

If the gateway has any alarms the

AudioCodes MP-114 FXS

MP-114 FXS Home Page

Configuration Maintenance Status & Diagnostics

Scenarios Search

Basic Full

Full System

Full VoIP

1 2 3 4

Uplink Fail Ready Power

General Information	
IP Address	192.168.06.16
Subnet Mask	255.255.255.0
Default Gateway Address	192.168.06.1
Firmware Version	6.65A.309.001
Protocol Type	SP
Gateway Operational State	UNLOCKED
Analog Ports Number	4

Color-Code Key	
○ Inactive	
● Handset Offhook	
● RTP Active	

Control Network

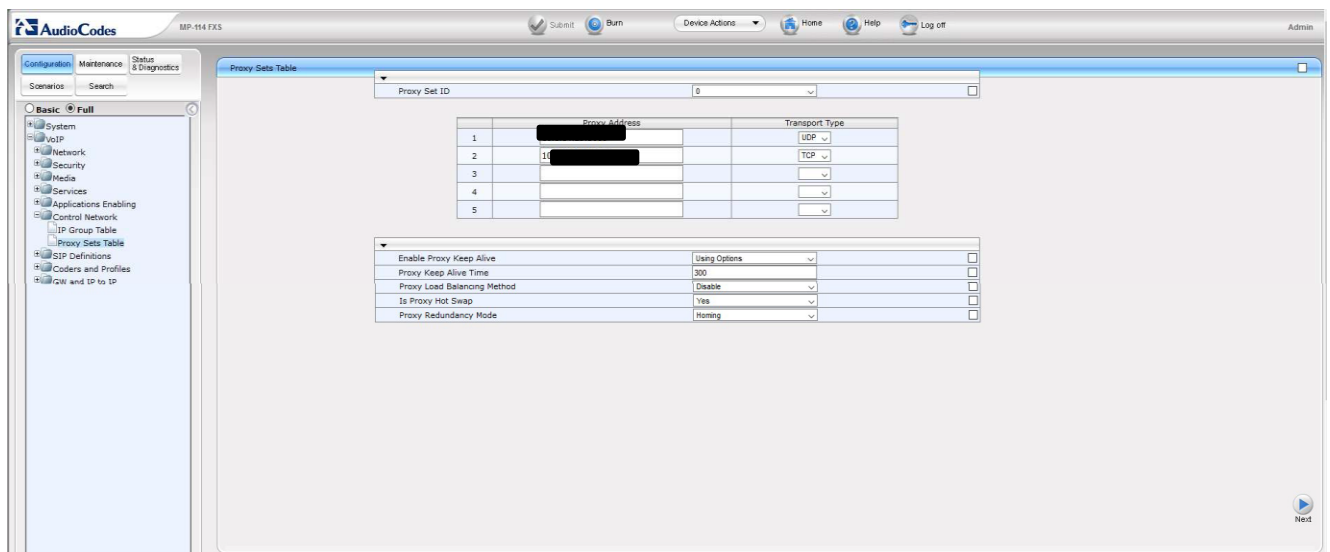
Registering the Cisco CUCM to the Audiocodes requires entering the CUCM IP address to the Audio Codes Gateway.

Proxy Sets Table

The Proxy Sets Table page allows you to define Proxy Sets. A Proxy Set is a group of Proxy servers defined by IP address or fully qualified domain name (FQDN). You can define up to 10 Proxy Sets, each with up to five Proxy server addresses. For each Proxy server address you can define the transport type (i.e., UDP, TCP, or TLS). The total number of IP addresses that can be resolved from a DNS query is 15. In addition, Proxy load balancing and redundancy mechanisms can be applied per Proxy Set if it contains more than one Proxy address.

Ex: IP Address: port

CUCM `xxx.xxx.xxx.xxx:5060`



SIP Definitions

AudioCodes MP114 configuration of SIP parameters

Proxy and Registration

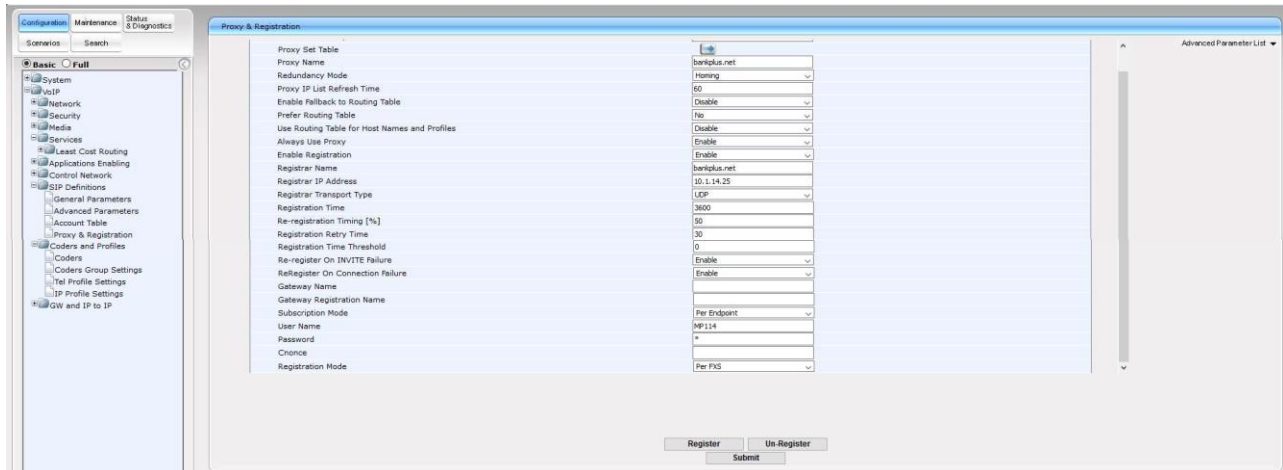
The Proxy & Registration page allows you to configure the Proxy server and registration parameters.

Enter the Proxy Name: `customer.net`

Registrar IP Address: `CUCM IP address`

Subscription Mode: `Per Endpoint`

Registration Mode: `Per Mode`



GW and IP to IP

The Gateway application refers to IP-to-Tel call routing and vice versa

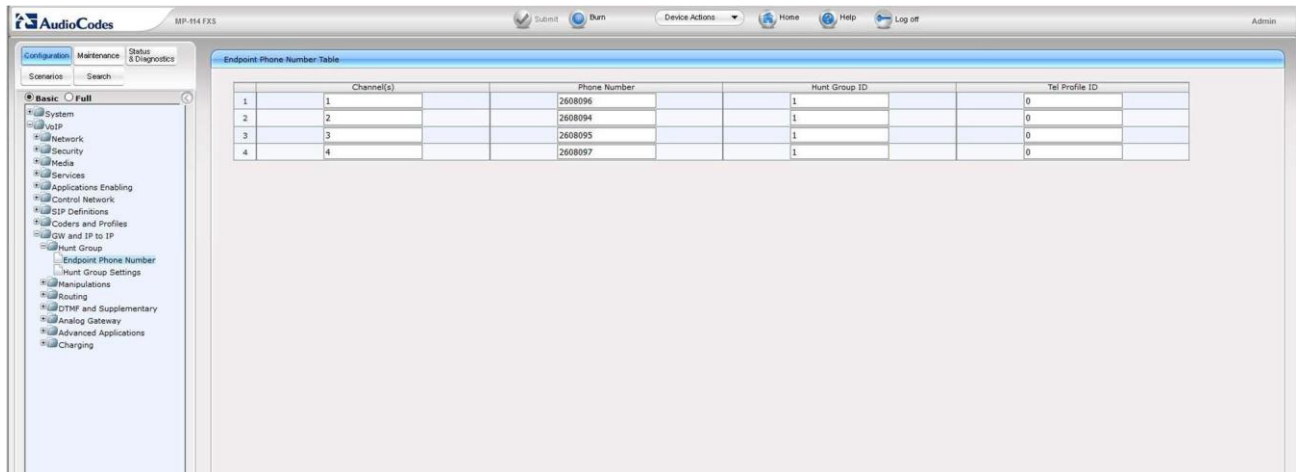
Endpoint Phone Number

The Endpoint Phone Number Table page allows you to activate the device's ports (channels or endpoints), by defining telephone numbers for the endpoints and assigning them to Hunt Groups and Tel Profiles.

DN: XXXXXXXXXXXX

Hunt Group: 1

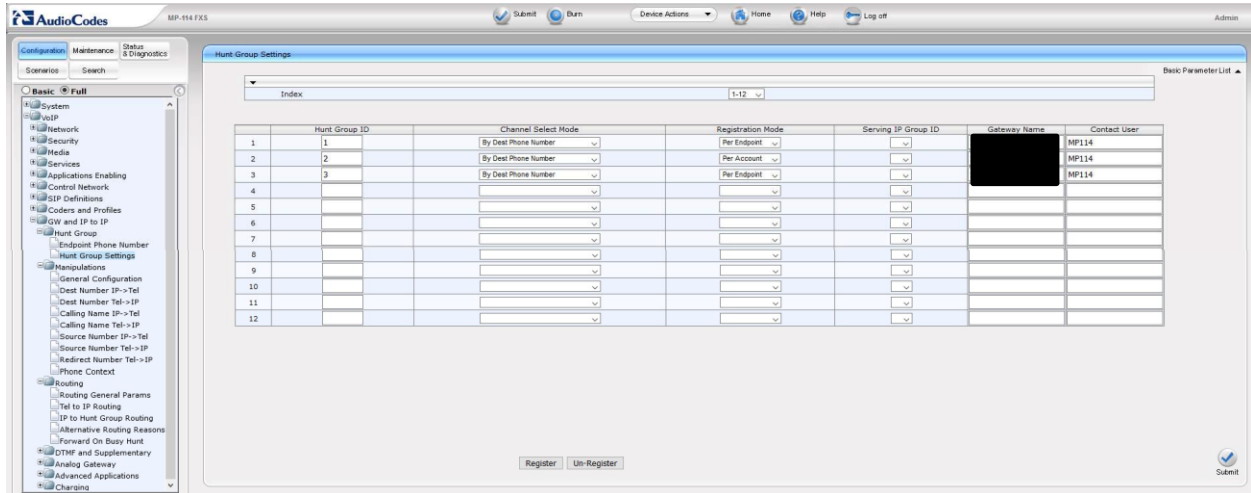
Tel Profile ID: 0



Hunt Group

The Hunt Group Settings allows you to configure the following per Hunt Group: Channel select method by which IP-to-Tel calls are assigned to the Hunt Group's channels. Registration method for registering Hunt Groups to selected Serving IP Group IDs

Hunt Group ID: 1
 Channel Select Mode: **By Dest Phone Number**
 Registration Mode: **Per Endpoint**
 Gateway Name: **CUCM IP Address**
 Contact User: **End User created in CUCM**



Routing

This section describes the configuration of call routing rules.

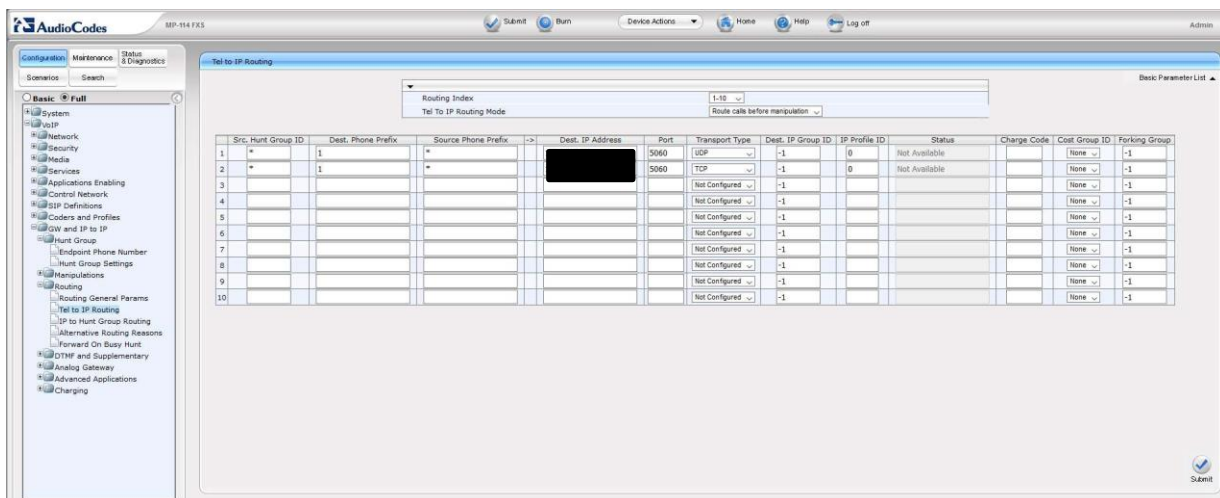
Tel IP Routing

The Tel to IP Routing page allows you to configure up to 50 Tel-to-IP call routing rules. The device uses these rules to route calls from the Tel to a user-defined IP destination.

There will be 2 entries

The Destination IP Address: **Gateway IP**

Protocol: **UDP and TCP**



Analog Gateway

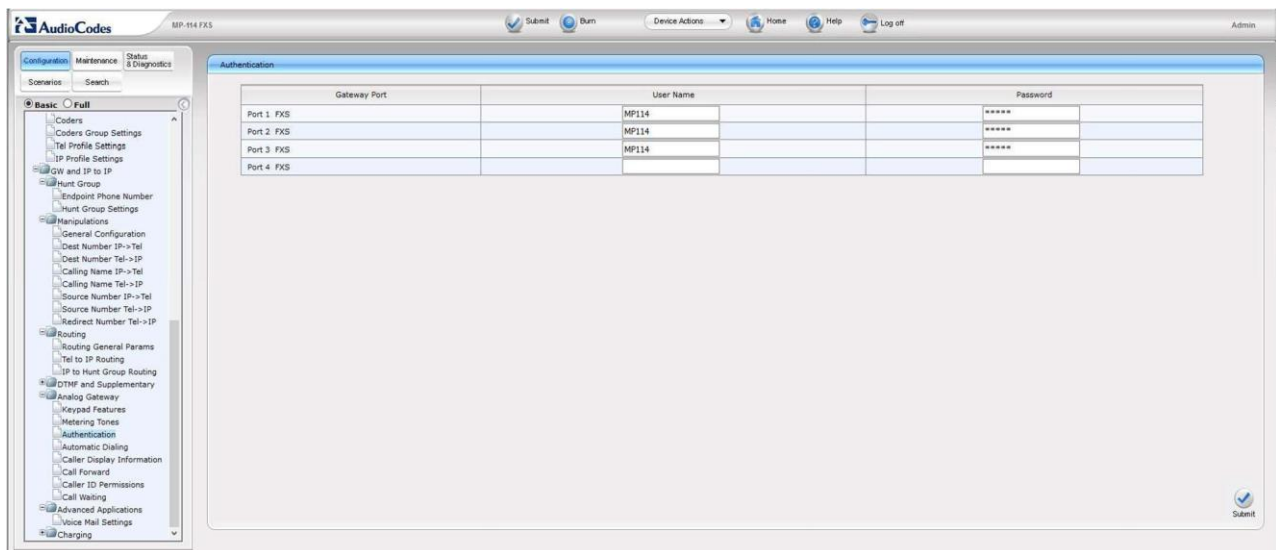
Authentication

The Authentication page defines a user name and password for authenticating each device port. Authentication is typically used for FXS interfaces, but can also be used for FXO interfaces.

Gateway port

Username: End user created during CUCM configuration

Password: Password created during CUCM configuration



Troubleshooting

There are some basic troubleshooting areas to check. The CUCM will show unregistered, partial register, or registered. The registrations are generally related to the end user and digest credentials. You can re-enter digest credentials. Check the Analog Gateway ->Authentication credentials.

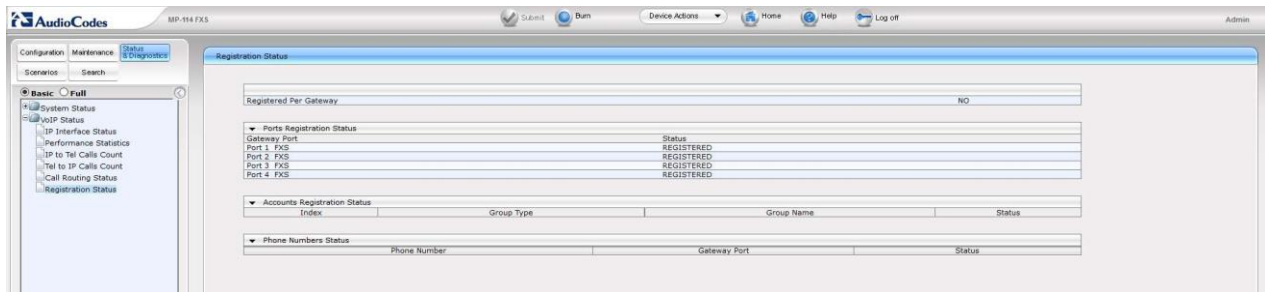
VoIP Status

To check the registration status.

Click on Status & Diagnostics

VoIP Status-> Registration Status.

Look for all ports **REGISTERED**



In the event that any ports are unregistered, click on System Status->Message Log



The Message Log will provide registration status and issues.

