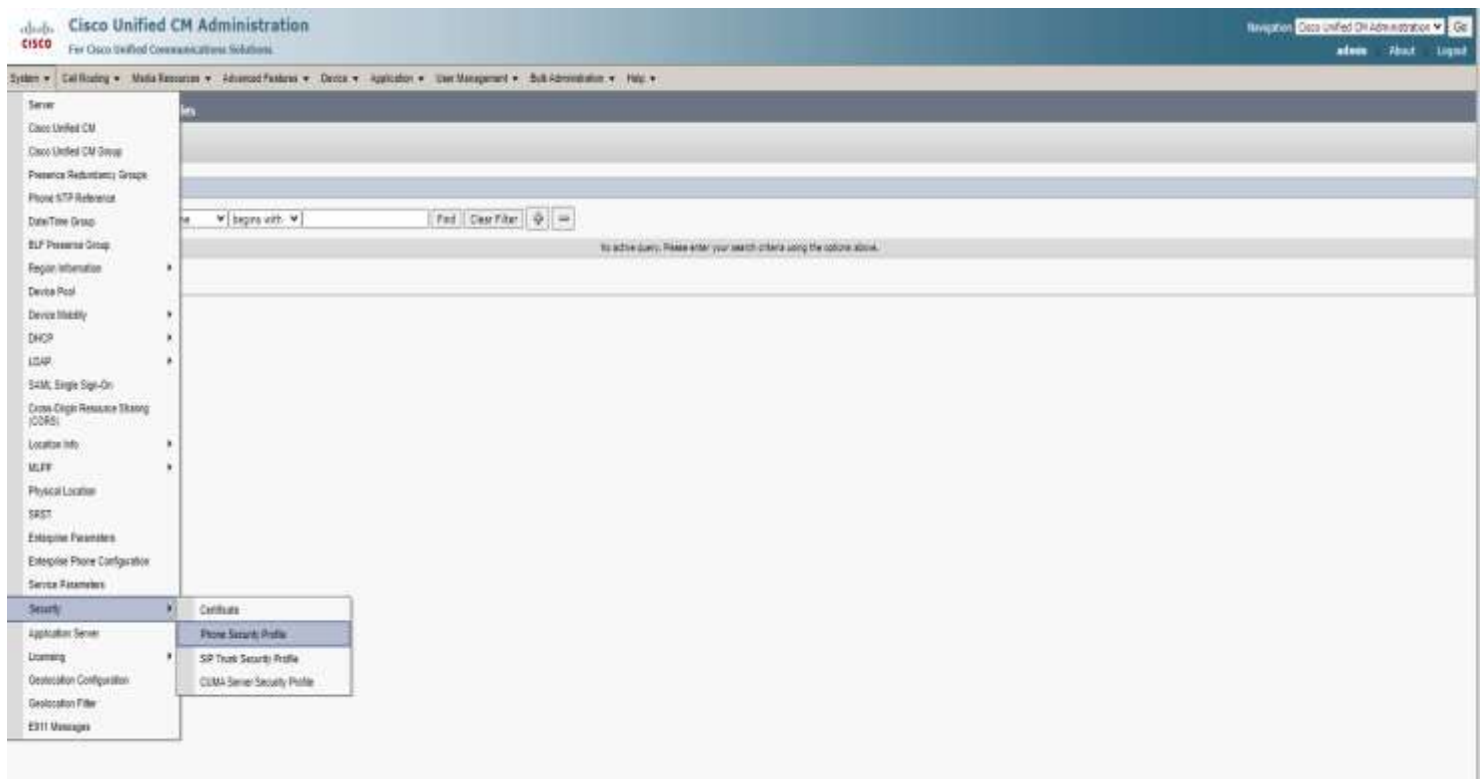


Valcom Session Initiation Protocol (SIP) VIP devices are compatible with Cisco Unified Communications Manager (formerly Cisco Unified CallManager) (SIP enabled versions). The Valcom device is added to the Communications Manager as a Third-party SIP Device (Basic or Advanced). Third-party SIP Device (Basic) supports one line, Third-party SIP Device (Advanced) supports up to eight lines.

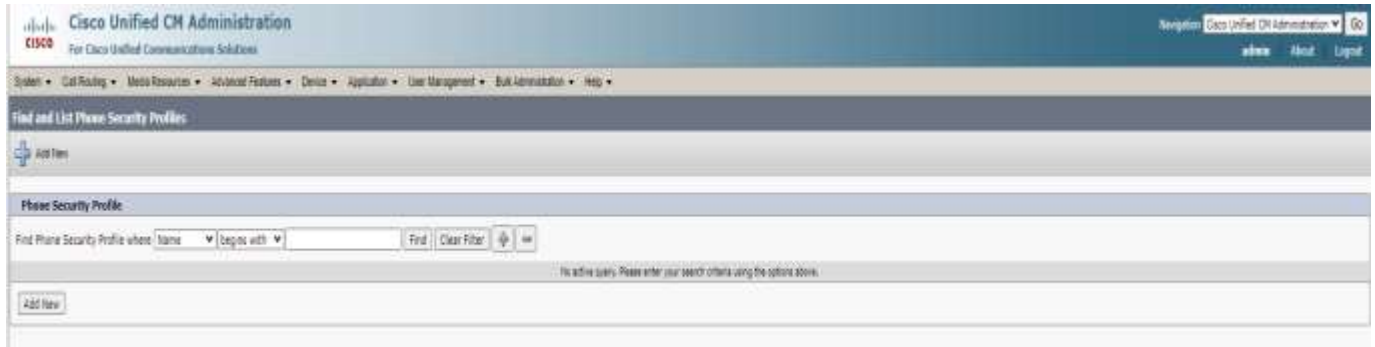
Default, non-secure Phone Security Profiles do not require authentication for a phone to register. To enable digest authentication, a new Phone Security Profile must be configured. If an appropriate profile has already been defined, it may be used for the Valcom device. Skip to Step 5 if an existing profile will be used, or if authentication is not required and a built-in (non-secure) profile will be used.

Navigate your web browser to the IP address of your Cisco Unified Communications Manager server and login.

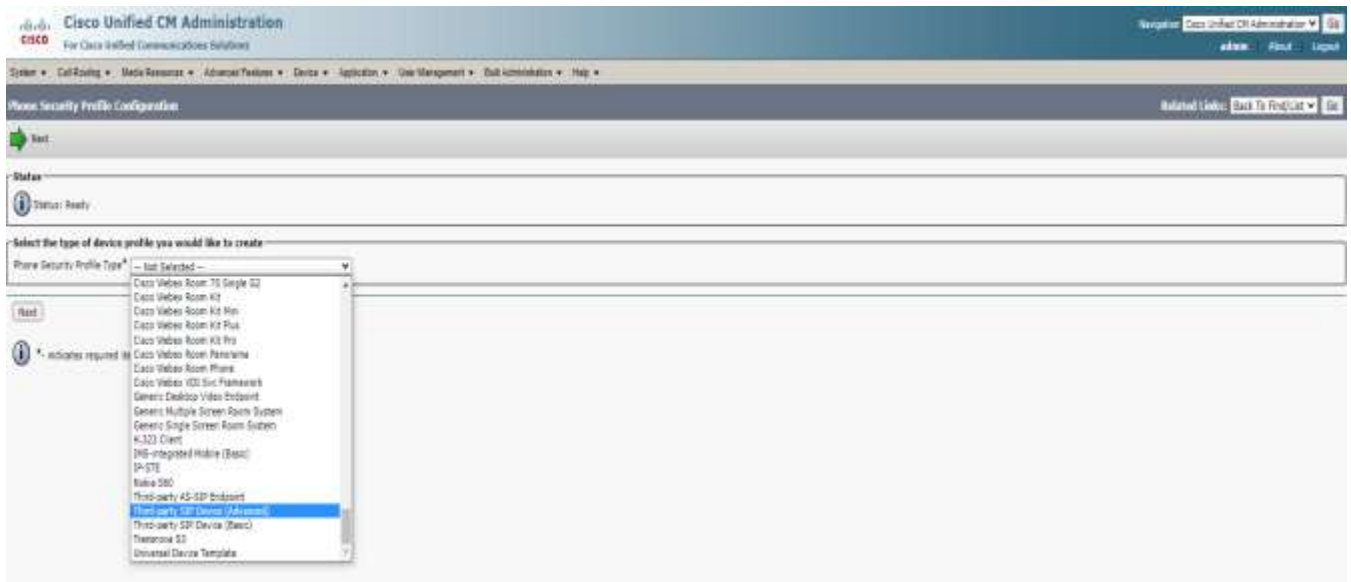
1. Go to the “System” menu, and then click “Security Profile”, then click “Phone Security Profile”.



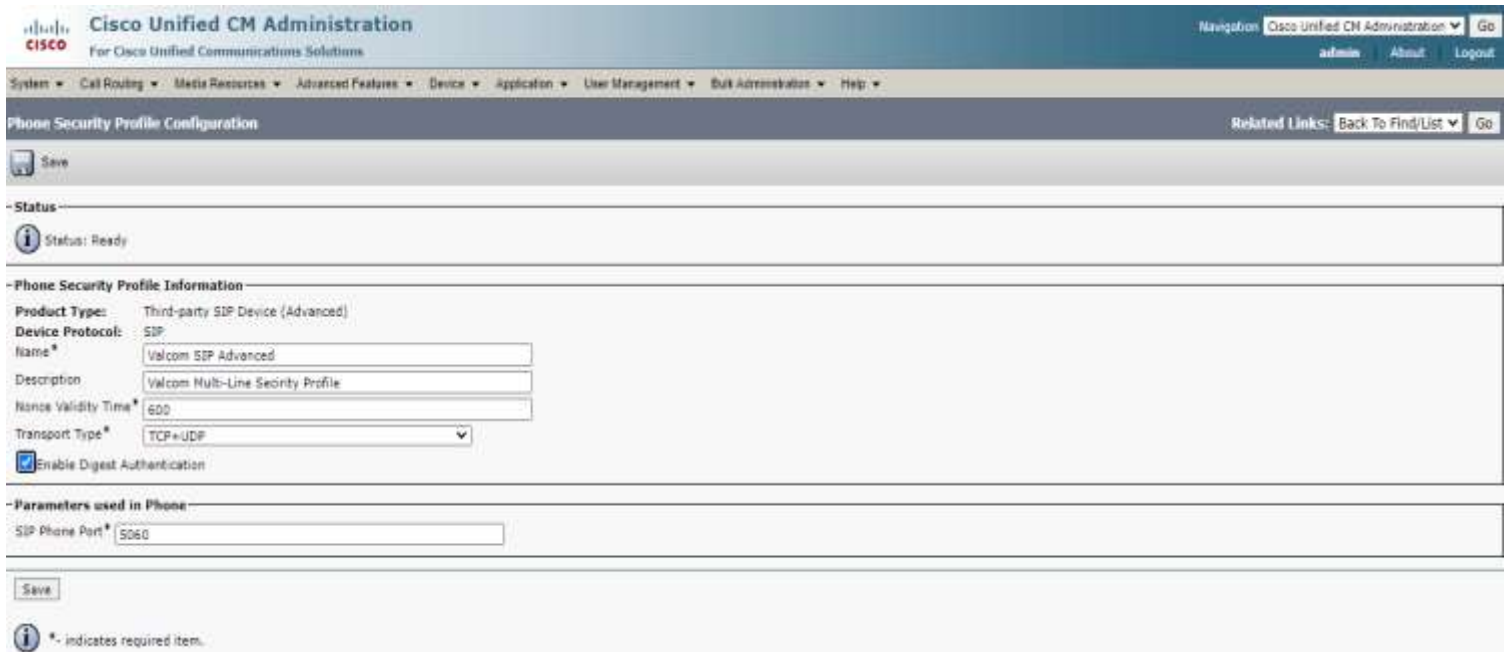
2. Click on “Add New”



3. On the Phone Security Profile Configuration screen, select the appropriate Profile Type from the dropdown list. For Valcom devices, the type will be either Third-party SIP Device (Advanced) or Third-party SIP Device (Basic). The profile being created will only be available for the phone type that is selected. Use Basic for devices that only have a single SIP identity (such as a SIP speaker). Select Advanced for devices that have multiple SIP identities (such as the VIP-201 Paging Server). Click “Next” after selecting the Type.



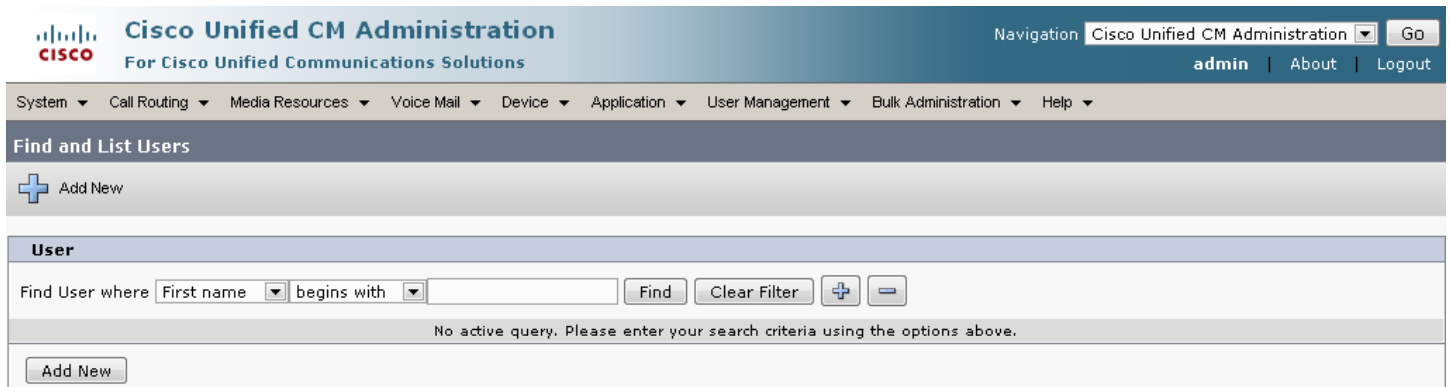
4. Enter the Phone Security Profile Information.
 - A) Enter “Name*” (ex. Valcom SIP Advanced)
 - B) Enter “Nonce Validity Time*” in seconds (default 600)
 - C) For “Transport Type*” select “UDP” or TCP+UDP from the dropdown list
 - D) Check the box for “Enable Digest Authentication”
 - E) The “SIP Phone Port*” should be left at the default of 5060, unless it is also changed in the Valcom device.
 - F) Click the “Save” button when all fields have been entered.



The screenshot shows the 'Phone Security Profile Configuration' page in Cisco Unified CM Administration. The page includes a navigation menu at the top with options like System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The main content area is titled 'Phone Security Profile Configuration' and contains several sections: 'Status' (Ready), 'Phone Security Profile Information' (Product Type: Third-party SIP Device (Advanced), Device Protocol: SIP, Name: Valcom SIP Advanced, Description: Valcom Multi-Line Security Profile, Name Validity Time: 600, Transport Type: TCP+UDP, Enable Digest Authentication checked), and 'Parameters used in Phone' (SIP Phone Part: 5060). A 'Save' button is located at the bottom left, and an information icon indicates that asterisks denote required items.

The following steps outline the typical device configuration process:

1. Under the “User Management” menu, select “End User”
2. Click on “Add New”



The screenshot shows the 'Find and List Users' page in Cisco Unified CM Administration. The page includes a navigation menu at the top with options like System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. The main content area is titled 'Find and List Users' and contains an 'Add New' button. Below this is a 'User' section with a search form. The search form has a 'Find User where' label and two dropdown menus: 'First name' and 'begins with'. There are also 'Find', 'Clear Filter', and '+' buttons. Below the search form, a message states: 'No active query. Please enter your search criteria using the options above.' An 'Add New' button is located at the bottom left.

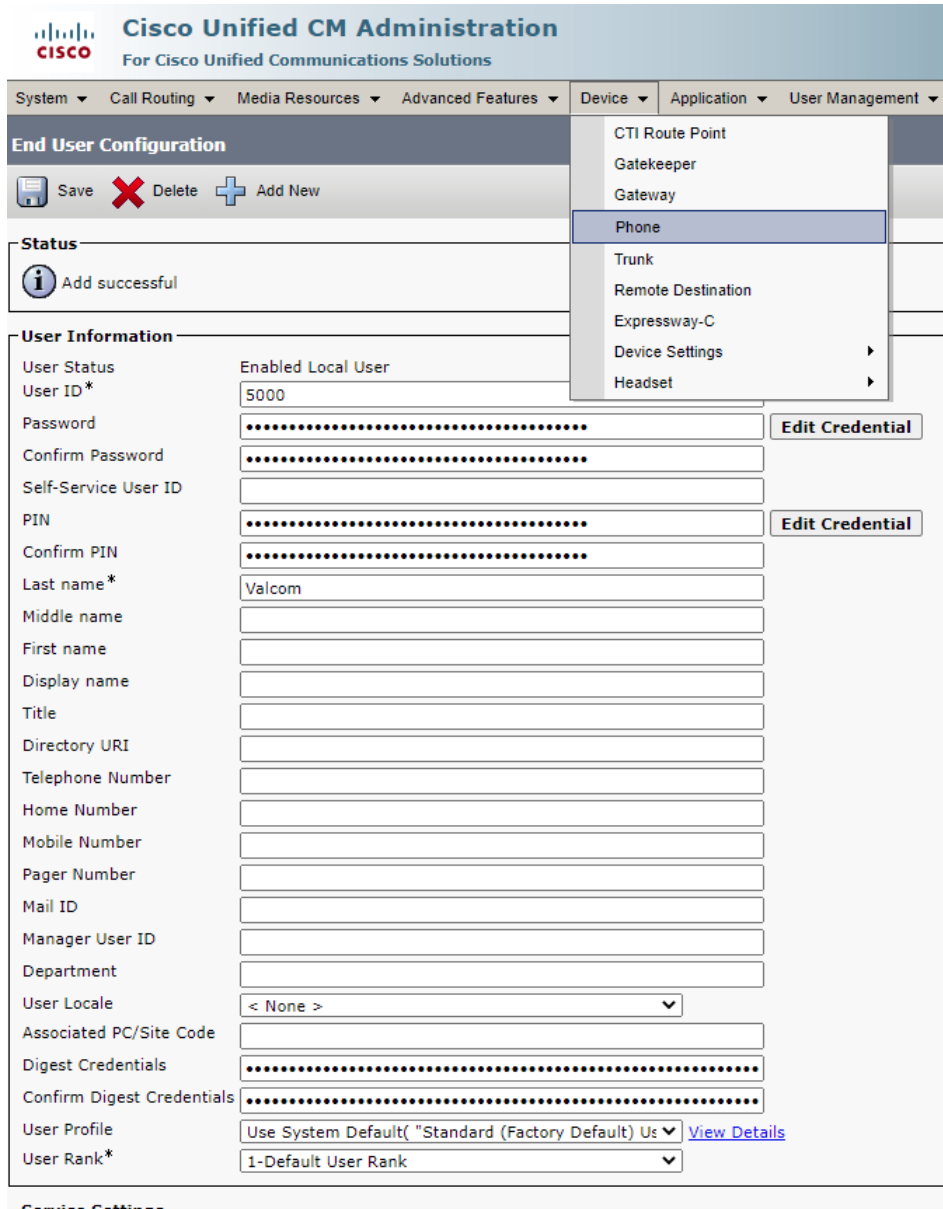
3. Complete the following steps:
 - A) Enter “User ID*” (ex. 5000) –[required for Valcom device]
 - B) Enter “Last name*” (ex. 5000) –[required for Call Manager only]
 - C) Enter “Digest Credentials” (ex. 1234) –[required for Valcom device]

- D) Enter "Confirm Digest Credentials" (ex. 1234) –[required for Valcom device]
- E) Select "Save" at the top of the screen

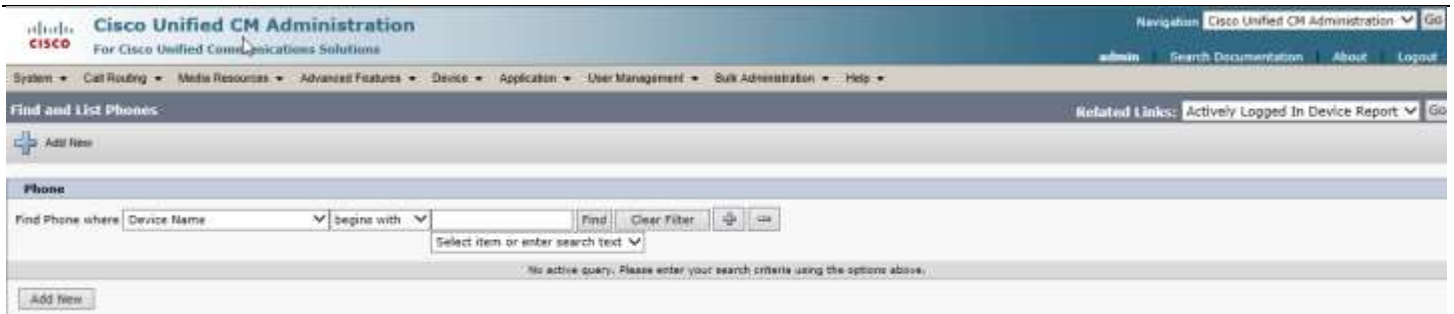
The screenshot displays the Cisco Unified CM Administration web interface. At the top, the Cisco logo and 'Cisco Unified CM Administration' are visible, along with the tagline 'For Cisco Unified Communications Solutions'. A navigation menu includes 'System', 'Call Routing', 'Media Resources', 'Advanced Features', 'Device', 'Application', and 'User Management'. The main heading is 'End User Configuration', with a 'Save' button below it. The 'Status' section shows 'Status: Ready'. The 'User Information' section contains a form with the following fields:

User Status	Enabled Local User
User ID*	5000
Password	
Confirm Password	
Self-Service User ID	
PIN	
Confirm PIN	
Last name*	Valcom
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

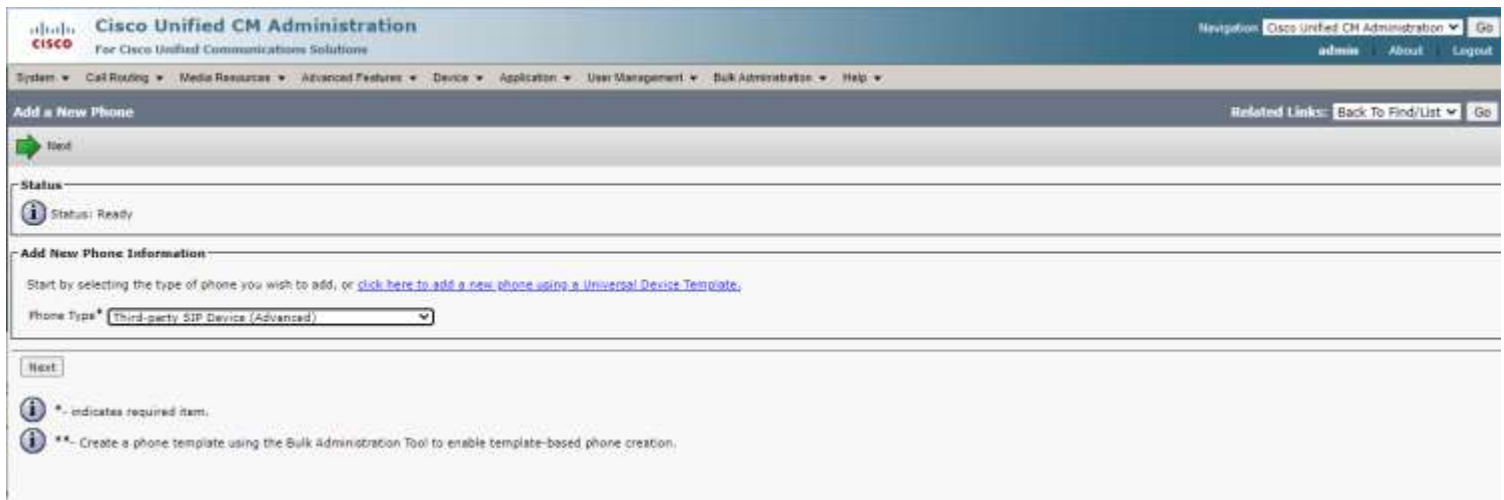
4. Click on “Device”, then click on “Phone”



and Click on “Add New”



5. Select “Third-party SIP Device (Basic)” or “Third-party SIP Device (Advanced)” from the dropdown, then click “Next”
(VIP speakers would be “Basic”, other VIP devices can be either, depending on whether more than one extension/Directory Number will be used on a VIP device)



6. Complete the following steps :
 - in Device Information Section
 - A) Enter “MAC Address*” (ex. 00D05F01D32C, use the MAC address from the Valcom device that will be registered)
 - B) Select “Device Pool*” → “Default” (or what is valid for your installation)
 - C) Select “Phone Button Template*” → “Third-party SIP Device (Basic)” or “Third-party SIP Device (Advanced)”
 - D) Select “Common Phone Profile*” → “Standard Common Phone Profile”
 - E) Select “Location*” → “Hub_None” (or what is valid for your installation)
 - F) Select “Owner” → Anonymous
 - G) Remaining Options in Device Information section can be left as default

- Status

Status: Ready

Phone Type

Product Type: Third-party SIP Device (Advanced)
Device Protocol: SIP

Device Information

Device is not trusted

MAC Address*	<input type="text" value="00D05F01D32C"/>
Description	<input type="text" value="SEP00D05F01D32C"/>
Device Pool*	<input type="text" value="Default"/> View Details
Common Device Configuration	<input type="text" value="< None >"/> View Details
Phone Button Template*	<input type="text" value="Third-party SIP Device (Advanced)"/>
Common Phone Profile*	<input type="text" value="Standard Common Phone Profile"/> View Details
Calling Search Space	<input type="text" value="< None >"/>
AAR Calling Search Space	<input type="text" value="< None >"/>
Media Resource Group List	<input type="text" value="< None >"/>
Location*	<input type="text" value="Hub_None"/>
AAR Group	<input type="text" value="< None >"/>
Device Mobility Mode*	<input type="text" value="Default"/>
Owner	<input type="radio"/> User <input checked="" type="radio"/> Anonymous (Public/Shared Space)
Owner User ID	<input type="text" value=""/>
Mobility User ID	<input type="text" value="< None >"/>
Use Trusted Relay Point*	<input type="text" value="Default"/>
Always Use Prime Line*	<input type="text" value="Default"/>
Always Use Prime Line for Voice Message*	<input type="text" value="Default"/>
Geolocation	<input type="text" value="< None >"/>
<input checked="" type="checkbox"/> Retry Video Call as Audio	
<input type="checkbox"/> Ignore Presentation Indicators (internal calls only)	
<input checked="" type="checkbox"/> Logged Into Hunt Group	
<input type="checkbox"/> Remote Device	

-in Protocol Specific Information Section

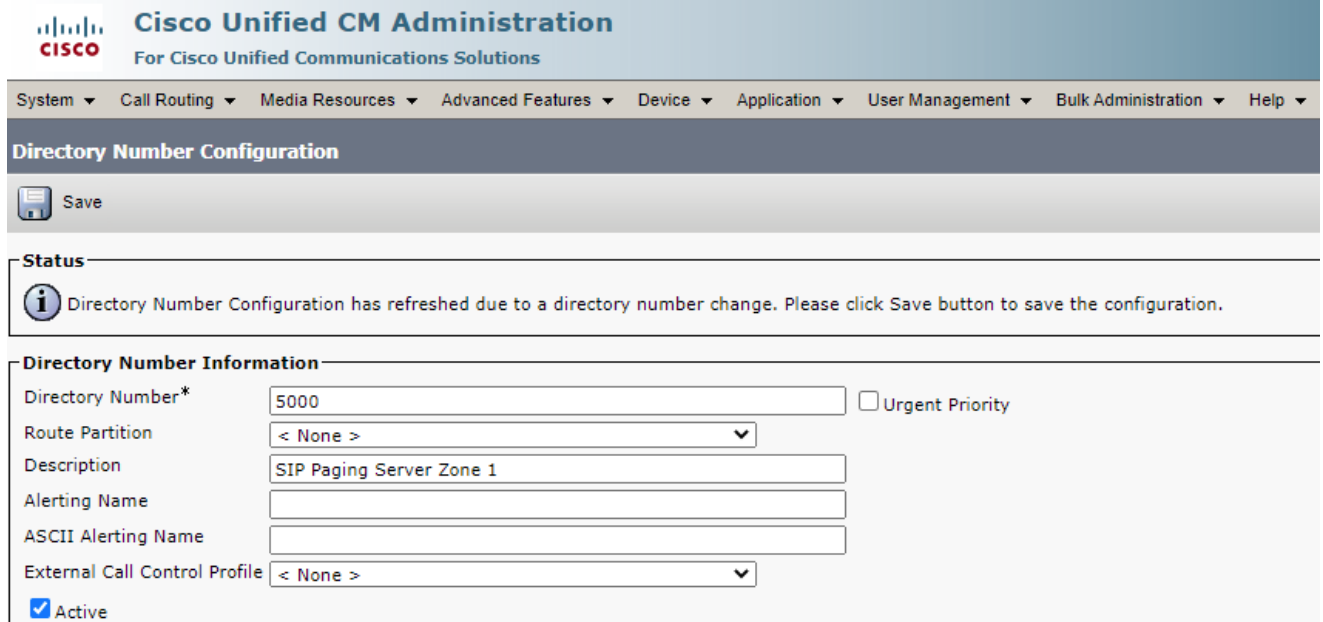
- H) Select “Presence Group*” → “Standard Presence group” (or what is valid for your installation)
- I) Select “MTP Preferred Originating Codec*” → “711ulaw”
- J) Select “Device Security Profile*” → “Third-party SIP Device Basic – Standard SIP Non-Secure Profile” (or a Secure Profile that you may have created –see Step 1 at the beginning of this document)
- K) Select “SIP Profile*” → “Standard SIP Profile”
- L) Select “Media Termination Point Required”
- M) Select “Digest User” → The “User ID” that was created in Step 3A. (ex. 5000)
- N) All other fields can be left at default or configure per your server/site.
- O) Select “Save” at the top of the screen.

The screenshot displays the Cisco Unified CM Administration web interface. At the top, the Cisco logo and 'Cisco Unified CM Administration' are visible, along with a navigation menu including System, Call Routing, Media Resources, Advanced Features, Device, Application, and User Management. The main heading is 'Phone Configuration'. Below this, there is a 'Save' button and a checkbox for 'Use Device Pool Calling Party Transformation CSS (Caller ID For Calls From This Phone)'. A section titled 'Remote Number' contains a dropdown for 'Calling Party Transformation CSS' (set to '< None >') and another checkbox for 'Use Device Pool Calling Party Transformation CSS (Device Mobility Related Information)'. The 'Protocol Specific Information' section includes several dropdown menus: 'BLF Presence Group*' (Standard Presence group), 'MTP Preferred Originating Codec*' (711ulaw), 'Device Security Profile*' (Valcom SIP Device Advanced - Standard SIP Secure), 'Rerouting Calling Search Space' (< None >), 'SUBSCRIBE Calling Search Space' (< None >), 'SIP Profile*' (Standard SIP Profile), and 'Digest User' (5000). There is also a 'View Details' link next to the SIP Profile dropdown. Below these are several checkboxes: 'Media Termination Point Required' (checked), 'Unattended Port', 'Require DTMF Reception', 'Allow Presentation Sharing using BFCP', and 'Allow iX Applicable Media'. The 'MLPP and Confidential Access Level Information' section has three dropdown menus: 'MLPP Domain' (< None >), 'Confidential Access Mode' (< None >), and 'Confidential Access Level' (< None >). A 'Save' button is located at the bottom of the form.

7. Select “Line [1] – Add a new DN” under “Association”.



8. Complete the following steps:
 - in Directory Number Information
 - A) Enter “Directory Number*” (ex. 5000)
 - B) Route Partition use default or what is applicable to your site
 - C) Enter “Description” (ex SIP Paging Server Zone 1)
 - D) Check the Active checkbox, if not already checked
 - in Directory Number Settings
 - E) Select “Presence Group*” → “Standard Presence group” (or what is valid for your installation)



The screenshot shows the Cisco Unified CM Administration interface. At the top, there is a navigation menu with options: System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. Below this is a header for "Directory Number Configuration" with a "Save" button. A status message indicates that the configuration has refreshed due to a directory number change and prompts the user to click the Save button. The main configuration area, titled "Directory Number Information", contains the following fields:

- Directory Number*: 5000
- Route Partition: < None >
- Description: SIP Paging Server Zone 1
- Alerting Name: (empty)
- ASCII Alerting Name: (empty)
- External Call Control Profile: < None >
- Urgent Priority: (unchecked)
- Active: (checked)

-in Line 1 on Device SEP00D05F01D32C

F) Key in "Display (Caller ID)" with a name or number to identify this (DN) extension **useful if using talkback speakers that can call into the Call Manager.*

-in Multiple Call/Call Waiting Settings on Device SEP00D05F01D32C

G) Enter "Maximum Number of Calls*" → "2"

H) Enter "Busy Trigger*" → "2"

-in Forwarded Call Information Display on Device SEP00D05F01D32C

I) Check "Caller Name"

J) Check "Dialed Number"

K) Select "Save" at the bottom or top of the screen

L) Click "Apply Config" at top of screen

Save

Target (Destination)

HLRP Calling Search Space

HLRP No Answer Ring Duration (seconds)

Confidential Access Mode

Confidential Access Level

Call Control Agent Profile

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*

Line 1 on Device SEP00D05F01D32C

Display (Caller ID) Display text for a line appearance is intended for displaying text such as a name instead of a directory number for calls. If you specify a number, the person receiving a call may not see the proper identity of the caller.

ASCII Display (Caller ID)

External Phone Number Mask

Monitoring Calling Search Space

Multiple Call/Call Waiting Settings on Device SEP00D05F01D32C

Note: The range to select the Max Number of calls is: 1-15

Maximum Number of Calls*

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

Forwarded Call Information Display on Device SEP00D05F01D32C

Caller Name

Caller Number

Redirected Number

Dialed Number

Save

M) Click Related Links: Configure Device Go button to return to device screen

Save **Delete** **Reset** **Apply Config** **Add New**

The screenshot displays the Cisco Unified CM Administration web interface. At the top, the navigation menu includes System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The main header reads "Cisco Unified CM Administration For Cisco Unified Communications Solutions".

The "Phone Configuration" section is active, showing a toolbar with Save, Delete, Copy, Reset, Apply Config, and Add New buttons. Below this, the "Status" section indicates "Status: Ready".

The "Association" panel on the left lists eight lines, each with a "Modify Button Items" button and a link to "Add a new DN".

The main configuration area is divided into several sections:

- Phone Type:** Product Type: Third-party SIP Device (Advanced), Device Protocol: SIP.
- Real-time Device Status:** Registration: Unknown, IPv4 Address: None.
- Device Information:**
 - Device is Active:
 - Device is not trusted:
 - MAC Address*: 00D05F01D32C (SEP00D05F01D32C)
 - Description: SEP00D05F01D32C
 - Device Pool*: Default (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 (View Current Device Mobility Settings)
 - Owner: User Anonymous (Public/Shared Space)
 - Owner User ID: < None >
 - Mobility User ID: < None >
 - Use Trusted Relay Point*: Default
 - Always Use Prime Line*: Default
 - Always Use Prime Line for Voice Message*: Default
 - Geolocation: < None >
 - Retry Video Call as Audio:
 - Ignore Presentation Indicators (internal calls only):
 - Logged Into Hunt Group:
 - Remote Device:

- N) You can repeat steps A-F if configuring more than 1 extension
- O) When done Click "Apply Config"



Cisco Unified Communications Manager 12.5 SIP Configuration Guide

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9. Open the VIP-102B tool interface for the Valcom SIP enabled VIP device.

Note: The information contained in this guide is limited to configuration of the “SIP” tab in the VIP-102B IP Solutions Setup Tool for the Valcom VIP device that is to be registered to the SIP server. More information on Valcom VIP device configuration, such as IP address assignment, relay activation, etc, may be found in the VIP-102B Reference Manual. This document may be downloaded from our website at <http://www.valcom.com>

In order to Register:

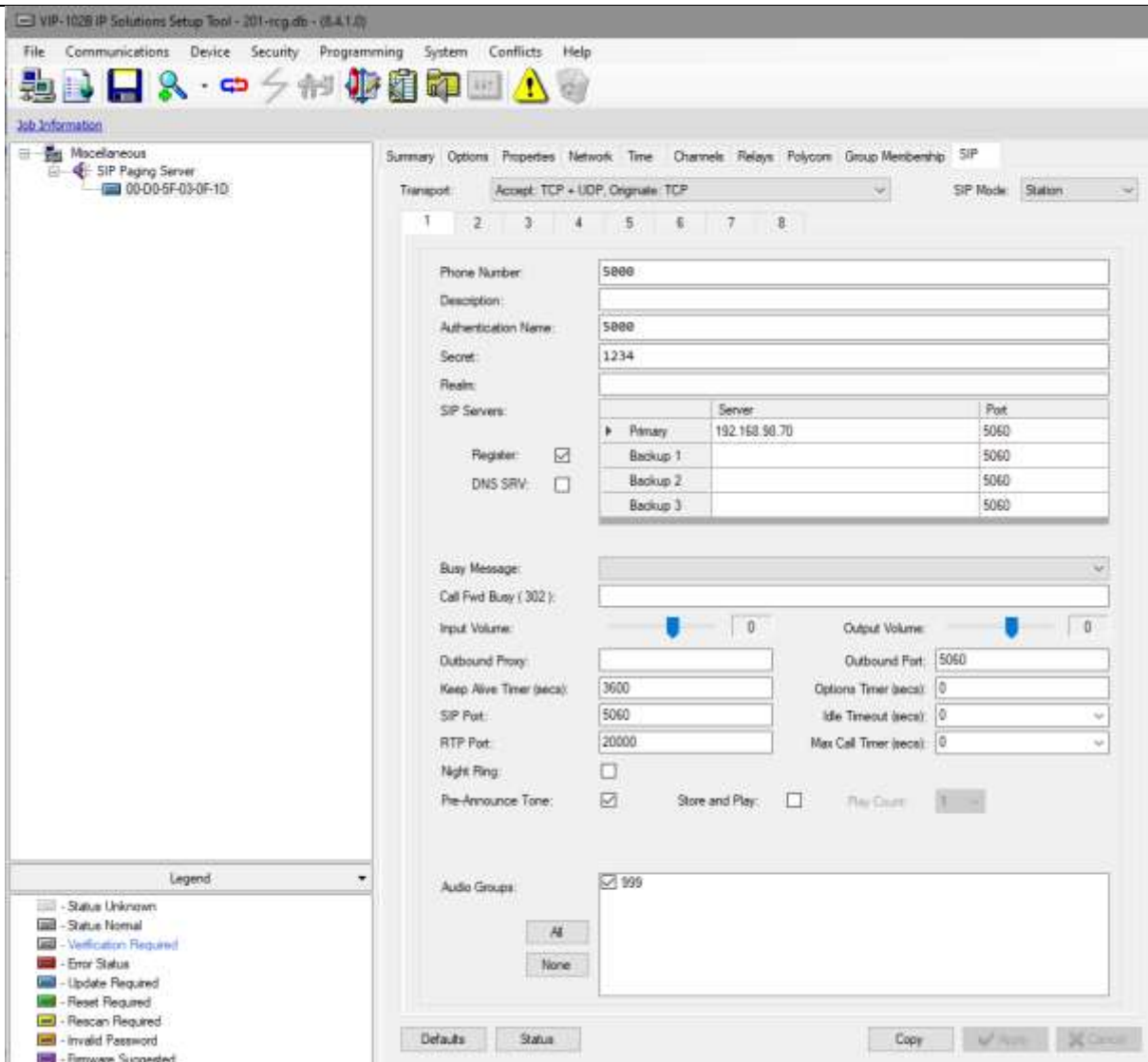
Required Fields: Phone Number, Authentication Name, Secret, SIP Server (primary), Register, SIP Server Port, SIP Port, RTP Port

Optional Fields: Description, Realm, SIP Server Backup 1, 2, and 3, DNS SRV, CID Name, CID Number.

*In our example, the SIP Server IP address is the same as our Cisco Call Manager, “192.168.98.70”. If using a host name here you must specify at least one DNS server on the Network tab to resolve the name.
Phone Number is the same as our Directory Number in the Cisco Call Manager configuration, “5000”.
Secret is the same as our Digest Credentials in the Cisco Call Manager configuration, “1234”.
SIP Server Port is the port number, on which the Cisco Call Manager SIP server is listening for SIP data.
SIP Port is the port number, on which the Valcom VIP device is listening for SIP data. By default this is set for “5060”.
RTP Port is the port number, on which the Valcom VIP device is set to send/receive audio packets, via SIP. By default this is set for “20000”. All other optional fields may be used based on your server/site requirements.*

For this particular device, the SIP paging server, other fields on the SIP tab relate to functionality of the device. Definition of these fields may be found in the VIP-102B Reference Manual under the SIP Tab (VIP-201, VIP-204). This document may be downloaded from our website at <http://www.valcom.com>

When the Valcom VIP device configuration is complete, select the “Update Changed Devices” button, at the upper left. When update is complete, click reset, to reboot the device.



- To confirm a successful configuration, return to Call Manager and click on "Device", then Phone, then locate the VIP device in the search results. If successfully registered, the status column should show the VIP device is registered to the IP address of the Call Manager with the VIP device's IP address in the next column under "IP Address"