Cisco Unified Communications Manager 14.0 SIP Configuration Guide

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".
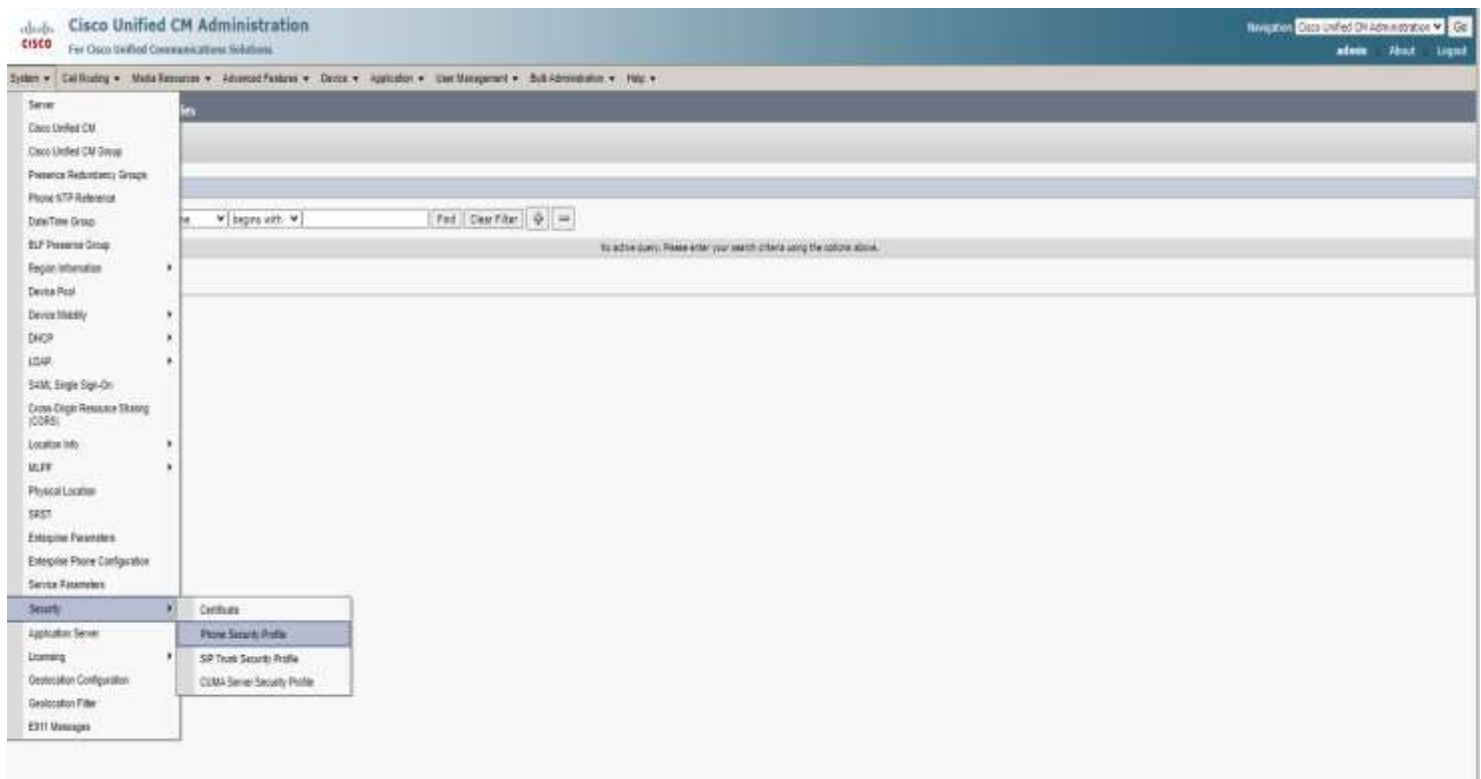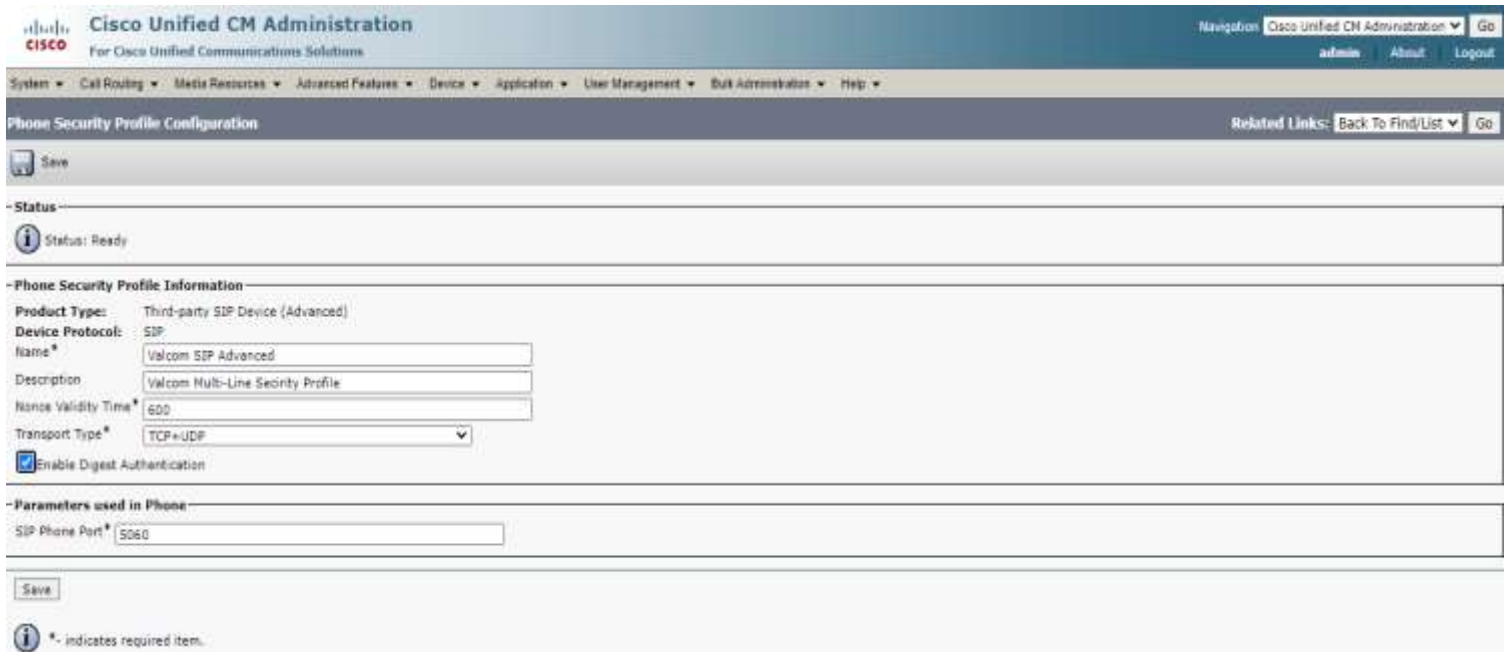
2. Click on "Add New"

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.
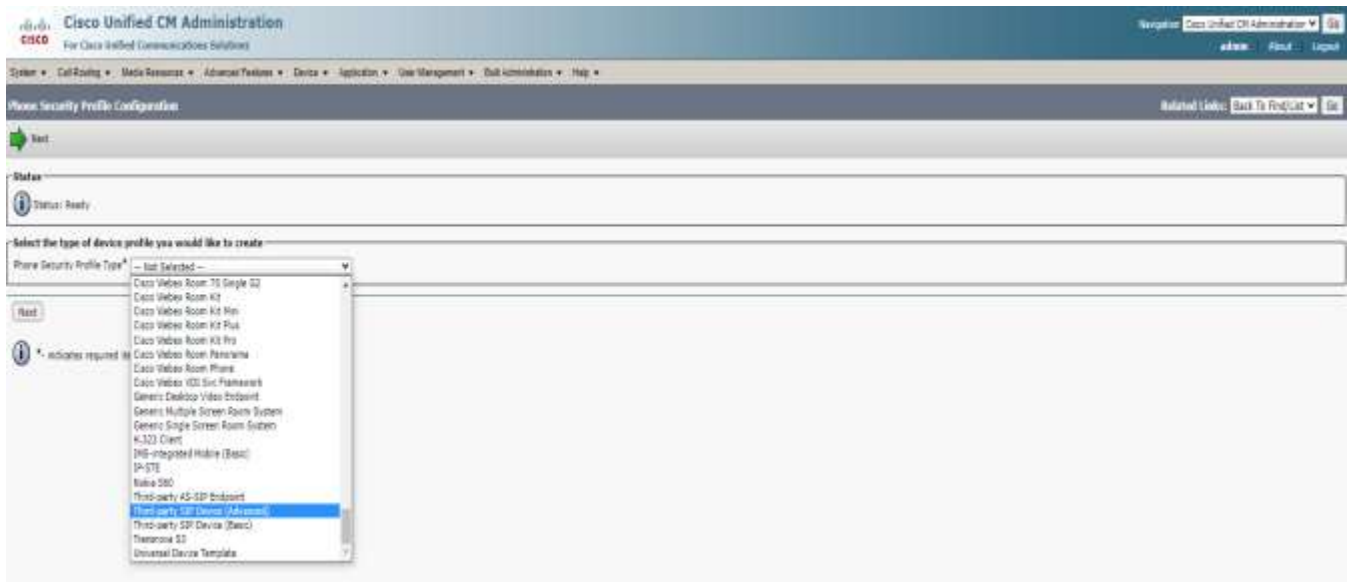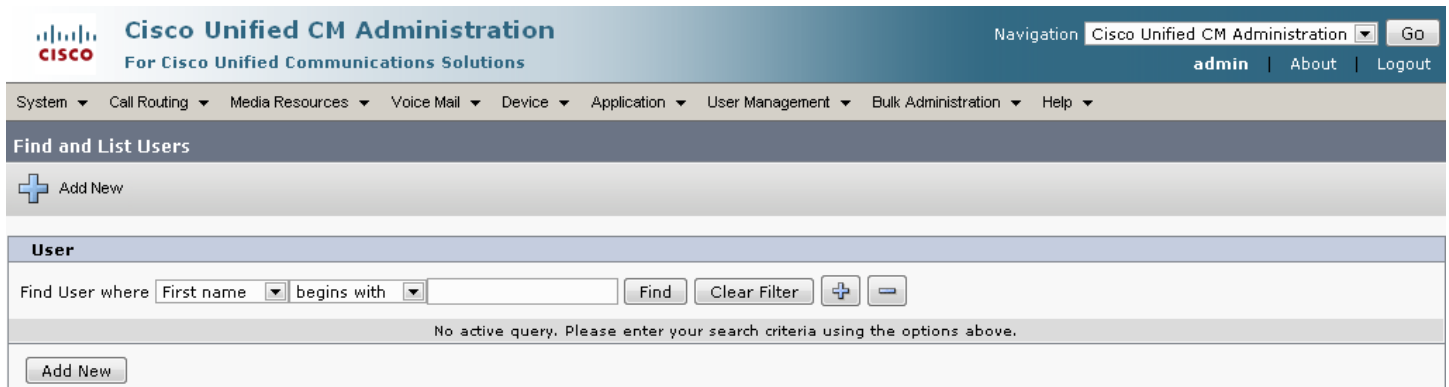
4. Enter the Phone Security Profile Information.
> A) Enter "Name*" (ex. Valcom SIP Advanced)
> B) Enter "Nonce Validity Time*" in seconds (default 600)
> C) For "Transport Type*" select "UDP" or TCP+UDP from the dropdown list
> D) Check the box for "Enable Digest Authentication"
> E) The "SIP Phone 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 following steps outline the typical device configuration process:

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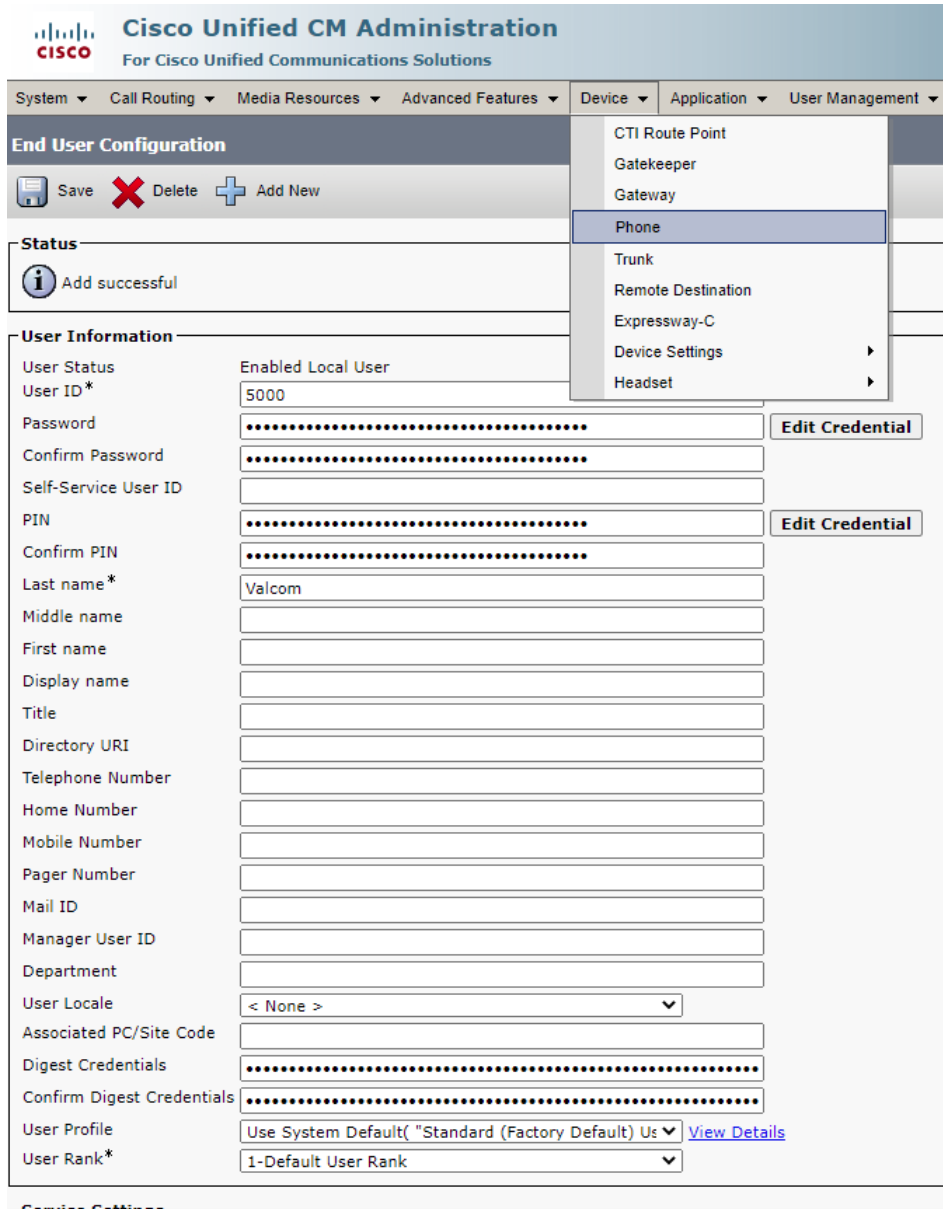
2. Click on "Add New"

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*]
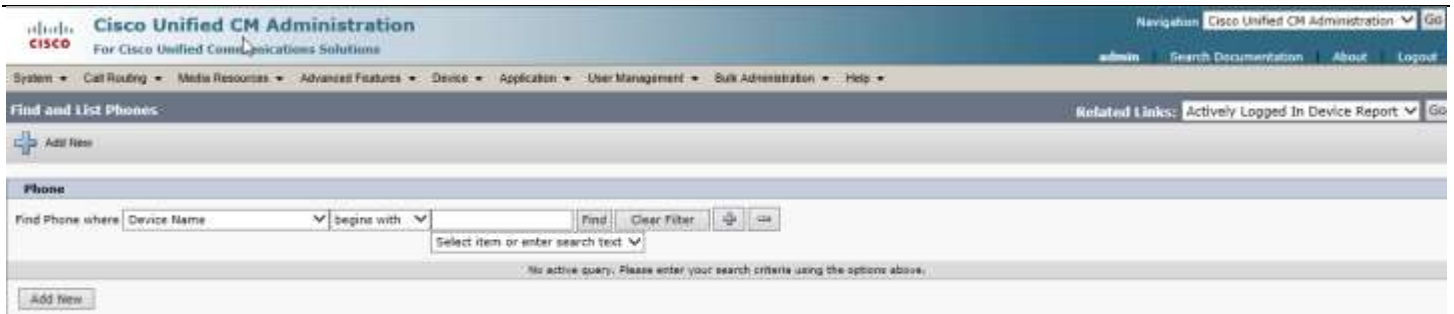
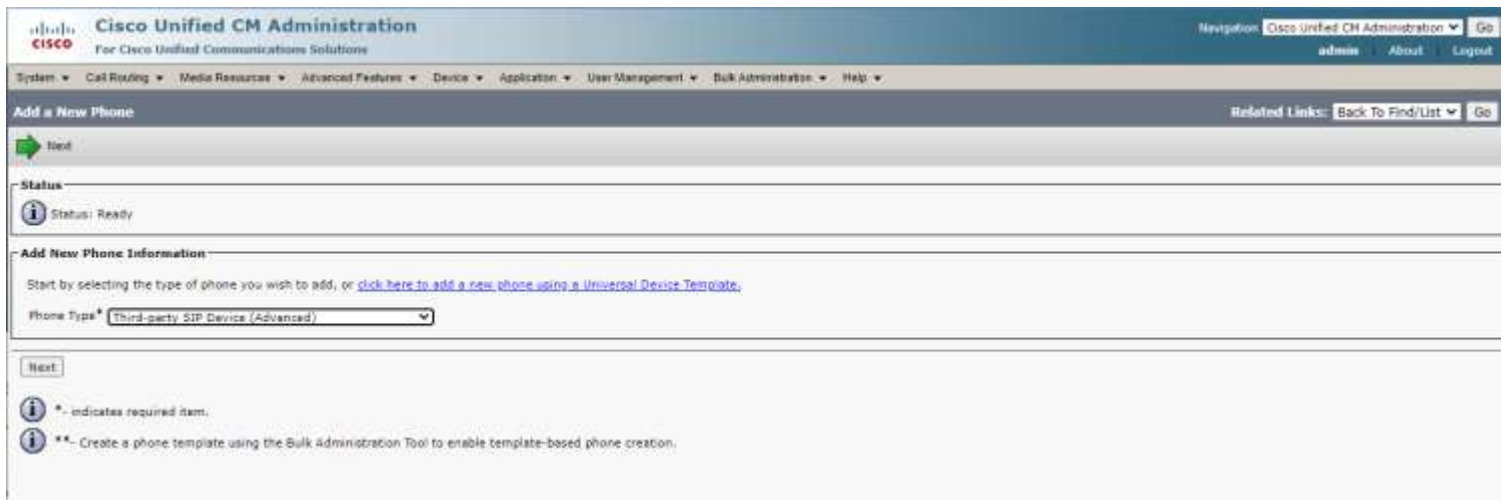E) Select "Save" at the top of the screen

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 is 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*	00D05F01D32C
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 ⌄
Owner	○ User ● Anonymous (Public/Shared Space)
Owner User ID	⌄
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)
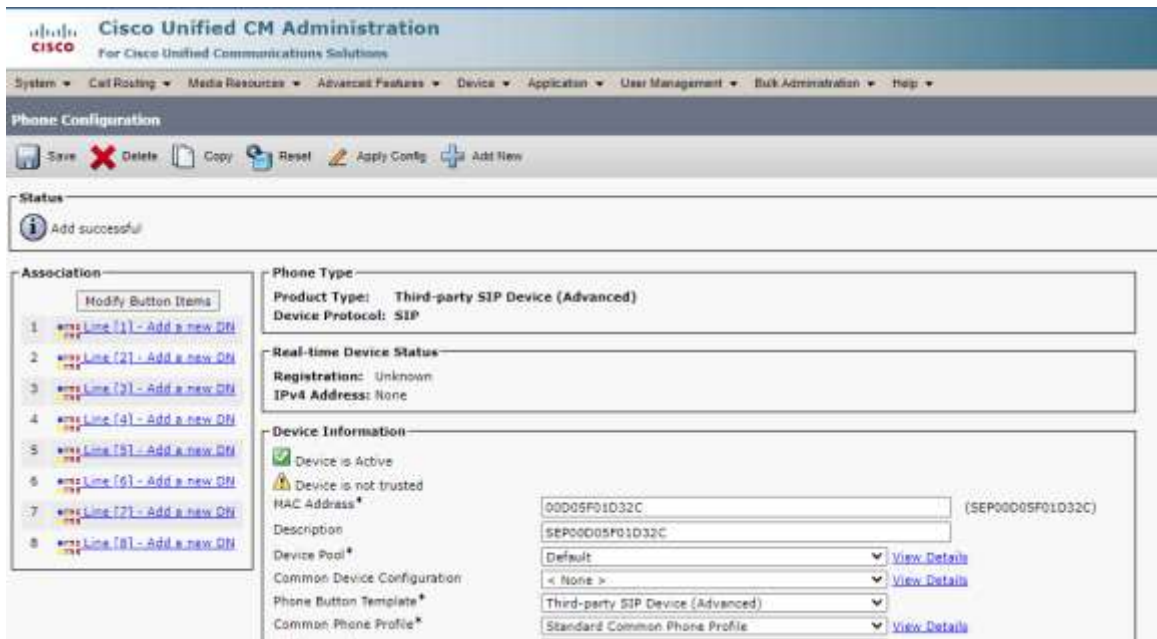
☑ Logged Into Hunt Group

☐ Remote Device

-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.

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*)

-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

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

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

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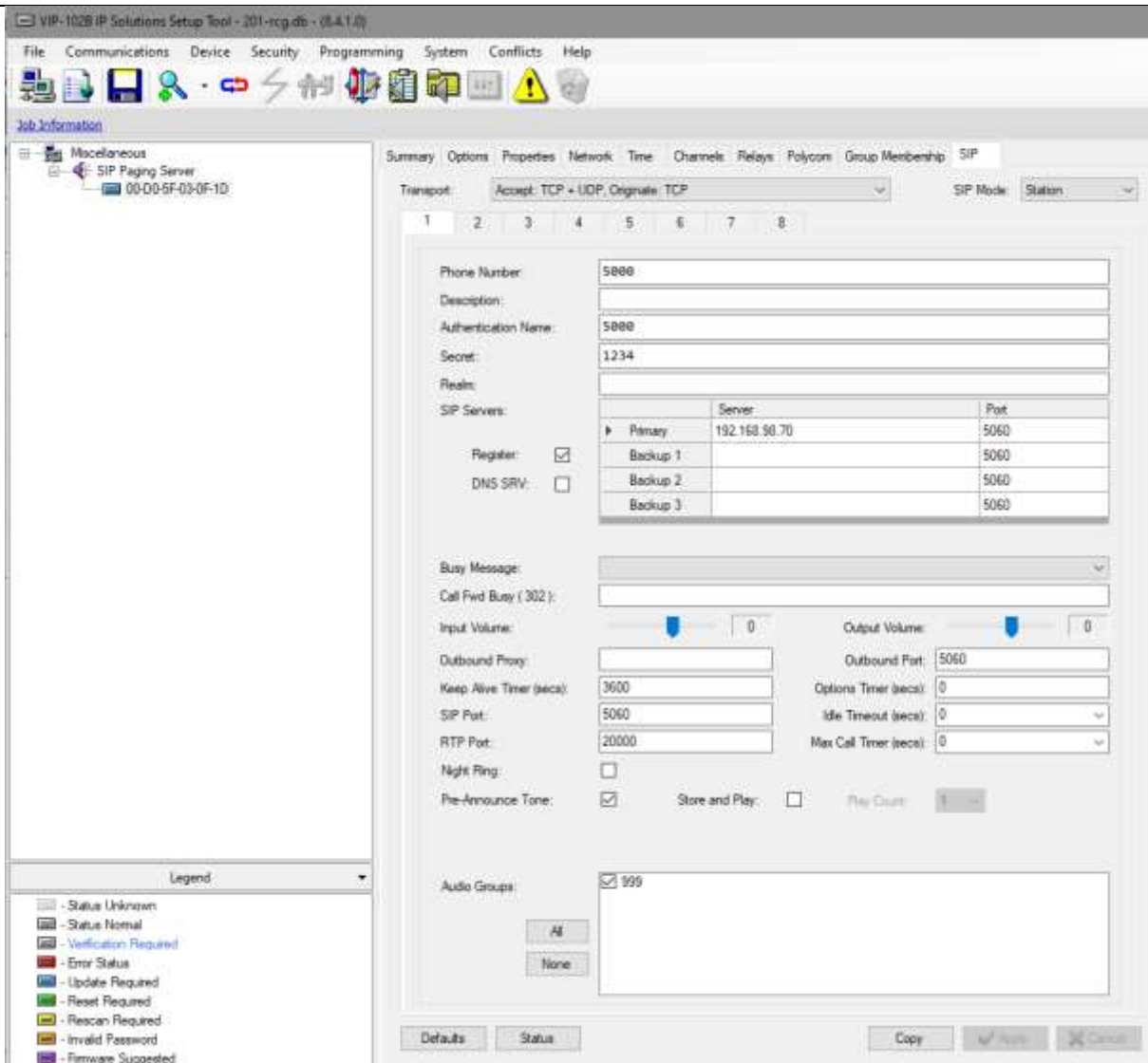
10.   To confirm a successful configuration, return to Call Manager and click on "Device", then Phone, then locate the VIP device in the search results. If successfully registered, the status column should show the VIP device is registered to the IP address of the Call Manager with the VIP device's IP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