

## I. Introduction

This Cisco UCM Integration Guide provides general instructions for integration of the **VOIP-600 Series Phone** with a Cisco Unified Communications Manager. It is recommended to read this instruction set completely before starting any installation. For detailed VOIP-600 Series Phone setup instructions, please consult the **VOIP-600 Series Phone Manual**.

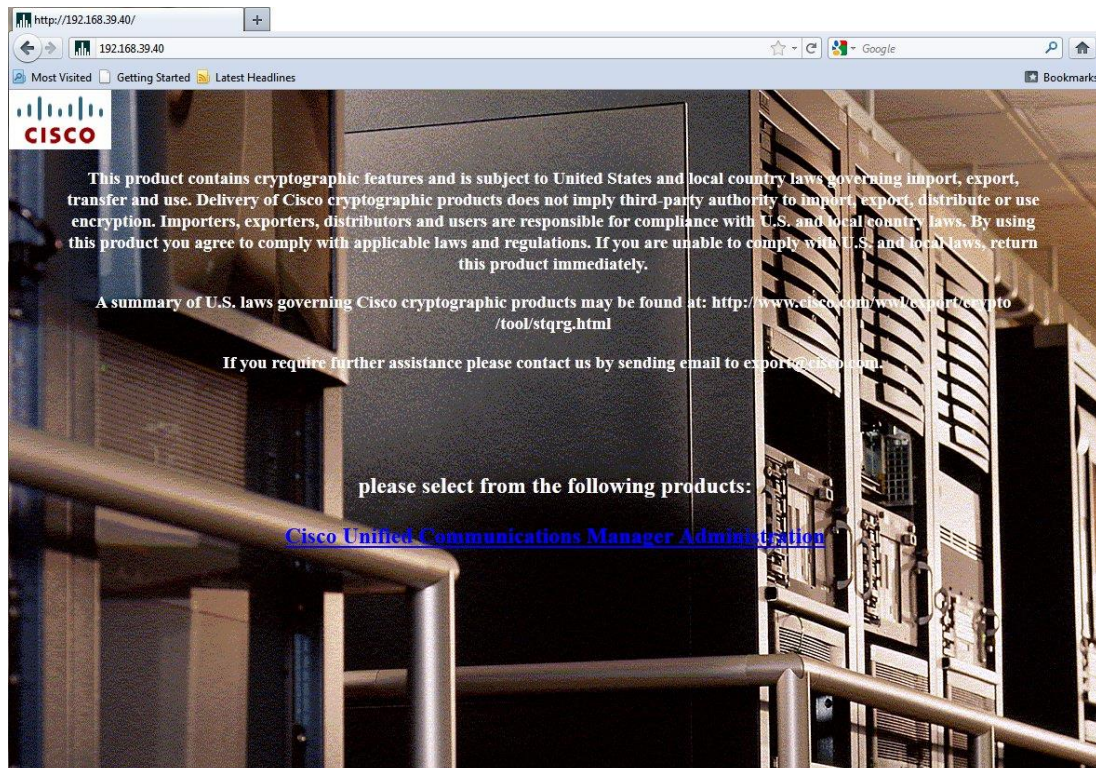
## II. Prerequisites

- Cisco Unified Communications Manager, version 8.0 pre-installed
- CM and TFTP services licensed and enabled on CUCM
- SIP Device Licensing for Third-party SIP (Basic) devices
- Network access to the CUCM Server, **VOIP-600 Series Phones** and all network services (SIP, TFTP, HTTP, FTP, DNS, RTP/SRTP)

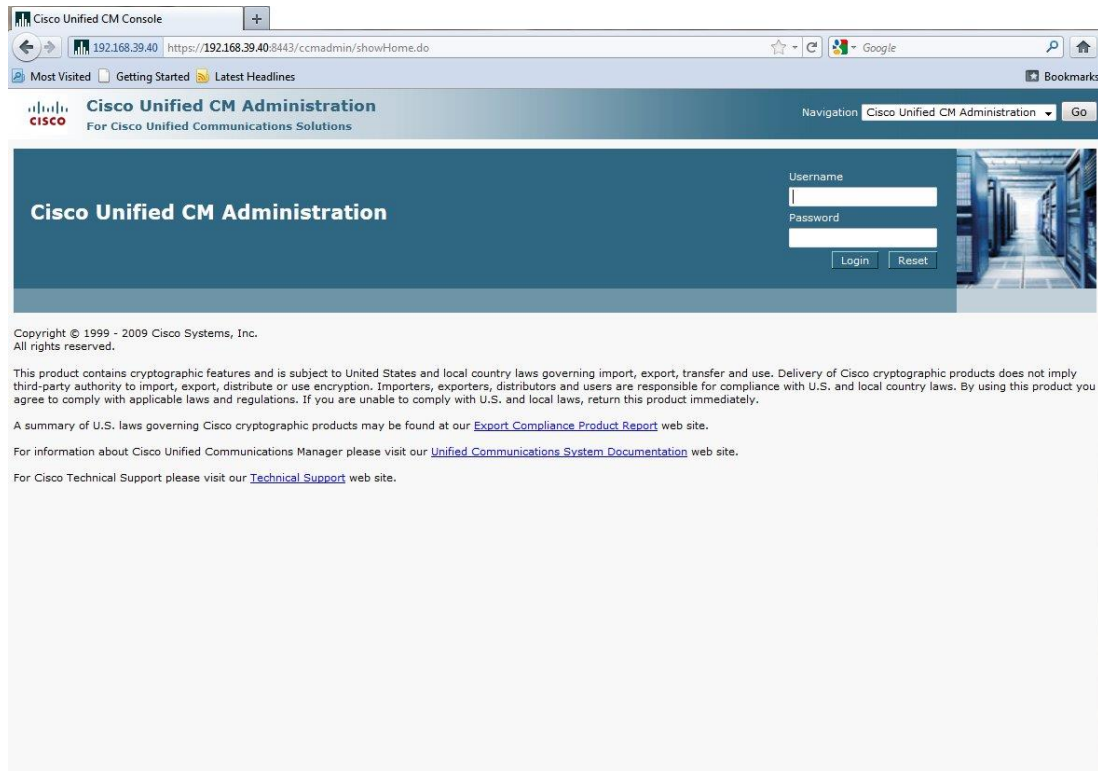
## III. CUCM Basic Configuration

Basic instructions for integrating a **VOIP-600 Series Phone** with a Cisco UCM are included. Advanced setup of Cisco UCM features are outside the scope of this document.

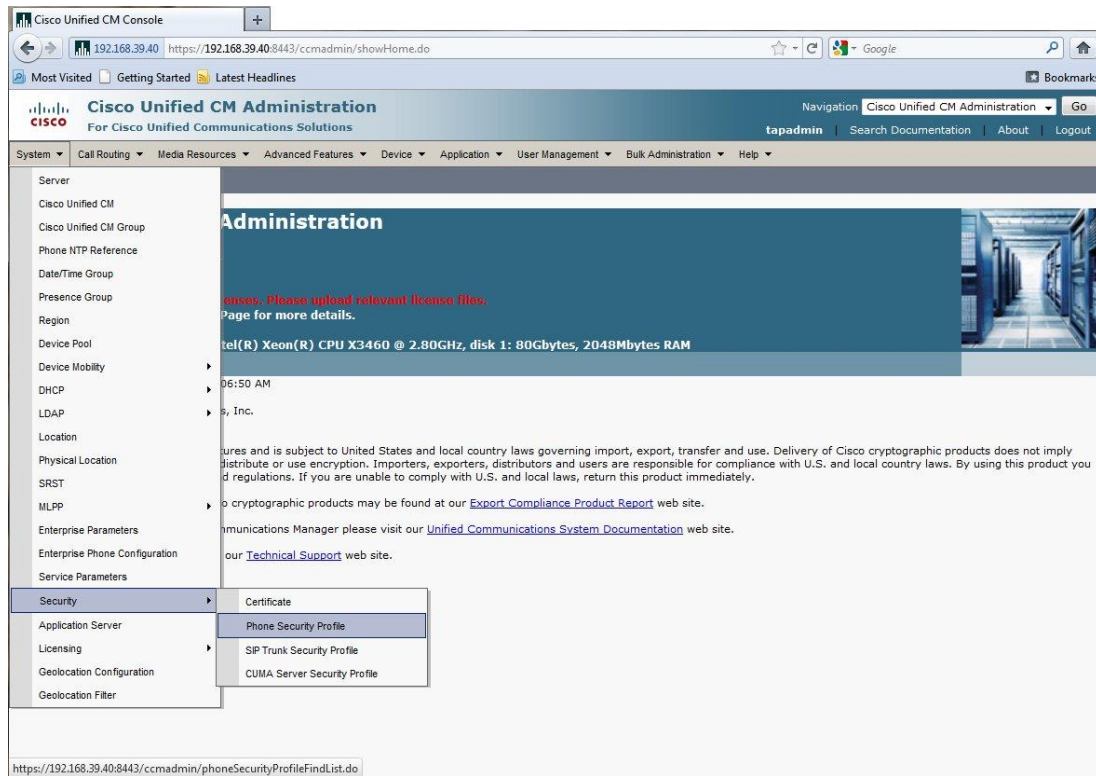
1. Using a web browser, enter the IP address (or FQDN if configured) of the Cisco UCM Server in the address bar:



**2. Login to Cisco Unified CM Administration:**



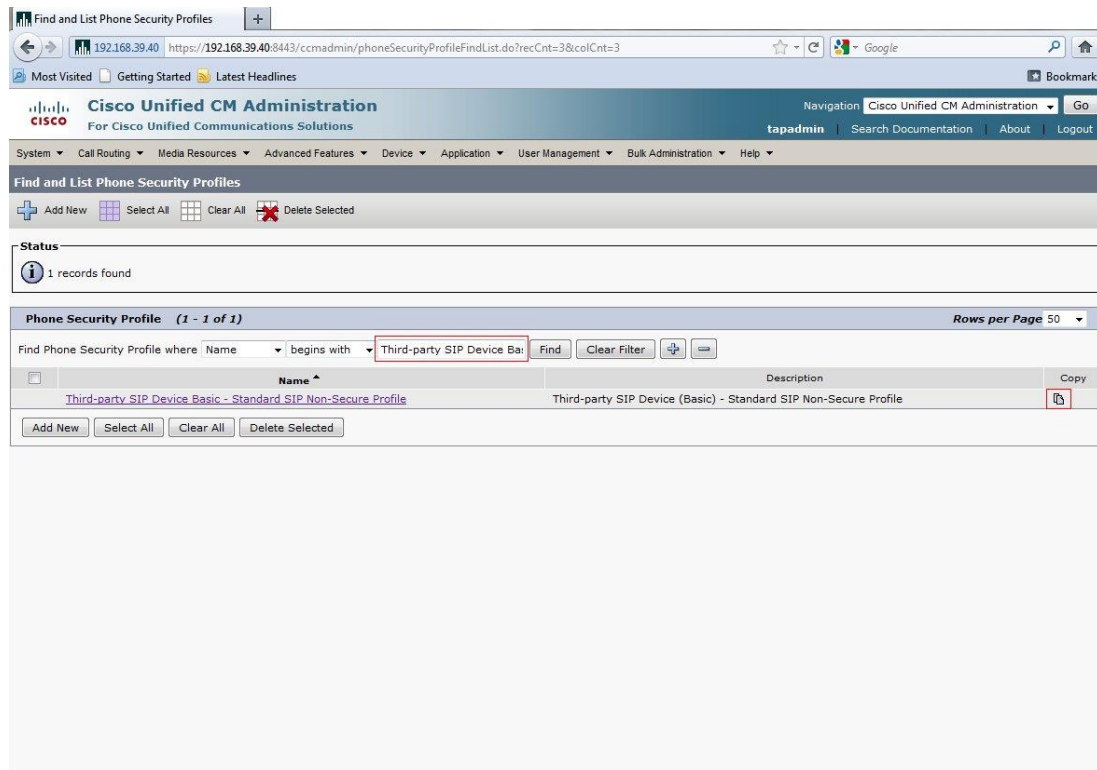
3. A Phone Security Profile with Digest Authentication enabled is required for VOIP-600 integration. Begin by selecting **System > Security > Phone Security Profile** in the CUCM menu:



- In the Find Phone Security Profile field, enter **Third-party SIP Device Basic** and press **Enter**.

You should note a single entry for **Third-party SIP Device Basic – Standard SIP Non-Secure Profile**.

Click the **Copy** button to the right of this entry.



5. Modify the new Phone Security Profile Name to **Third-party SIP Device Basic – Digest Required**.

Modify the Description to **Third-party SIP Device (Basic) – Digest Required**.

Check the **Enable Digest Authentication** box, and click **Save**.

Phone Security Profile Configuration

Navigation: Cisco Unified CM Administration | Go

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

Phone Security Profile Configuration | Related Links: Back To Find/List | Go

Save

**Status**

Status: Ready

**Phone Security Profile Information**

Product Type: Third-party SIP Device (Basic)

Device Protocol: SIP

Name\*: Third-party SIP Device Basic - Digest Required

Description: Third-party SIP Device (Basic) - Digest Required

Nonce Validity Time\*: 600

Transport Type\*: TCP+UDP

Enable Digest Authentication

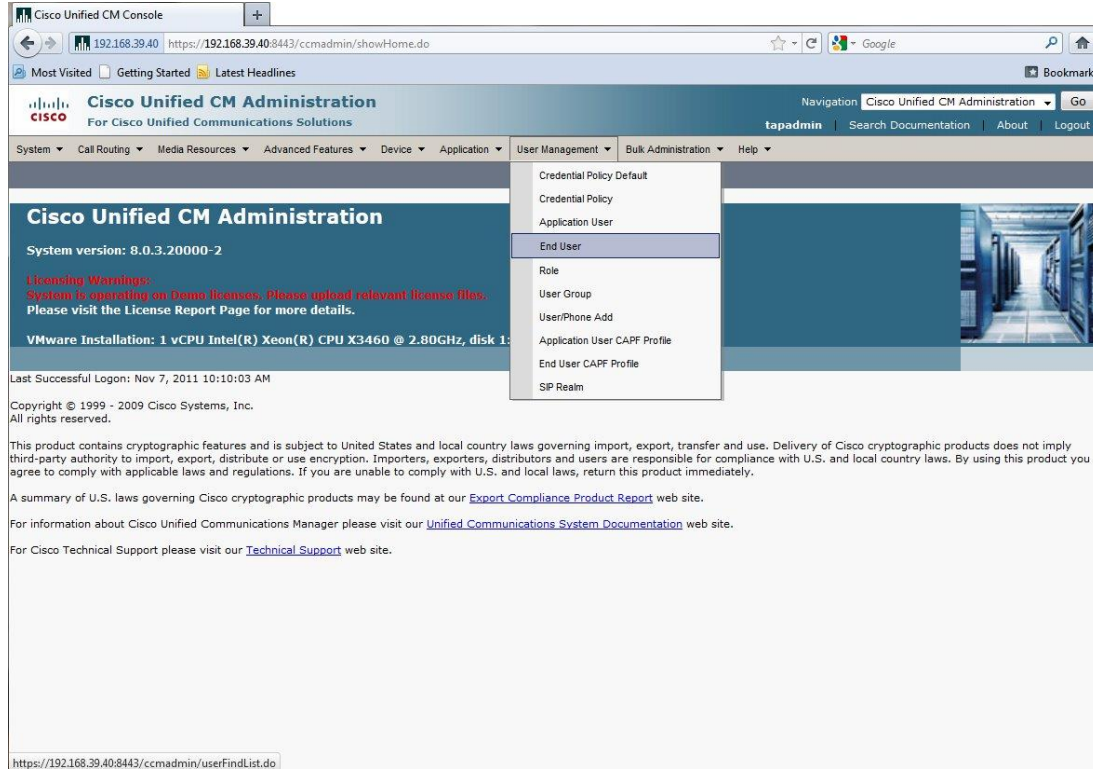
**Parameters used in Phone**

SIP Phone Port\*: 5060

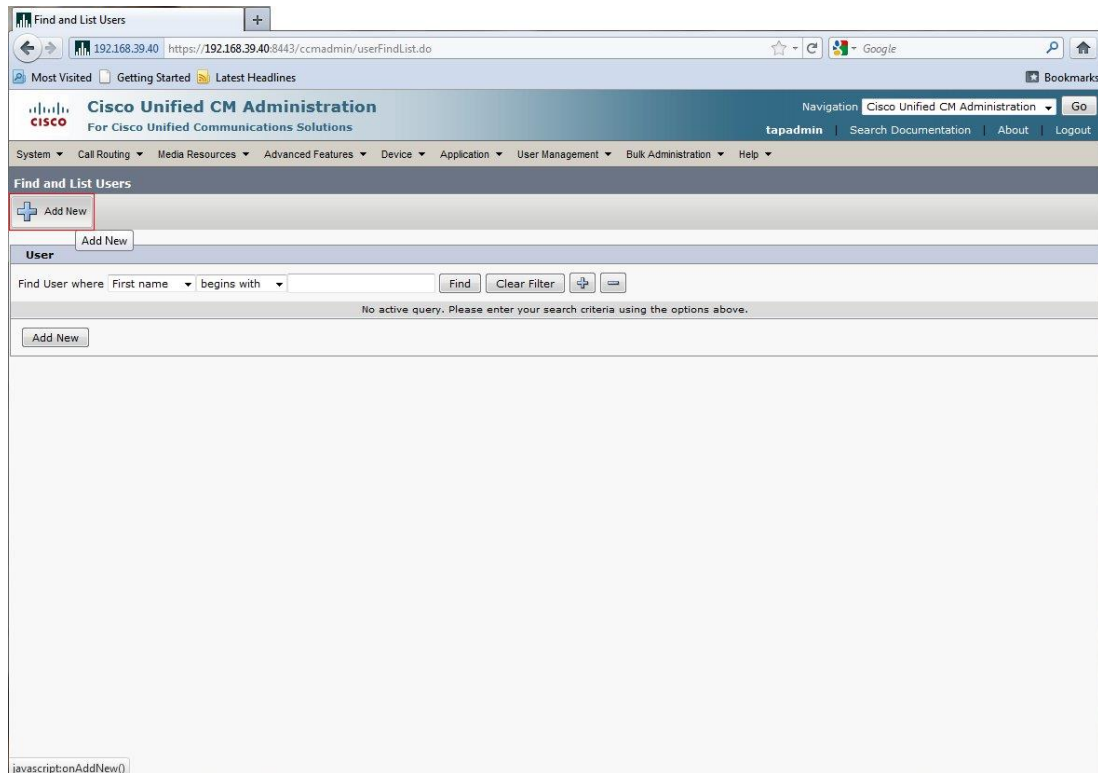
Save

\* - indicates required item.

- Each VOIP-600 Series Phone should have a unique End User. In the CUCM main menu, select **User Management > End User**.



7. Create a new End User. First, click **Add New**.



8. Enter the required fields to create a new End User:

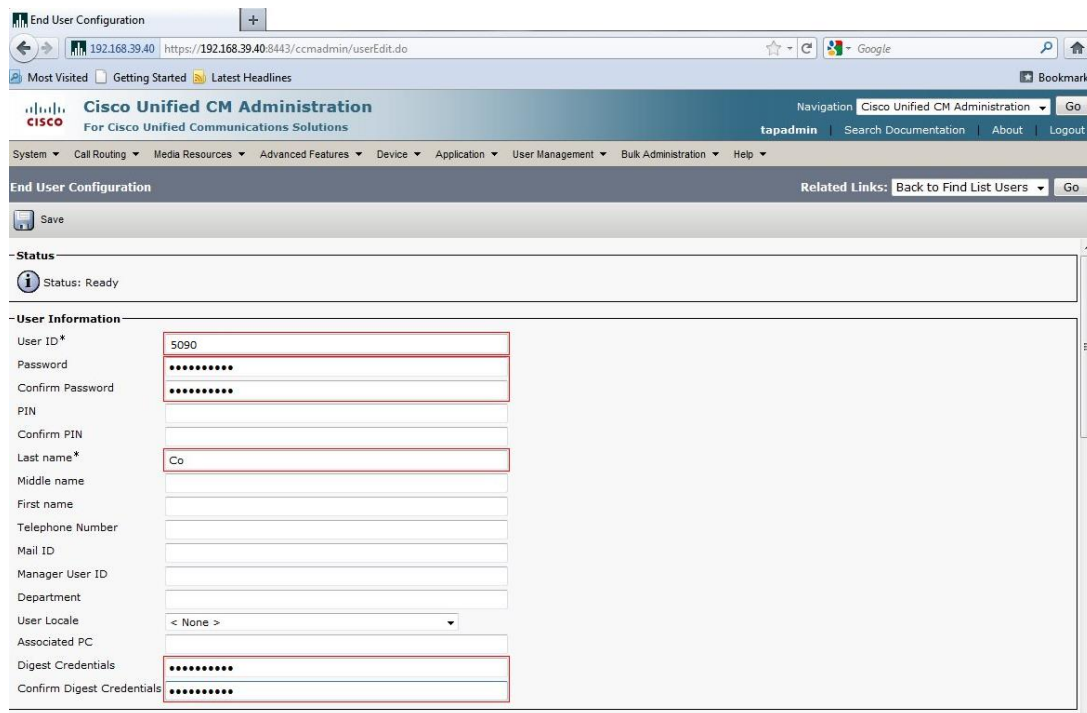
User ID: A unique username for each **VOIP-600 Series Phone**.

*Cisco recommends that the Username of the VOIP-600 Series Phone should be the same as the DN or the 'Directory Number' of the VOIP-600 in the CUCM.*

Password: A unique password for each **VOIP-600 Series Phone**.

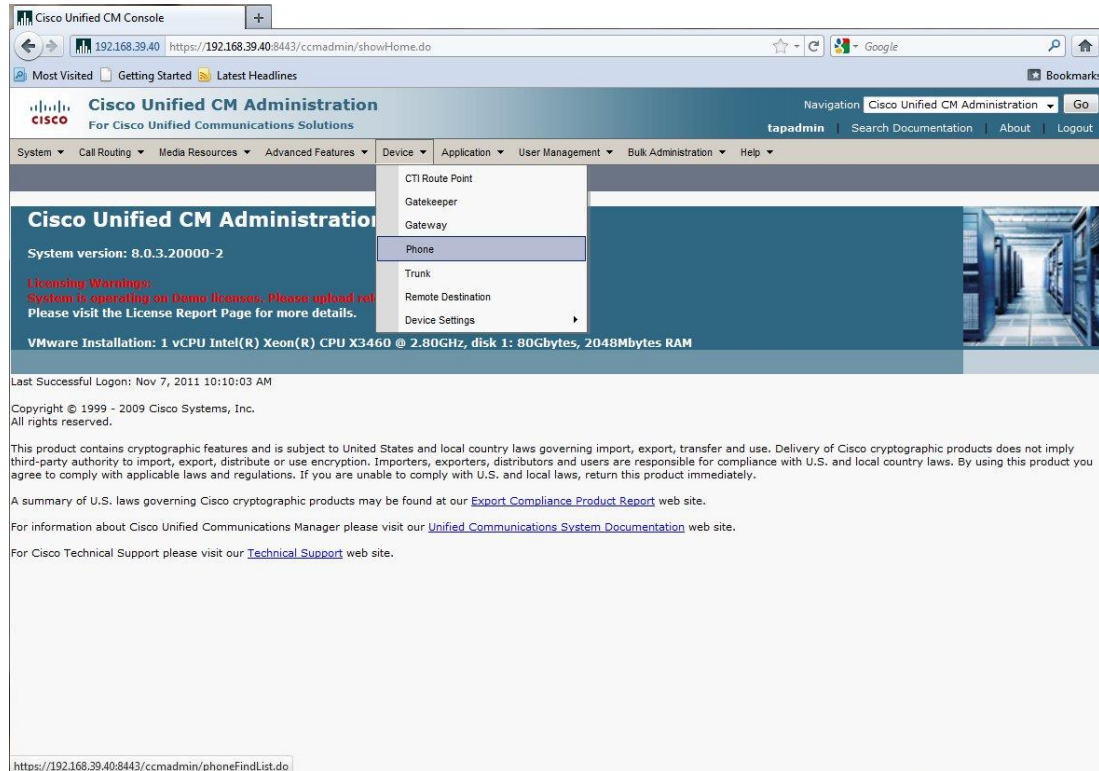
Last name: (required by CUCM).

Digest Credentials: (required by **VOIP-600 Series Phone**) It is recommended to match the Digest Credentials with the Password for manageability.

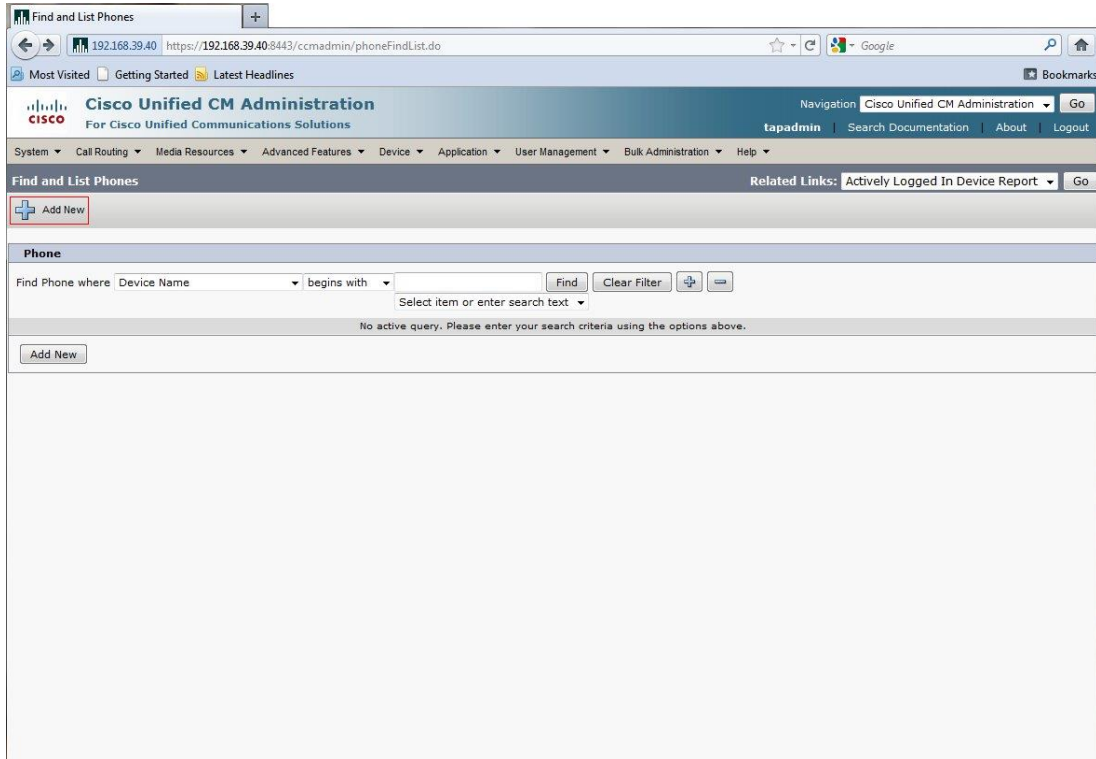




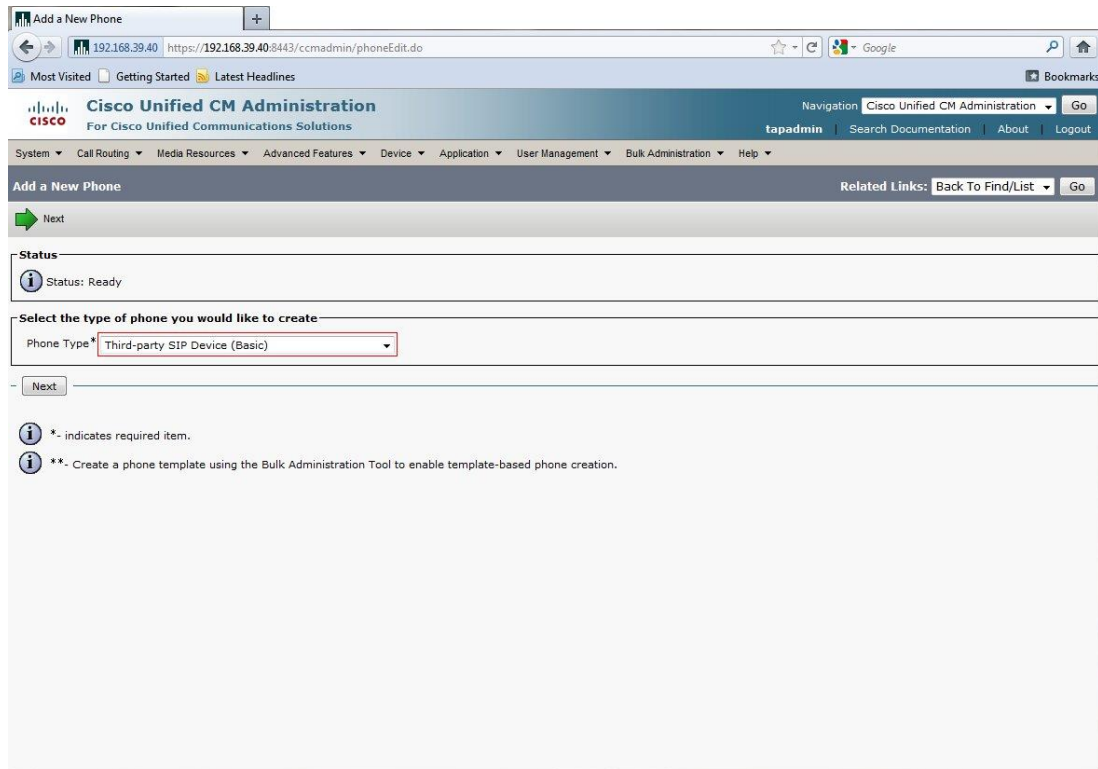
Next, add each **VOIP-600 Series Phone** as a Device in CUCM. In the CUCM main menu, select **Device > Phone**.



**9. Click Add New.**



10. Select **Third-party SIP Device (Basic)** from the Phone Type menu.



11. In the Phone Configuration page enter the following fields. Then click **Save**.

MAC Address: H/W address of VOIP-600

Description: (Auto-filled)

Phone Button Template: **Third-party SIP Device (Basic)**

Common Phone Profile: **Standard Common Phone Profile**

Owner User ID: Enter the User ID(same as the DN) created in Step 8

Presence Group: **Standard Presence Group**

Device Security Profile: **Third-party SIP Device Basic – Digest Required**

SIP Profile: **Standard SIP Profile**

Digest User: Enter the User ID created in Step 8

Phone Configuration

192.168.39.40 https://192.168.39.40:8443/ccmadmin/phoneEdit.do

Cisco Unified CM Administration

Navigation: Cisco Unified CM Administration

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

Phone Configuration Related Links: Back To Find/List Go

Save

**Status**

Status: Ready

**Phone Type**

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

**Device Information**

Device is not trusted

MAC Address\* 001EEB000118

Description SEP001EEB000118

Device Pool\* Default [View Details](#)

Common Device Configuration < None > [View Details](#)

Phone Button Template\* Third-party SIP Device (Basic)

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 [View Current Device Mobility Settings](#)

Owner User ID 5090

Use Trusted Relay Point\* Default

Always Use Prime Line\* Default

Always Use Prime Line for Voice Message\* Default

Calling Party Transformation CSS < None >

Geolocation < None >

Use Device Pool Calling Party Transformation CSS

Ignore Presentation Indicators (internal calls only)

Logged Into Hunt Group

Remote Device

**Protocol Specific Information**

Presence Group\* Standard Presence group

MTP Preferred Originating Codec\* 711ulaw

Device Security Profile\* Third-party SIP Device - Digest Required

Rerouting Calling Search Space < None >

SUBSCRIBE Calling Search Space < None >

SIP Profile\* Standard SIP Profile

Digest User 5090

Media Termination Point Required

Unattended Port

Require DTMF Reception

**MLPP Information**

MLPP Domain < None >

Save

\* - indicates required item.

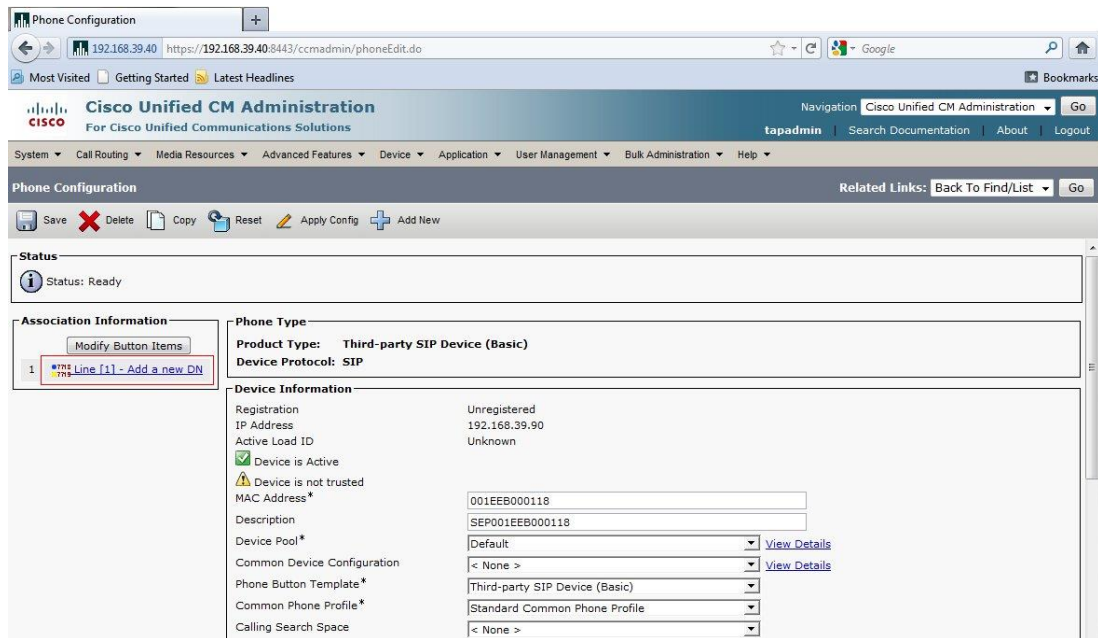
\*\* - Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.

\*\*\*Note: Security Profile Contains Addition CAPF Settings.

\*\*\*\*Note: A Protected device means it is capable of playing Secure and Non-Secure Tones. When the checkbox is checked, the user will hear a Secure or Non-Secure Tone when the call is connected.

\*\*\*\*\*Note: A custom Softkey template without supplementary service Softkeys must be used for a Hot line Device.

12. Each device needs to have a unique Directory Number. Click **Line [1] – Add a new DN**.

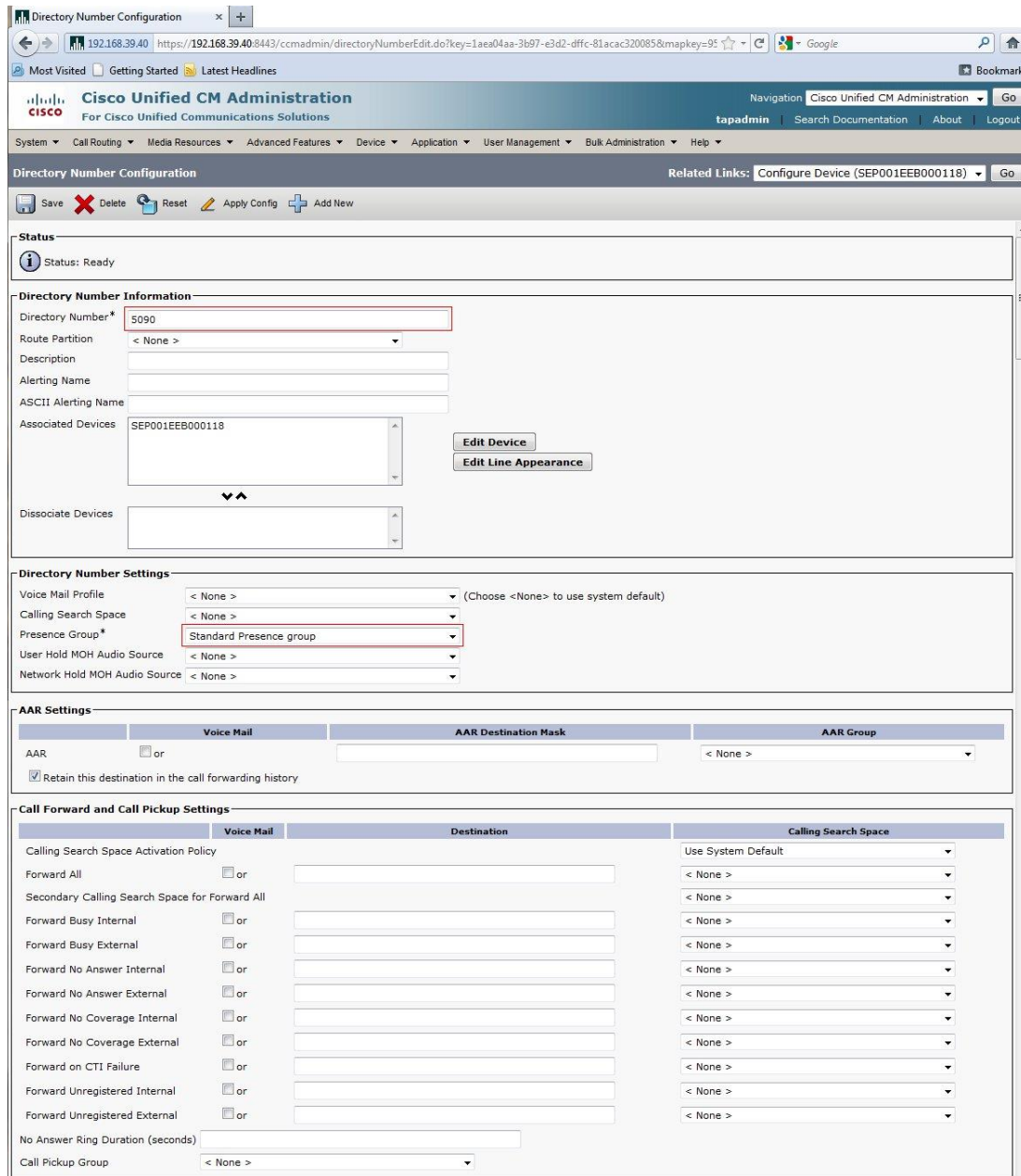


The screenshot shows the Cisco Unified CM Administration interface for a phone configuration. The page title is "Phone Configuration" and the URL is "https://192.168.39.40/ccmadmin/phoneEdit.do". The interface includes a navigation menu with options like System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, and Bulk Administration. The main content area is titled "Phone Configuration" and includes a "Status" section showing "Status: Ready". Below this is the "Association Information" section, which contains a table with one row: "1" and a button labeled "Line [1] - Add a new DN". To the right of this section is the "Phone Type" section, which shows "Product Type: Third-party SIP Device (Basic)" and "Device Protocol: SIP". Below that is the "Device Information" section, which contains a table of configuration details:

Registration	Unregistered
IP Address	192.168.39.90
Active Load ID	Unknown
<input checked="" type="checkbox"/> Device is Active	
Device is not trusted	
MAC Address*	001EEB000118
Description	SEP001EEB000118
Device Pool*	Default <a href="#">View Details</a>
Common Device Configuration	< None > <a href="#">View Details</a>
Phone Button Template*	Third-party SIP Device (Basic)
Common Phone Profile*	Standard Common Phone Profile
Calling Search Space	< None >

**13. On the Directory Number Configuration page, enter the following fields. Then click **Save**.**

Directory Number: Enter a Unique extension for the phone  
 Presence Group: **Standard Presence Group**  
 Maximum Number of Calls: **1**  
 Busy Trigger: **1**



The screenshot displays the 'Directory Number Configuration' page in the Cisco Unified CM Administration interface. The browser address bar shows the URL: https://192.168.39.40:8443/ccmadmin/directoryNumberEdit.do?key=1aa04aa-3b97-e3d2-dffc-81acac320085&mapkey=95. The page title is 'Directory Number Configuration'. The 'Status' section indicates 'Ready'. The 'Directory Number Information' section includes fields for 'Directory Number\*' (5090), 'Route Partition' (< None >), 'Description', 'Alerting Name', 'ASCII Alerting Name', 'Associated Devices' (SEP001EEB000118), and 'Dissociate Devices'. The 'Directory Number Settings' section includes 'Voice Mail Profile' (< None >), 'Calling Search Space' (< None >), 'Presence Group\*' (Standard Presence group), 'User Hold MOH Audio Source' (< None >), and 'Network Hold MOH Audio Source' (< None >). The 'AAR Settings' section includes 'Voice Mail' (checked), 'AAR Destination Mask', 'AAR Group' (< None >), and a checked option 'Retain this destination in the call forwarding history'. The 'Call Forward and Call Pickup Settings' section includes 'Calling Search Space Activation Policy', 'Forward All', 'Secondary Calling Search Space for Forward All', 'Forward Busy Internal', 'Forward Busy External', 'Forward No Answer Internal', 'Forward No Answer External', 'Forward No Coverage Internal', 'Forward No Coverage External', 'Forward on CTI Failure', 'Forward Unregistered Internal', 'Forward Unregistered External', 'No Answer Ring Duration (seconds)', and 'Call Pickup Group' (< None >).

(Continued on next page)

Park Monitoring			
	Voice Mail	Destination	Calling Search Space
Park Monitoring Forward No Retrieve Destination External	<input type="checkbox"/> or	<input type="text"/>	< None > A blank value means to call the parker's line.
Park Monitoring Forward No Retrieve Destination Internal	<input type="checkbox"/> or	<input type="text"/>	< None > A blank value means to call the parker's line.
Park Monitoring Reversion Timer	<input type="text"/>	A blank value will use value set in Park Monitoring Reversion Timer service parameter	
MLPP Alternate Party Settings			
Target (Destination)	<input type="text"/>		
MLPP Calling Search Space	< None >		
MLPP No Answer Ring Duration (seconds)	<input type="text"/>		
Line Settings for All Devices			
Hold Reversion Ring Duration (seconds)	<input type="text"/>	Setting the Hold Reversion Ring Duration to zero will disable the feature	
Hold Reversion Notification Interval (seconds)	<input type="text"/>	Setting the Hold Reversion Notification Interval to zero will disable the feature	
Party Entrance Tone*	Default		
Line 1 on Device SEP001EEB000118			
Display (Internal Caller ID)	<input type="text"/> Display text for a line appearance is intended for displaying text such as a name instead of a directory number for internal calls. If you specify a number, the person receiving a call may not see the proper identity of the caller.		
ASCII Display (Internal Caller ID)	<input type="text"/>		
External Phone Number Mask	<input type="text"/>		
Monitoring Calling Search Space	< None >		
Multiple Call/Call Waiting Settings on Device SEP001EEB000118			
Note: The range to select the Max Number of calls is: 1-2			
Maximum Number of Calls*	<input type="text" value="1"/>		
Busy Trigger*	<input type="text" value="1"/> (Less than or equal to Max. Calls)		
Forwarded Call Information Display on Device SEP001EEB000118			
<input checked="" type="checkbox"/> Caller Name <input type="checkbox"/> Caller Number <input type="checkbox"/> Redirected Number <input checked="" type="checkbox"/> Dialed Number			
Users Associated with Line			
<input type="button" value="Associate End Users"/>			
<input type="button" value="Save"/> <input type="button" value="Delete"/> <input type="button" value="Reset"/> <input type="button" value="Apply Config"/> <input type="button" value="Add New"/>			
<p><b>i</b> *- indicates required item.</p> <p><b>i</b> **- Changes to Line or Directory Number settings require restart.</p>			

14. If adding multiple VOIP-600 Series Phones, repeat Steps 6-14 for each device.



#### IV. VOIP-600 Series Phone Configuration

- Using a web browser, enter the IP address of the **VOIP-600 Series Phone** that you are programming. Login to the device with the configured Username and Password.
- In the VOIP-600 main menu, select **Network > SIP Settings**.
- Enter the following fields on the **SIP Settings** page. Then click **Apply**.  
**Assign a phone number:**  
 Phone Number: Enter the DN created in Step III-14

**Specify domain name:**

Domain Name: Enter the IP address of the Cisco UCM Server

**Enable/disable SIP registration:**

Register: Checked

**Specify SIP registrar:**

Username: Enter the User ID(same as the DN) create in Step III-8

Password: Enter the Digest Credentials created in Step III-8

IP Address: Enter the IP address of the CUCM Server

Port: (default: 5060)

Re-registration Time: (default: 3600)

**Specify outbound proxy:**

Username: Enter the User ID(same as the DN) create in Step III-8

Password: Enter the Digest Credentials created in Step III-8

IP Address: Enter the IP address of the CUCM Server

Port: (default: 5060)

- Repeat steps 1-4 for any additional **VOIP-600 Series Phones**.