

I. Introduction

This CUCM 8.0.3a Integration Guide provides general instructions for integration of the **VOIP-500 Series Phone** with a Cisco Call Manager installation. It is recommended to read this instruction set completely before starting any installation. For detailed VOIP-500 Series Phone setup instructions, please consult the **VOIP-500 Series Phone Manual**.

Talk-A-Phone's VOIP-500 Series Phone has tested compatible with Cisco UCM 7.1 and UCM 8.6. The Cisco Compatible logo signifies that Talk-A-Phone's product has undergone interoperability testing by Talk-A-Phone together with Cisco and a third-party test house based on testing criteria set by Cisco. Talk-A-Phone is solely responsible for the support and warranty of its product. Cisco makes no warranties, express or implied, with respect to Talk-A-Phone's product or its interoperation with the listed Cisco product(s) and disclaims any implied warranties of merchantability, fitness for a particular use, or against infringement.



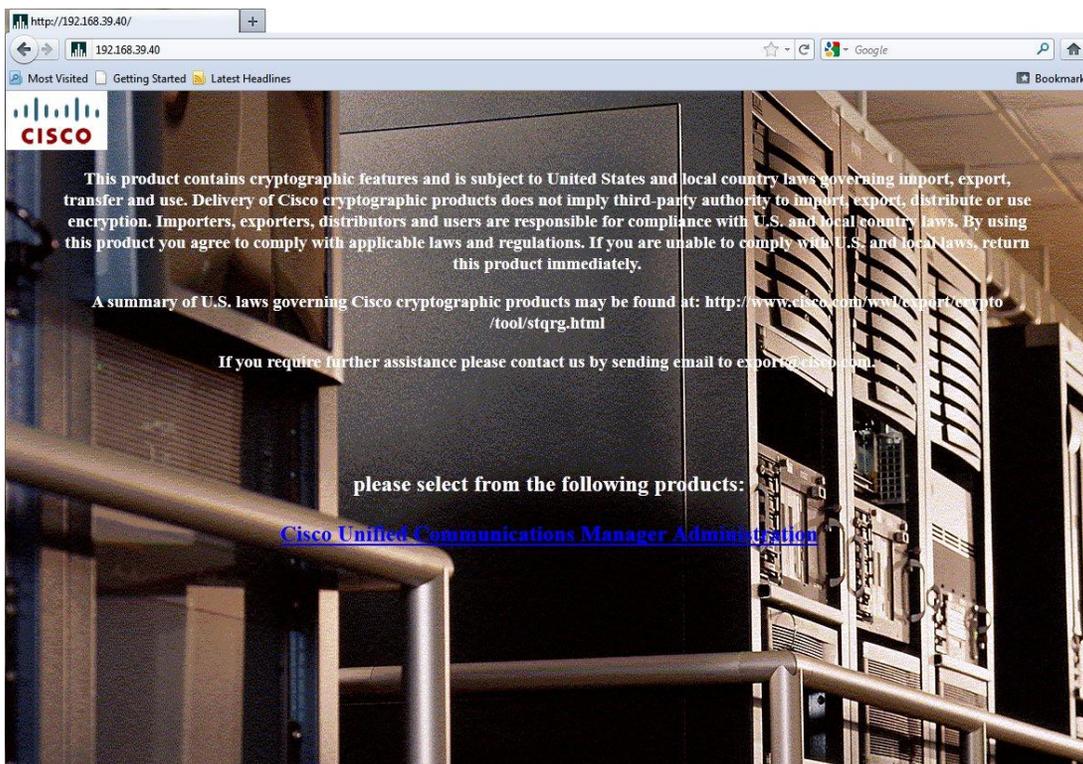
II. Prerequisites

- 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-500 Series Phones** and all network services (SIP, TFTP, HTTP, FTP, DNS, RTP/SRTP)

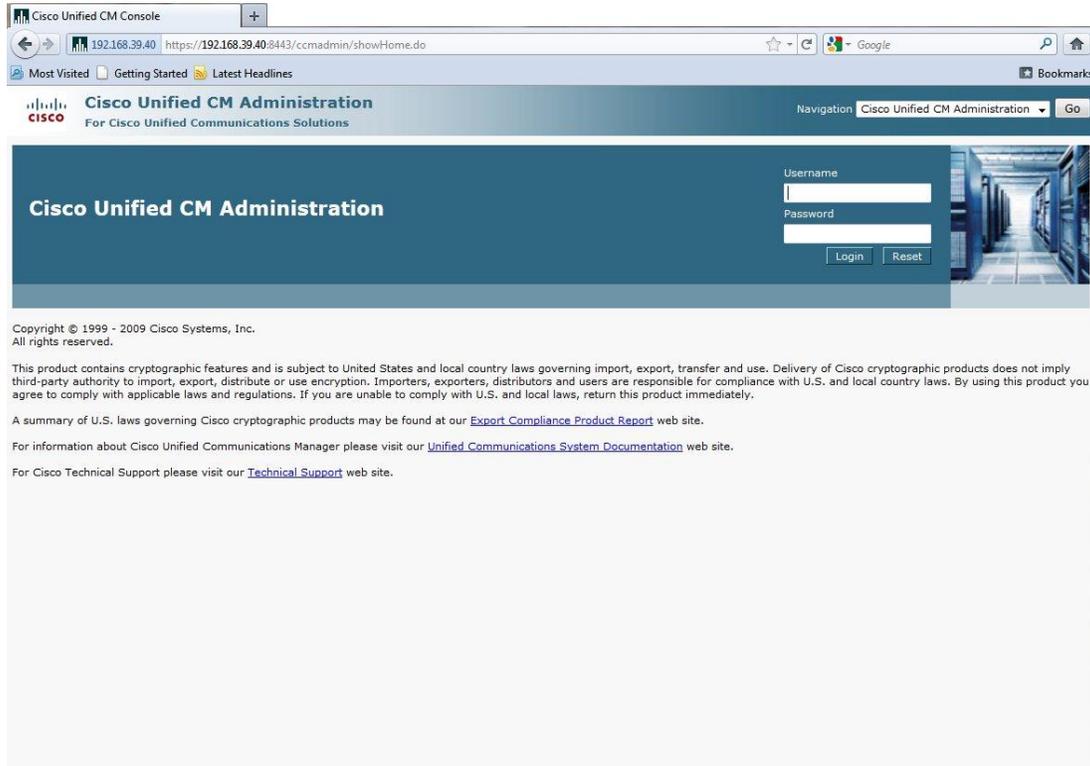
III. CUCM Basic Configuration

Basic instructions for integrating a **VOIP-500 Series Phone** with a Cisco Unified Call Manager 8.0.3a are included. Advanced setup of CUCM features is outside the scope of this document.

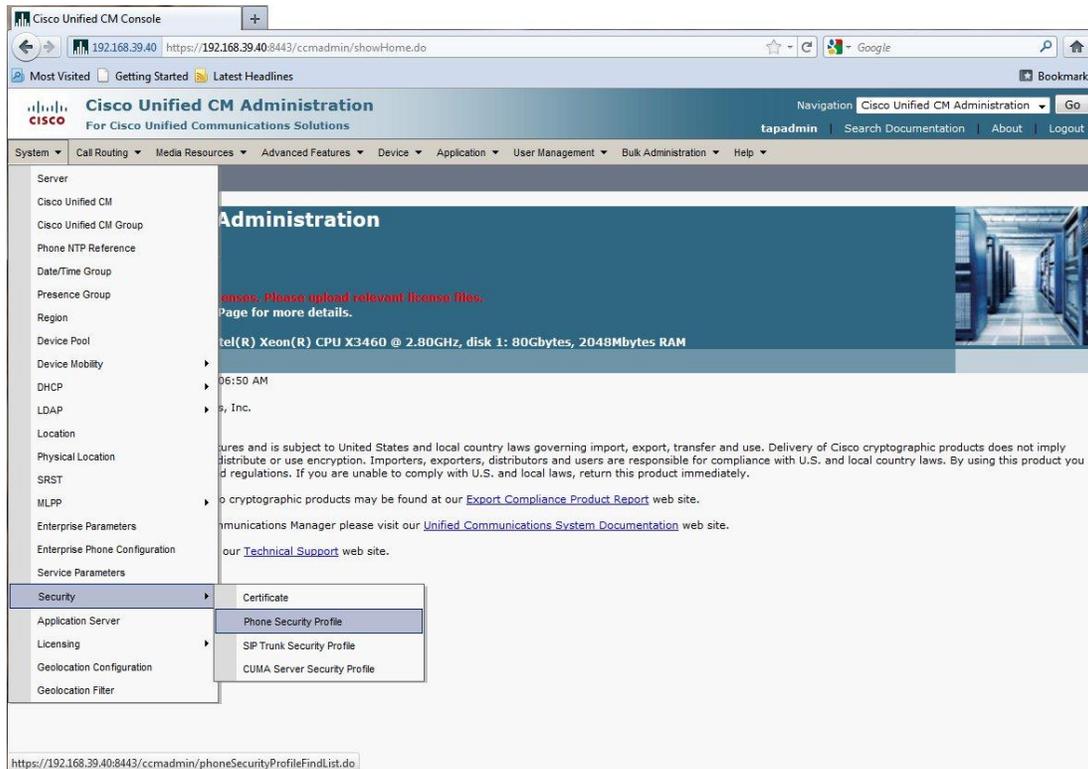
1. Using a web browser, enter the IP address (or FQDN if configured) of the CUCM Server in the address bar:



2. Login to Cisco Unified CM Administration:



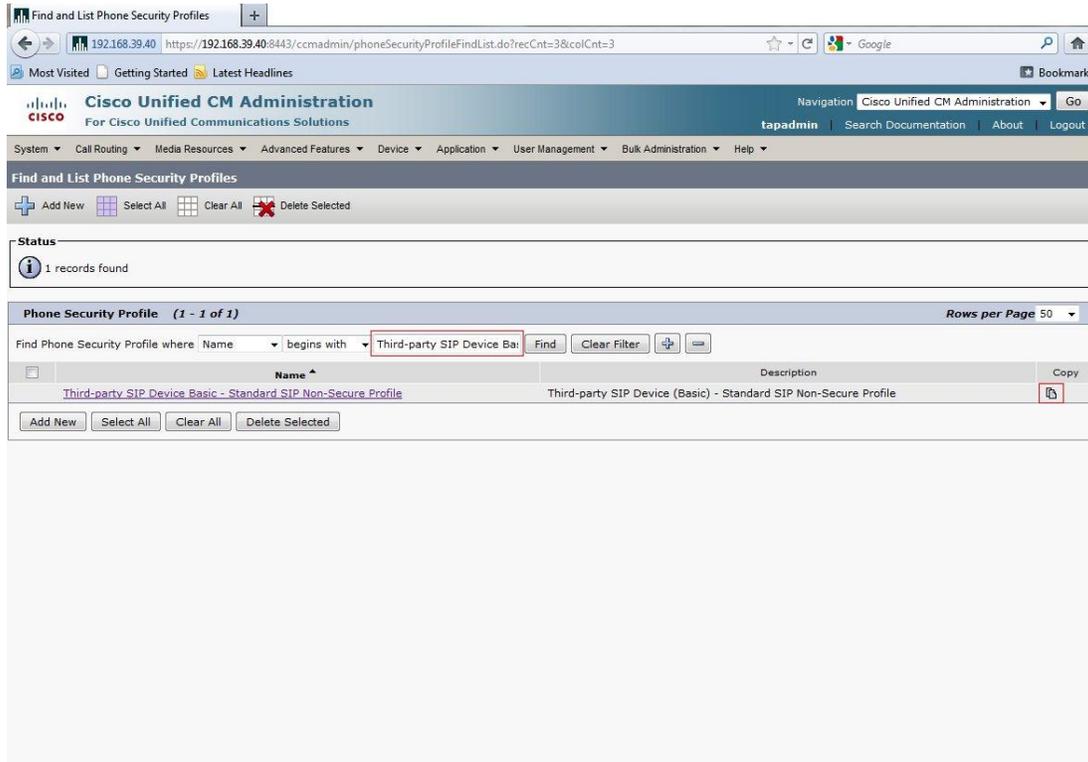
3. A Phone Security Profile with Digest Authentication enabled is required for VOIP-500 integration. Begin by selecting **System > Security > Phone Security Profile** in the CUCM menu:



- 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.



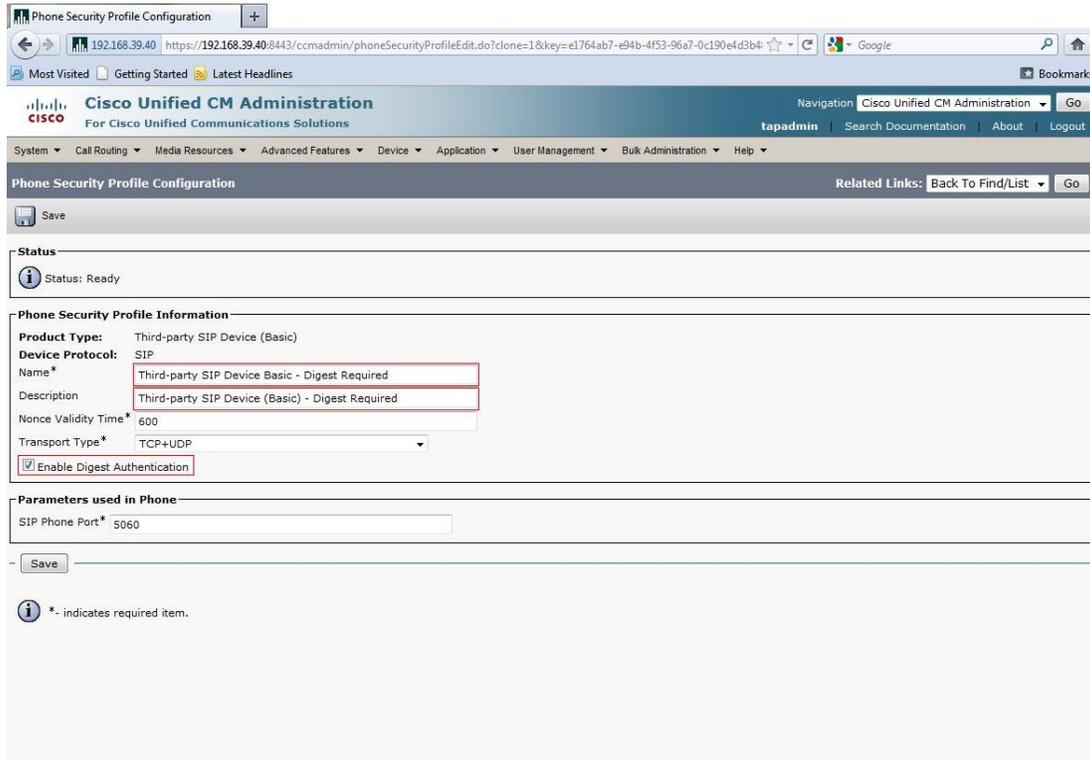
The screenshot shows the Cisco Unified CM Administration interface. The page title is "Find and List Phone Security Profiles". The search criteria is "Third-party SIP Device Basic". The search results table shows one entry:

Name	Description	Copy
Third-party SIP Device Basic - Standard SIP Non-Secure Profile	Third-party SIP Device (Basic) - Standard SIP Non-Secure Profile	

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**.



The screenshot shows the Cisco Unified CM Administration interface for configuring a Phone Security Profile. The browser address bar shows the URL: <https://192.168.39.40:8443/ccmadmin/phoneSecurityProfileEdit.do?clone=1&key=e1764ab7-e94b-4f53-96a7-0c190e4d3b4>. The page title is "Phone Security Profile Configuration".

Status: Ready

Phone Security Profile Information

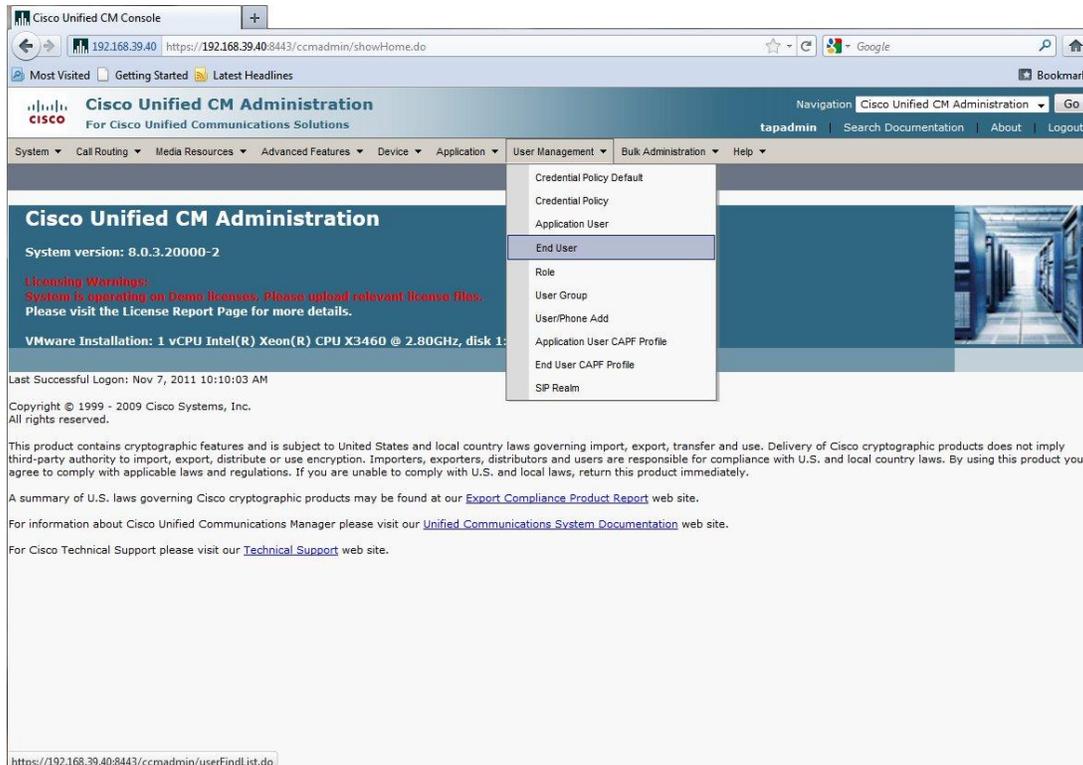
Product Type:	Third-party SIP Device (Basic)
Device Protocol:	SIP
Name*	Third-party SIP Device Basic - Digest Required
Description	Third-party SIP Device (Basic) - Digest Required
Nonce Validity Time*	600
Transport Type*	TCP+UDP
<input checked="" type="checkbox"/> Enable Digest Authentication	

Parameters used in Phone

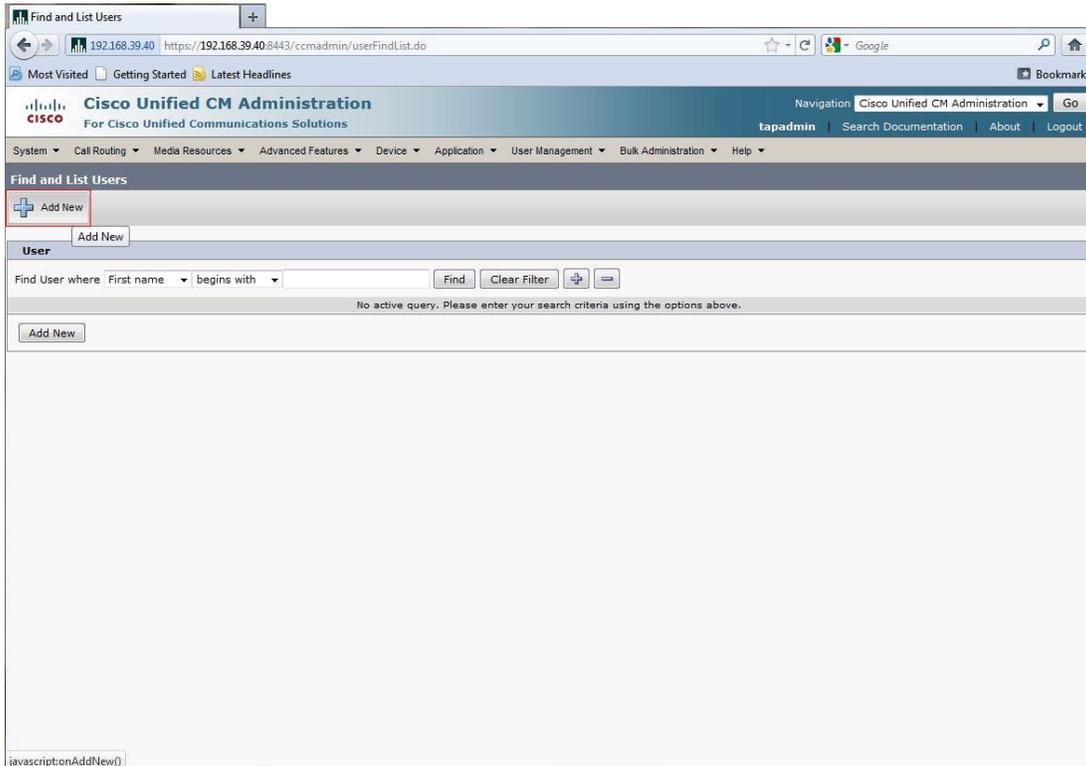
SIP Phone Port*	5060
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***** - indicates required item.

- Each VOIP-500 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**.



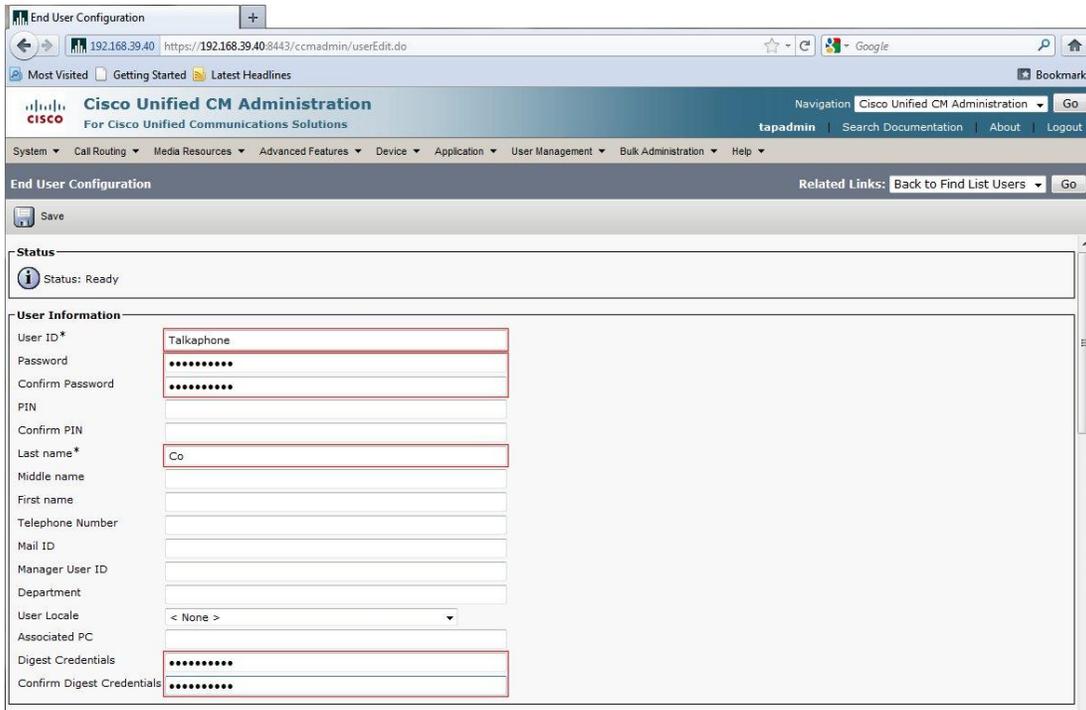
8. Enter the required fields to create a new End User:

User ID: A unique username for each **VOIP-500 Series Phone**.

Password: A unique password for each **VOIP-500 Series Phone**.

Last name: (required by CUCM).

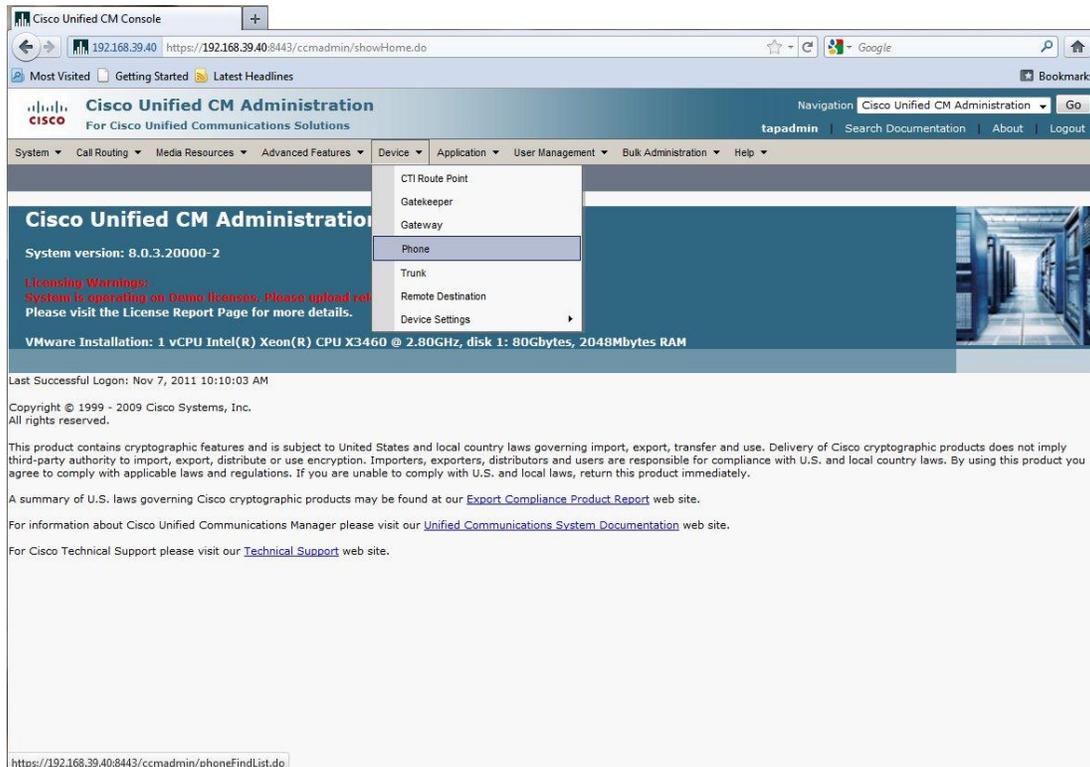
Digest Credentials: (required by **VOIP-500 Series Phone**) It is recommended to match the Digest Credentials with the Password for manageability.



The screenshot shows the 'End User Configuration' page in the Cisco Unified CM Administration interface. 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'. Below the navigation is a 'Save' button and a 'Status' section indicating 'Ready'. The main 'User Information' section contains the following fields:

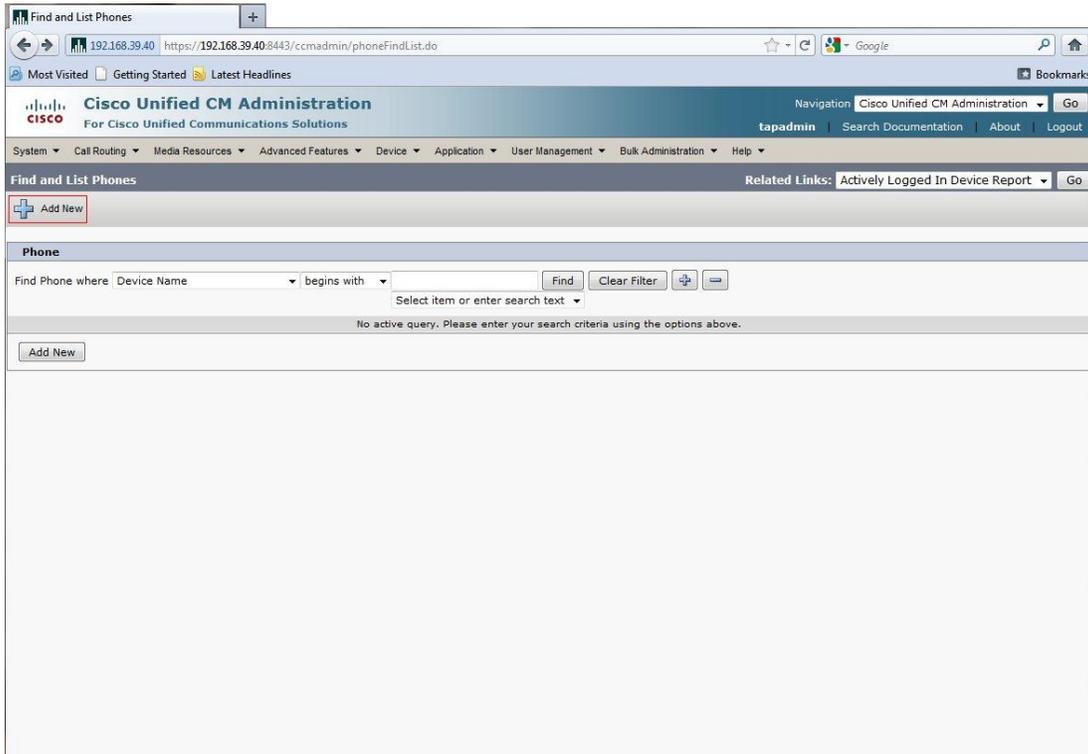
- User ID*: Talkaphone
- Password: [Masked]
- Confirm Password: [Masked]
- PIN: [Empty]
- Confirm PIN: [Empty]
- Last name*: Co
- Middle name: [Empty]
- First name: [Empty]
- Telephone Number: [Empty]
- Mail ID: [Empty]
- Manager User ID: [Empty]
- Department: [Empty]
- User Locale: < None >
- Associated PC: [Empty]
- Digest Credentials: [Masked]
- Confirm Digest Credentials: [Masked]

Next, add each **VOIP-500 Series Phone** as a Device in CUCM. In the CUCM main menu, select **Device > Phone**.

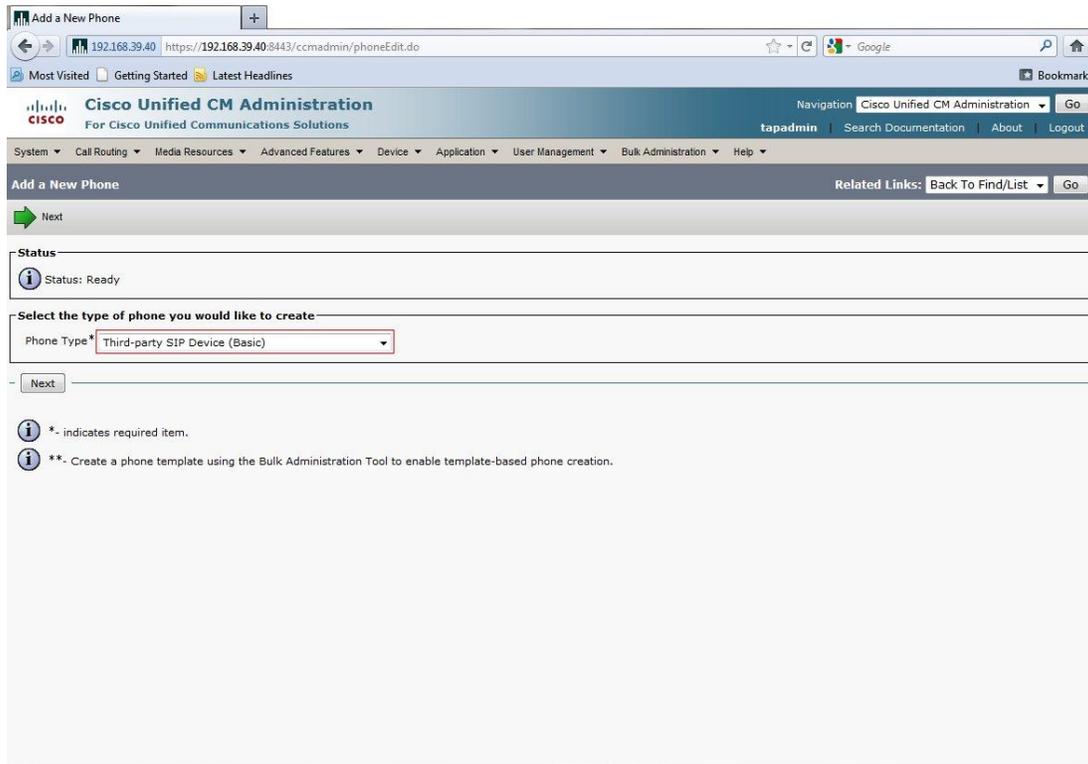


The screenshot shows the Cisco Unified CM Administration web interface. The browser address bar displays the URL `https://192.168.39.40/ccmadmin/showHome.do`. The page title is "Cisco Unified CM Administration" and the system version is "8.0.3.20000-2". A navigation menu is visible at the top, with the "Device" menu expanded to show options: CTI Route Point, Gatekeeper, Gateway, Phone (highlighted), Trunk, Remote Destination, and Device Settings. The page also displays a "License Warning" and system information: "VMware Installation: 1 vCPU Intel(R) Xeon(R) CPU X3460 @ 2.80GHz, disk 1: 80Gbytes, 2048Mbytes RAM".

9. Click **Add New**.



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



The screenshot shows the 'Add a New Phone' page in the Cisco Unified CM Administration interface. The browser address bar shows the URL: <https://192.168.39.40:8443/ccmadmin/phoneEdit.do>. The page title is 'Add a New Phone'. The navigation menu includes: System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The 'Add a New Phone' section has a 'Next' button and a 'Status' indicator showing 'Status: Ready'. Below this, the 'Select the type of phone you would like to create' section has a 'Phone Type*' dropdown menu set to 'Third-party SIP Device (Basic)'. A 'Next' button is located below the dropdown. At the bottom, there are two informational messages: '*- indicates required item.' and '**- Create a phone template using the Bulk Administration Tool to enable template-based phone creation.'

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

MAC Address: H/W address of VOIP-500

Description: (Auto-filled)

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

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

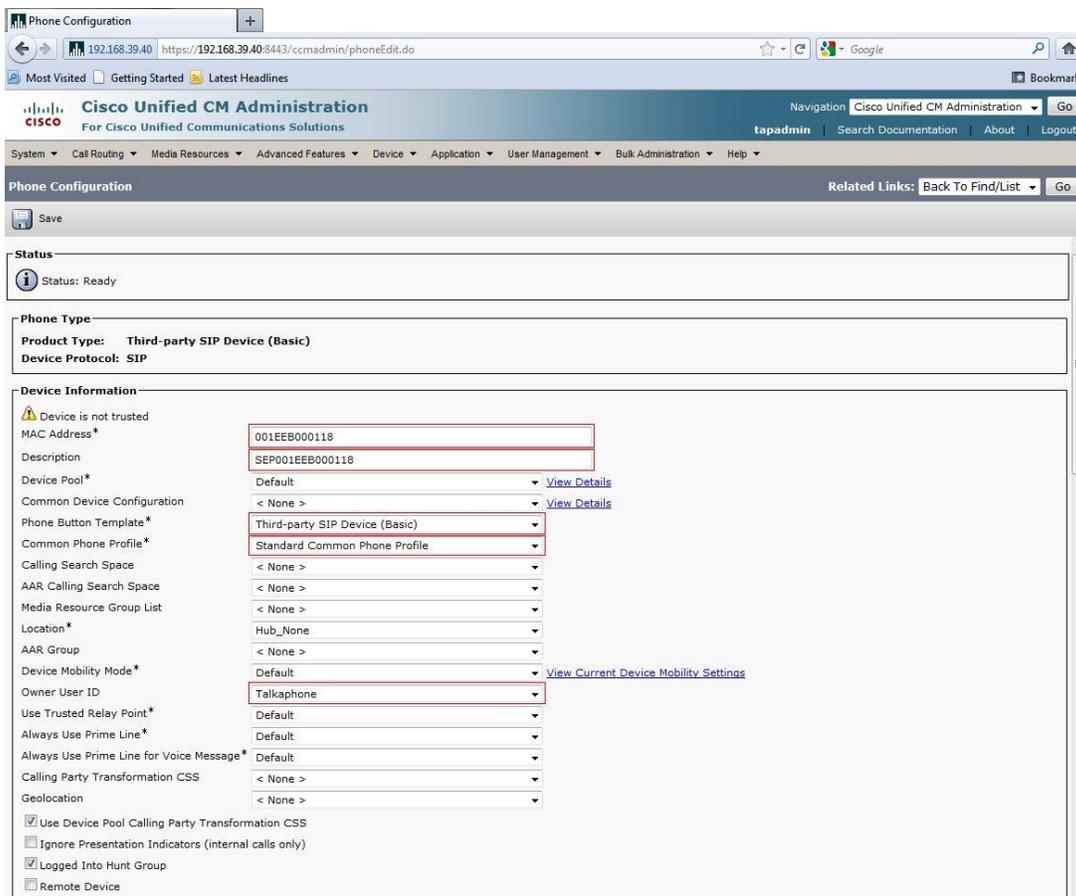
Owner User ID: Enter the User ID created in Step 8

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

Navigation Cisco Unified CM Administration Go

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 Talkphone

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

(Continued on next page)

Protocol Specific Information

Presence Group*

MTP Preferred Originating Codec*

Device Security Profile*

Rerouting Calling Search Space

SUBSCRIBE Calling Search Space

SIP Profile*

Digest User

Media Termination Point Required

Unattended Port

Require DTMF Reception

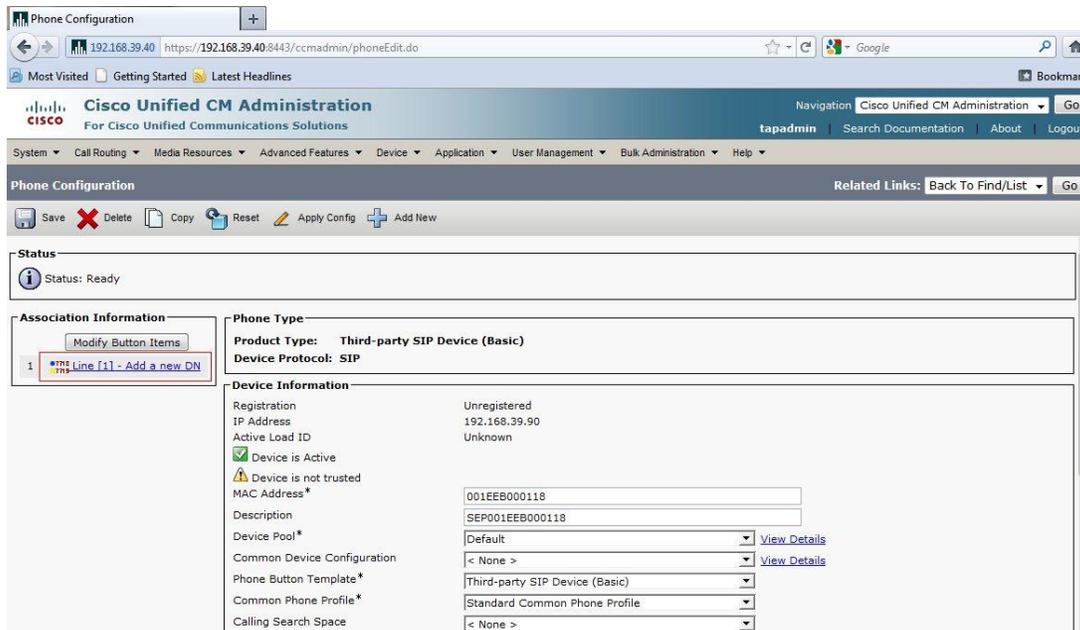
MLPP Information

MLPP Domain

Notes:

- * - 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 web interface. The main content area is titled "Phone Configuration" and shows the configuration for a "Third-party SIP Device (Basic)".

Association Information: A table with one entry: "1 Line [1] - Add a new DN".

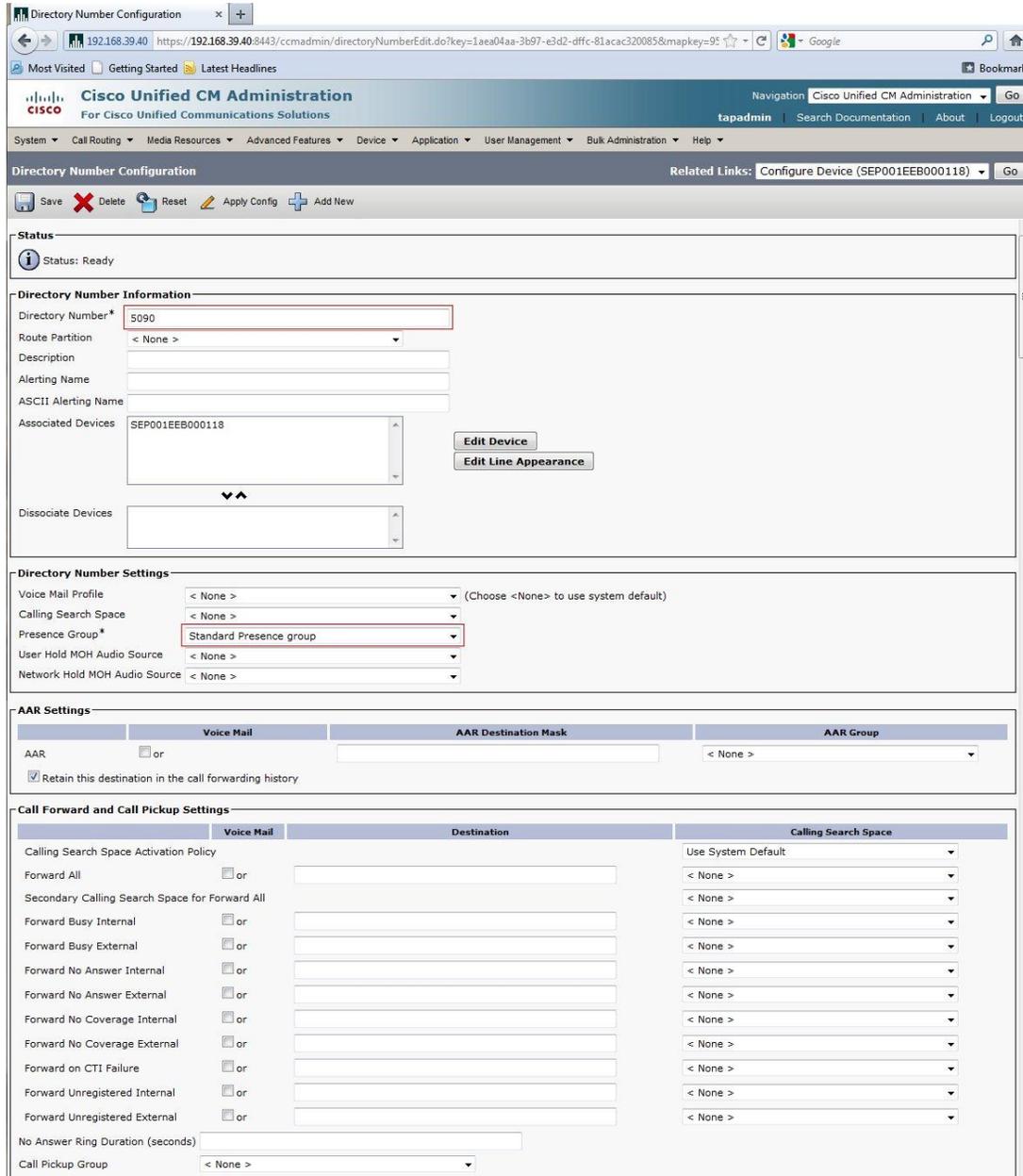
Phone Type: Product Type: Third-party SIP Device (Basic); Device Protocol: SIP.

Device Information:

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 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 >

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 shows the 'Directory Number Configuration' page in the Cisco Unified CM Administration interface. The page is titled 'Directory Number Configuration' and includes a navigation menu at the top. The main content area is divided into several sections:

- Directory Number Information:** This section contains fields for 'Directory Number*' (set to 5090), 'Route Partition' (set to '<None>'), 'Description', 'Alerting Name', 'ASCII Alerting Name', and 'Associated Devices' (set to SEP001EEB000118). There are also buttons for 'Edit Device' and 'Edit Line Appearance'.
- Directory Number Settings:** This section contains dropdown menus for 'Voice Mail Profile' (set to '<None>'), 'Calling Search Space' (set to '<None>'), 'Presence Group*' (set to 'Standard Presence group'), 'User Hold MOH Audio Source' (set to '<None>'), and 'Network Hold MOH Audio Source' (set to '<None>').
- AAR Settings:** This section contains a table with columns for 'Voice Mail', 'AAR Destination Mask', and 'AAR Group'. The 'Voice Mail' checkbox is checked, and the 'AAR Group' dropdown is set to '<None>'. There is also a checkbox for 'Retain this destination in the call forwarding history' which is checked.
- Call Forward and Call Pickup Settings:** This section contains a table with columns for 'Voice Mail', 'Destination', and 'Calling Search Space'. It includes various forwarding options such as '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', and 'Forward Unregistered External'. The 'Calling Search Space' dropdown is set to 'Use System Default'.

(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>i * - indicates required item.</p> <p>i ** - Changes to Line or Directory Number settings require restart.</p>			

14. If adding multiple **VOIP-500 Series Phones**, repeat Steps 6-14 for each device.

IV. VOIP-500 Series Phone Configuration

1. Using a web browser, enter the IP address of the **VOIP-500 Series Phone** that you are programming. Login to the device with the configured Username and Password.
2. In the VOIP-500 main menu, select **Network > SIP Settings**.
3. 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 CUCM Server

Enable/disable SIP registration:

Register: Checked

Specify SIP registrar:

Username: Enter the User ID 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)

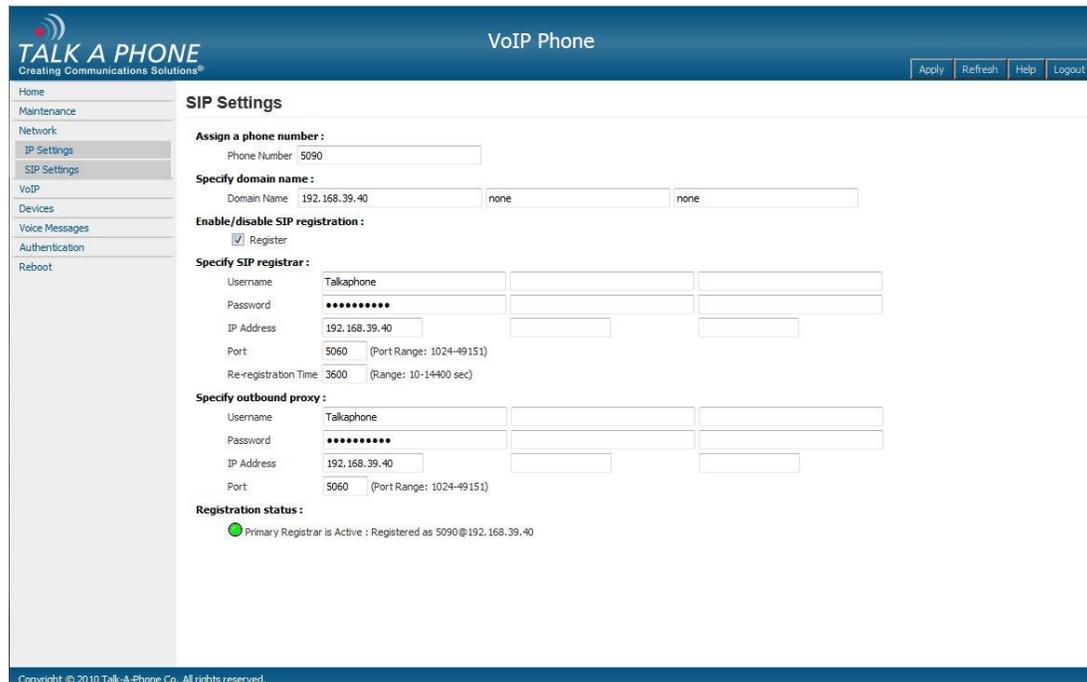
Specify outbound proxy:

Username: Enter the User ID 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)



The screenshot shows the 'SIP Settings' page for a VOIP Phone. The page has a blue header with the 'TALK A PHONE' logo and 'VOIP Phone' text. A navigation menu on the left includes Home, Maintenance, Network, IP Settings, SIP Settings (selected), VoIP, Devices, Voice Messages, Authentication, and Reboot. The main content area contains several sections: 'Assign a phone number' with a text field for 'Phone Number' (5090); 'Specify domain name' with a 'Domain Name' field (192.168.39.40) and two dropdown menus (none); 'Enable/disable SIP registration' with a checked 'Register' checkbox; 'Specify SIP registrar' with fields for Username (Talkaphone), Password (masked), IP Address (192.168.39.40), Port (5060), and Re-registration Time (3600); 'Specify outbound proxy' with fields for Username (Talkaphone), Password (masked), IP Address (192.168.39.40), and Port (5060); and 'Registration status' showing a green dot and the text 'Primary Registrar is Active : Registered as 5090@192.168.39.40'. At the bottom, there is a copyright notice: 'Copyright © 2010 Talk-A-Phone Co. All rights reserved.'

4. Repeat steps 1-4 for any additional **VOIP-500 Series Phones**.