

# Configuring Cisco Unified Communications Manager with Comrex STAC-VIP



Vortex Communications Ltd, 75 The Grove, Ealing, London W5 5LL, UK

Tel: +44 (0)20-8579 2743 \* Fax: +44 (0)20-8840 0018

Email: [info@vtx.co.uk](mailto:info@vtx.co.uk) \* Web: [www.vtx.co.uk](http://www.vtx.co.uk)

Copyright © Vortex Communications Ltd 2015

## CONTENTS

Introduction .....	3
Preliminary Steps .....	4
Enable All Services on CUCM .....	4
Connecting STAC-VIP to CUCM .....	6
Add a Phone Security Profile for STAC-VIP .....	6
Add an End User for STAC-VIP .....	6
Add a Phone for STAC-VIP .....	7
Add a Directory Number (DN) for STAC-VIP .....	10
Register STAC-VIP to CUCM .....	11
Connecting a Softphone to CUCM .....	14
Add a Phone Security Profile for the Softphone .....	14
Add an End User for the Softphone .....	15
Add a Phone for the Softphone .....	16
Add a Directory Number (DN) for the Softphone .....	18
Register the Softphone to CUCM .....	18
Notes .....	21
References .....	23

## **INTRODUCTION**

This guide assumes you are starting with a clean install of Cisco UCM (refer to [1], [2] or [3] for installation details) and a STAC-VIP in its factory-default state. STAC-VIP should be running firmware version 1.1p4 or newer and both CUCM and STAC-VIP Mainframe should have static IP addresses assigned. In this guide we set up a basic lab test scenario which allows calls to be made from hardware IP phones or softphones to STAC-VIP via CUCM. CUCM is a very flexible and complex product; you are encouraged to refer to [4], [5] or [6] and / or your Cisco support contact for assistance in developing the configuration which best suits your requirements.

The officially-supported browsers and operating systems for the CUCM 10 web control panel are as follows:

Internet Explorer 8, Internet Explorer 9 (Windows XP, Windows Vista, Windows 7)

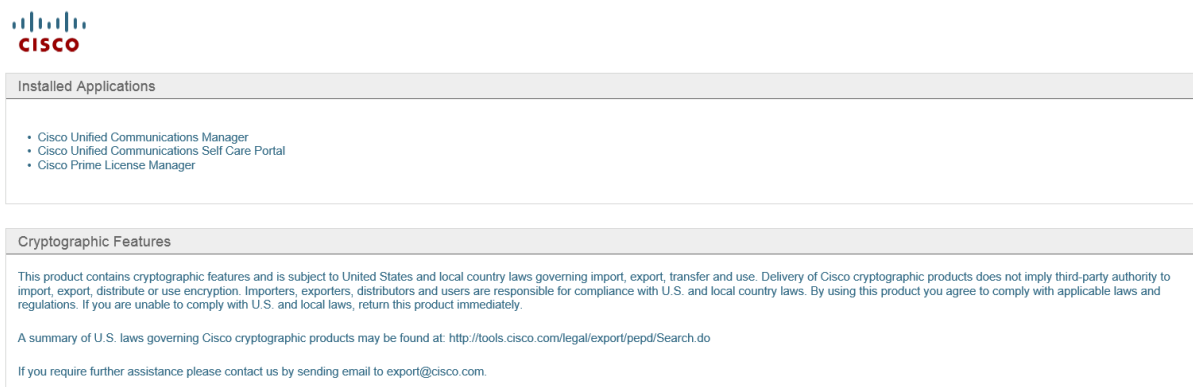
Firefox 4.x, Firefox 10.x (Windows XP, Windows Vista, Windows 7, OSX)

Safari 5.x (OSX)

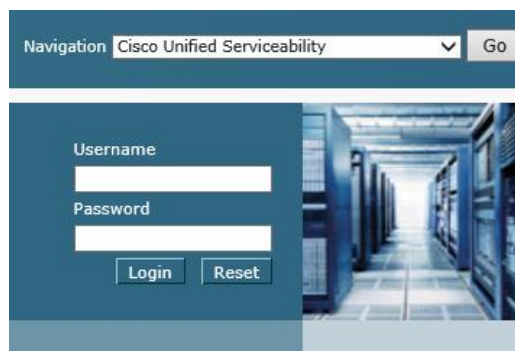
## PRELIMINARY STEPS

### Enable All Services on CUCM

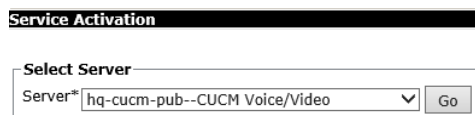
- a) Browse to the IP address of your CUCM installation and click on Cisco Unified Communications Manager; ignore any errors caused by the server's certificate not being trusted by your browser (in Internet Explorer, click 'Continue to this website (not recommended)').



- b) In the Navigation drop-down at the top-right of the screen, select Cisco Unified Serviceability and click Go. Then enter the credentials set up when installing the server in the Username and Password fields and click Login.



- c) In the top menu, click Tools, Service Activation. If asked to select a server, choose the CUCM server you wish to configure from the drop-down menu and click Go.



- d) Check the Check All Services box, click Save and OK in the warning dialog. Wait a few minutes for the 'Loading, please wait.' message to disappear and the page to refresh, at which point all services should show an Activation Status of Activated.

**Service Activation**

Save Set to Default Refresh

**Status:**  
Ready

**Select Server**

Server\* hq-cucm-pub--CUCM Voice/Video Go

Check All Services

- e) In the Navigation drop-down at the top-right of the screen, select Cisco Unified CM Administration and click Go.

# CONNECTING STAC-VIP TO CUCM

## Add a Phone Security Profile for STAC-VIP

- a) In the top menu, click System, Security, Phone Security Profile. In the 'Find Phone Security Profile where' filter, select Name, begins with, enter 'Third-party' (without quotes) in the empty field and click Find.

The screenshot shows the Cisco Unified CM Administration interface. The top navigation bar includes 'System', 'Call Routing', 'Media Resources', 'Advanced Features', 'Device', 'Application', 'User Management', 'Bulk Administration', and 'Help'. The main content area is titled 'Find and List Phone Security Profiles'. It features a search bar with 'Name' selected and 'begins with' 'Third-party' entered. Below the search bar, a table lists three profiles. The second profile, 'Third-party SIP Device Advanced - Standard SIP Non-Secure Profile', has a 'Copy' icon next to it.

- b) Click the Copy icon next to Third-party SIP Device Advanced - Standard SIP Non-Secure Profile.  
c) Enter the following information on the Phone Security Profile Configuration screen and click Save. You should see Status: Add successful.

**Name:** Third-party SIP Device Advanced - Digest Authentication

**Description:** Third-party SIP Device (Advanced) - Digest Authentication

**Enable Digest Authentication:** Checked

The screenshot shows the 'Phone Security Profile Configuration' screen. The status is 'Ready'. The configuration fields are: Product Type: Third-party SIP Device (Advanced), Device Protocol: SIP, Name: hird-party SIP Device Advanced - Digest Authentication, Description: rd-party SIP Device (Advanced) - Digest Authentication, Nonce Validity Time: 600, Transport Type: TCP+UDP, and Enable Digest Authentication: checked. The SIP Phone Port is 5060.

## Add an End User for STAC-VIP

- a) In the top menu, click User Management, End User, then click Add New.

b) Enter the following information on the End User Configuration screen and click Save. You should see Status: Add successful.

**User ID:** stacvip

**Password:** P@ssw0rdOfYourChoice [or any other password of your choice]

**Confirm Password:** P@ssw0rdOfYourChoice [must match Password]

**Last name:** STAC-VIP

**Digest Credentials:** P@ssw0rdOfYourChoice [or any other password of your choice]

**Confirm Digest Credentials:** P@ssw0rdOfYourChoice [must match Digest Credentials]

## Add a Phone for STAC-VIP

a) In the top menu, click Device, Phone, then click Add New.

- b) In the Phone Type drop-down, select Third-party SIP Device (Advanced) and click Next.
- c) Enter the following information on the Phone Configuration screen and click Save, then OK in the dialog prompting you to apply the configuration. You should see Status: Add successful.

**MAC Address:** This field is actually ignored in our setup and can be any unique 12-digit hex number (0-9, A-F) without punctuation; the actual MAC address of your STAC-VIP mainframe can be determined by running the Comrex Device Manager application [7], clicking Scan for Devices and noting the entry in the MAC Address column alongside STAC-VIP Mainframe

**Description:** STAC-VIP

**Device Pool:** Default

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

**Owner User ID:** stacvip [CUCM 9.1 and above only]

**Device Security Profile:** Third-party SIP Device Advanced - Digest Authentication

**SIP Profile:** Standard SIP Profile

**Digest User:** stacvip



**Phone Configuration**

Save

**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="0123456789AB"/>	
Description	<input type="text" value="STAC-VIP"/>	
Device Pool*	<input type="text" value="Default"/>	<a href="#">View Details</a>
Common Device Configuration	<input type="text" value="&lt; None &gt;"/>	<a href="#">View Details</a>
Phone Button Template*	<input type="text" value="Third-party SIP Device (Advanced)"/>	
Common Phone Profile*	<input type="text" value="Standard Common Phone Profile"/>	<a href="#">View Details</a>
Calling Search Space	<input type="text" value="&lt; None &gt;"/>	
AAR Calling Search Space	<input type="text" value="&lt; None &gt;"/>	
Media Resource Group List	<input type="text" value="&lt; None &gt;"/>	
Location*	<input type="text" value="Hub_None"/>	
AAR Group	<input type="text" value="&lt; None &gt;"/>	
Device Mobility Mode*	<input type="text" value="Default"/>	<a href="#">View Current Device Mobility Settings</a>
Owner	<input checked="" type="radio"/> User <input type="radio"/> Anonymous (Public/Shared Space)	
Owner User ID*	<input type="text" value="stacvip"/>	
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="&lt; None &gt;"/>	

- Retry Video Call as Audio
- Ignore Presentation Indicators (internal calls only)
- Logged Into Hunt Group
- Remote Device

**Number Presentation Transformation**

**Caller ID For Calls From This Phone**

Calling Party Transformation CSS

Use Device Pool Calling Party Transformation CSS (Caller ID For Calls From This Phone)

**Remote Number**

Calling Party Transformation CSS

Use Device Pool Calling Party Transformation CSS (Device Mobility Related Information)

**Protocol Specific Information**

BLF Presence Group*	<input type="text" value="Standard Presence group"/>	
MTP Preferred Originating Codec*	<input type="text" value="711ulaw"/>	
Device Security Profile*	<input type="text" value="Third-party SIP Device Advanced - Digest Authenti"/>	
Rerouting Calling Search Space	<input type="text" value="&lt; None &gt;"/>	
SUBSCRIBE Calling Search Space	<input type="text" value="&lt; None &gt;"/>	
SIP Profile*	<input type="text" value="Standard SIP Profile"/>	<a href="#">View Details</a>
Digest User	<input type="text" value="stacvip"/>	

- Media Termination Point Required
- Unattended Port
- Require DTMF Reception
- Allow Presentation Sharing using BFCP
- Allow iX Applicable Media

**MLPP and Confidential Access Level Information**

MLPP Domain	<input type="text" value="&lt; None &gt;"/>
Confidential Access Mode	<input type="text" value="&lt; None &gt;"/>
Confidential Access Level	<input type="text" value="&lt; None &gt;"/>

Save

## Add a Directory Number (DN) for STAC-VIP

- a) In the Association Information panel on the left of the screen, click Line [1] - Add a new DN.



- b) Enter the following information on the Directory Number Configuration screen and click Save. You should see a new Associated Devices list appear with a single entry representing the STAC-VIP which you have just added.

**Directory Number:** 100 [or any other convenient unused extension number]

**Description:** STAC-VIP

**Maximum Number of Calls:** 16

**Busy Trigger:** 16

The screenshot shows the 'Directory Number Configuration' page in Cisco Unified CM Administration. The form contains the following fields and values:

- Directory Number\*: 100
- Route Partition: < None >
- Description: STAC-VIP
- Alerting Name: (empty)
- ASCII Alerting Name: (empty)
- External Call Control Profile: < None >
- Active
- Voice Mail Profile: < None > (Choose <None> to use system default)
- Calling Search Space: < None >
- BLF Presence Group\*: Standard Presence group
- User Hold MOH Audio Source: < None >
- Network Hold MOH Audio Source: < None >
- Reject Anonymous Calls

**Line 1 on Device SEP0123456789AB**

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 SEP0123456789AB**

Note: The range to select the Max Number of calls is: 1-16

Maximum Number of Calls\*

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

---

**Forwarded Call Information Display on Device SEP0123456789AB**

Caller Name

Caller Number

Redirected Number

Dialed Number

c) Finally, click Apply Config and OK in the popup window which appears.

**Register STAC-VIP to CUCM**

- a) It should now be possible to register your STAC-VIP to CUCM. Browse to the IP address of your STAC-VIP Mainframe, log in, click Configure and log in again. Click Line Configuration, VoIP Providers, Add Provider, SIP Provider.
- b) On the SIP Provider screen, click each of the following settings, enter the appropriate value and click Save Setting each time.

**Name:** CUCM  
**Account username:** 100 [must match Directory Number associated with STAC-VIP Phone on CUCM above]  
**Account password:** P@sswOrdOfYourChoice [must match Digest Credentials associated with STAC-VIP End User on CUCM above]  
**Server/Realm:** <IP address of CUCM installation>

- c) Click Show Advanced, click each of the following settings, enter the appropriate value and click Save Setting each time. **Note that the SIP Port setting is assigned automatically and should not be changed from its default value.**

**Auth Username:** stacvip  
**Codec Priority:** No ISAC or Opus [G.722 > G.711 > G.729]  
**INVITE SDP Compatibility:** On

# CUCM

[Apply Changes](#) [Delete](#)

SIP provider allowing STAC VIP to manage one or more VoIP calls simultaneously.

Initializing  
Changes to this provider must be applied to take effect.

## General Settings

Name	CUCM
Network Port Binding	Primary
Color Code	Blue
SIP Provider	Generic SIP Pr...

## Account Information

Account username	100
Account password	*****
Auth Username	stacvip

## Codec Settings

Codec Priority	No ISAC or Op...
Codec Fairness	Fair Negotiation

## SIP Settings

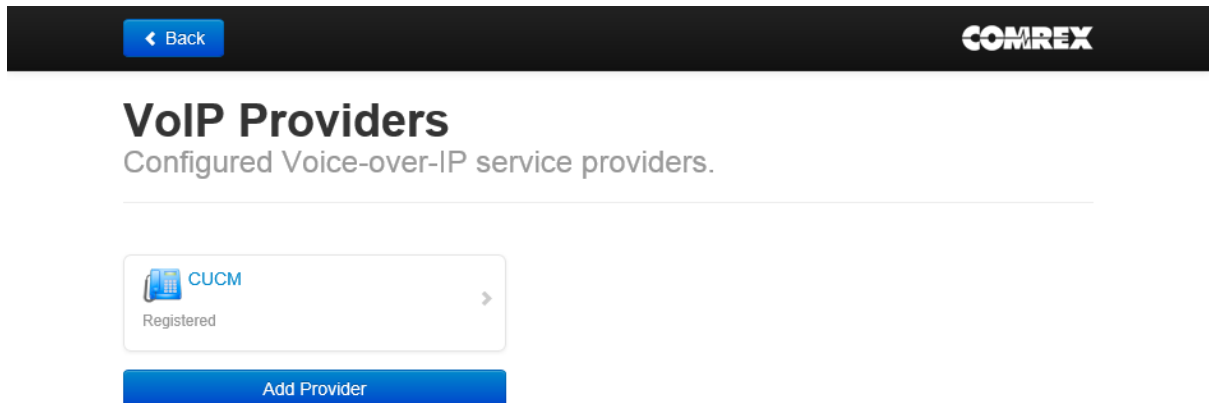
Outgoing Caller ID Name	
Outgoing Caller ID Number	
Provider Binding Port	5081
Server/Realm	10.10.20.1
Proxy Address	
Outbound Proxy Address	
Registration Proxy Address	
From Username	
From Domain	
Expire time	3600 seconds
Retry time	60 seconds
Register	Yes
Register Transport	UDP
INVITE SDP Compatibility	On
INVITE Contact Compatibility	Off

## Extra Settings

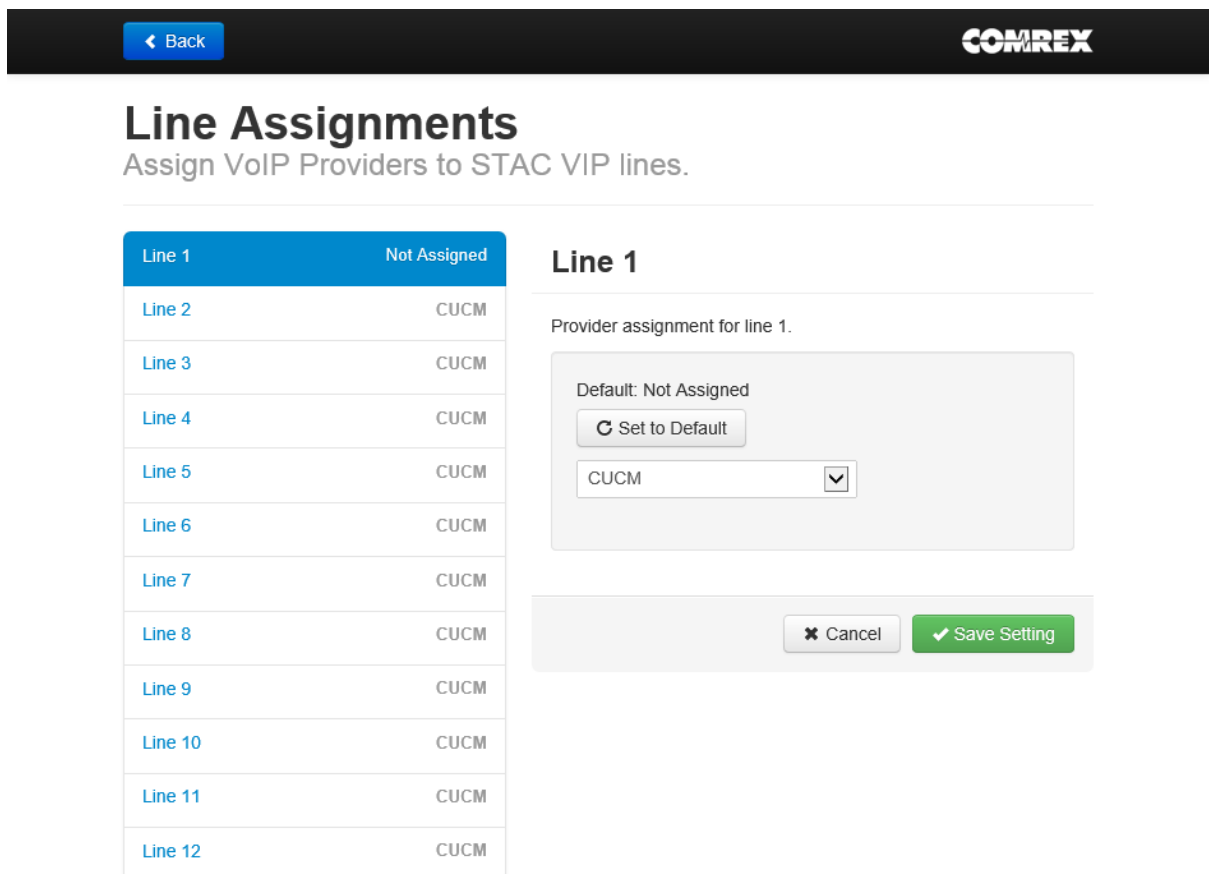
Outgoing Enabled	Yes
Transfer Enabled	No
Transfer Domain	
Destination Match	

[Hide Advanced](#)

- d) Click Apply Changes then Back, or Back, then Restart (Apply Changes) and OK in the confirmation dialog. You should see the status of the newly-added CUCM provider change to Registered.



- e) Click Back, then Line Assignments. Click each line which you wish to be associated with the CUCM Directory Number (100 in the example above), select CUCM from the Provider assignment drop-down and click Save Setting.



- f) The association of STAC-VIP to CUCM is now complete. If you already have software or hardware phones registered to CUCM, you should be able to call into STAC-VIP by dialling the Directory Number associated with it (100 in the example above). If not, continue with this guide to add a softphone account for testing.

# CONNECTING A SOFTPHONE TO CUCM

## Add a Phone Security Profile for the Softphone

- Return to the CUCM administration page by entering the IP address of the CUCM installation, clicking Cisco Unified Communications Manager, ensuring Cisco Unified CM Administration is selected in the Navigation drop-down at the top-right of the screen (if not, select it and click Go) and logging if necessary.
- In the top menu, click System, Security, Phone Security Profile. In the 'Find Phone Security Profile where' filter, select Name, begins with, enter 'Third-party' (without quotes) in the empty field and click Find.

The screenshot shows the Cisco Unified CM Administration interface. The top navigation bar includes 'Navigation' with 'Cisco Unified CM Administration' selected. The main menu shows 'System' > 'Security' > 'Phone Security Profile'. The 'Find and List Phone Security Profiles' page is active, displaying search filters for 'Name' and 'begins with' 'Third-party'. A table lists four profiles, with the last one, 'Third-party SIP Device Basic - Standard SIP Non-Secure Profile', highlighted. A 'Copy' icon is visible next to this profile.

- Click the Copy icon next to Third-party SIP Device Basic - Standard SIP Non-Secure Profile.
- Enter the following information on the Phone Security Profile Configuration screen and click Save. You should see Status: Add successful.

**Name:** Third-party SIP Device Basic - Digest Authentication

**Description:** Third-party SIP Device (Basic) - Digest Authentication

**Enable Digest Authentication:** Checked

The screenshot shows the 'Phone Security Profile Configuration' page. The status is 'Ready'. The 'Phone Security Profile Information' section contains the following fields: Product Type (Third-party SIP Device (Basic)), Device Protocol (SIP), Name (Third-party SIP Device Basic - Digest Authentication), Description (Third-party SIP Device (Basic) - Digest Authentication), Nonce Validity Time (600), and Transport Type (TCP+UDP). The 'Enable Digest Authentication' checkbox is checked. The 'Parameters used in Phone' section shows SIP Phone Port (5060).

## Add an End User for the Softphone

- a) In the top menu, click User Management, End User, then click Add New.

The screenshot shows the 'Find and List Users' interface in Cisco Unified CM Administration. At the top, there is a navigation bar with 'Cisco Unified CM Administration' and 'For Cisco Unified Communications Solutions'. Below this, a menu bar includes 'System', 'Call Routing', 'Media Resources', 'Advanced Features', 'Device', 'Application', 'User Management', 'Bulk Administration', and 'Help'. The main content area has a 'Find and List Users' header with an 'Add New' button. Below the header is a search section titled 'User' with a 'Find User where' dropdown set to 'First name', a 'begins with' dropdown, and a search input field. There are 'Find', 'Clear Filter', and navigation buttons. A message below the search area states: 'No active query. Please enter your search criteria using the options above.' At the bottom left, there is an 'Add New' button.

- b) Enter the following information on the End User Configuration screen and click Save. You should see Status: Add successful.

**User ID:** softphone1

**Password:** S3cr3tW0rd [or any other password of your choice]

**Confirm Password:** S3cr3tW0rd [must match Password]

**Last name:** Softphone 1

**Digest Credentials:** S3cr3tW0rd [or any other password of your choice]

**Confirm Digest Credentials:** S3cr3tW0rd [must match Digest Credentials]

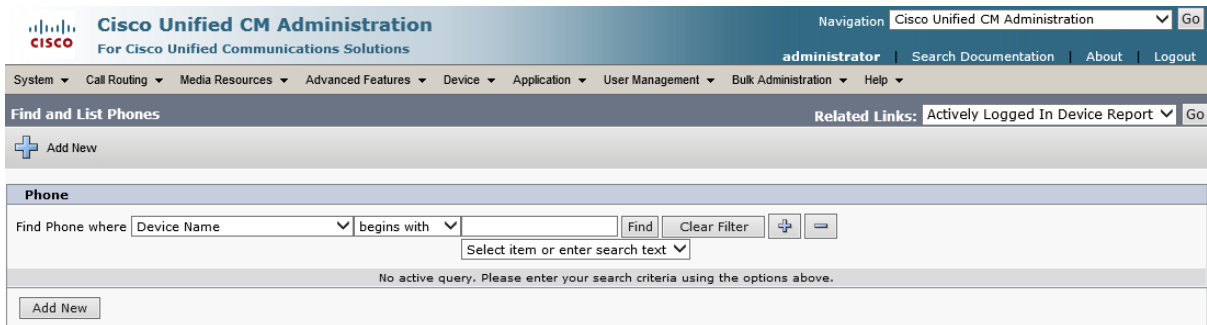
The screenshot shows the 'End User Configuration' page in Cisco Unified CM Administration. The top navigation bar is the same as in the previous screenshot. Below it, the 'End User Configuration' header includes a 'Save' button. The 'Status' section shows 'Status: Ready'. The 'User Information' section contains the following fields and values:

User Status	Enabled Local User
User ID*	softphone1
Password	.....
Confirm Password	.....
Self-Service User ID	
PIN	
Confirm PIN	
Last name*	Softphone 1
Middle name	
First name	
Title	
Directory URI	
Telephone Number	
Home Number	
Mobile Number	
Pager Number	
Mail ID	
Manager User ID	
Department	
User Locale	< None >
Associated PC	
Digest Credentials	.....
Confirm Digest Credentials	.....
User Profile	Use System Default( *Standard (Factory Default) L

At the bottom right of the 'User Profile' field, there is a 'View Details' link.

## Add a Phone for the Softphone

- a) In the top menu, click Device, Phone, then click Add New.



The screenshot shows the Cisco Unified CM Administration web interface. The top navigation bar includes the Cisco logo, the title 'Cisco Unified CM Administration', and the subtitle 'For Cisco Unified Communications Solutions'. The user is logged in as 'administrator'. The main menu includes System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The 'Device' menu is expanded, showing 'Find and List Phones'. Below this, there is a search bar with a dropdown for 'Device Name', a 'begins with' dropdown, and a search button. A message below the search bar reads: 'No active query. Please enter your search criteria using the options above.' There is an 'Add New' button at the bottom left of the search area.

- b) In the Phone Type drop-down, select Third-party SIP Device (Basic) and click Next.  
c) Enter the following information on the Phone Configuration screen and click Save, then OK in the dialog prompting you to apply the configuration. You should see Status: Add successful.

**MAC Address:** This field is actually ignored in our setup and can be any unique 12-digit hex number (0-9, A-F) without punctuation; the actual MAC address of the computer on which you will run the softphone can be determined by pressing Windows+R, typing cmd <enter>, then typing ipconfig /all <enter> and noting the Physical Address for the network connection currently in use

**Description:** Softphone 1

**Device Pool:** Default

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

**Owner User ID:** softphone1 [CUCM 9.1 and above only]

**Device Security Profile:** Third-party SIP Device Basic - Digest Authentication

**SIP Profile:** Standard SIP Profile

**Digest User:** softphone1



Phone Configuration

Save

Status

Status: Ready

Phone Type

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

Device Information

⚠ Device is not trusted

MAC Address*	FEDCBA987654	
Description	Softphone 1	
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	<a href="#">View Details</a>
Calling Search Space	< None >	
AAR Calling Search Space	< None >	
Media Resource Group List	< None >	
Location*	Hub_None	
AAR Group	< None >	
Device Mobility Mode*	Default	<a href="#">View Current Device Mobility Settings</a>
Owner	<input checked="" type="radio"/> User <input type="radio"/> Anonymous (Public/Shared Space)	
Owner User ID*	softphone1	
Use Trusted Relay Point*	Default	
Always Use Prime Line*	Default	
Always Use Prime Line for Voice Message*	Default	
Geolocation	< None >	
<input type="checkbox"/> Ignore Presentation Indicators (internal calls only)		
<input checked="" type="checkbox"/> Logged Into Hunt Group		
<input type="checkbox"/> Remote Device		

Number Presentation Transformation

<b>Caller ID For Calls From This Phone</b>	
Calling Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS (Caller ID For Calls From This Phone)	
<b>Remote Number</b>	
Calling Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS (Device Mobility Related Information)	

Protocol Specific Information

BLF Presence Group*	Standard Presence group	
MTP Preferred Originating Codec*	711ulaw	
Device Security Profile*	Third-party SIP Device Basic - Digest Authenticatic	
Rerouting Calling Search Space	< None >	
SUBSCRIBE Calling Search Space	< None >	
SIP Profile*	Standard SIP Profile	<a href="#">View Details</a>
Digest User	softphone1	
<input type="checkbox"/> Media Termination Point Required		
<input type="checkbox"/> Unattended Port		
<input type="checkbox"/> Require DTMF Reception		

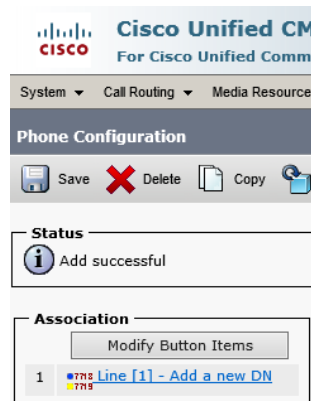
MLPP and Confidential Access Level Information

MLPP Domain	< None >
Confidential Access Mode	< None >
Confidential Access Level	< None >

Save

## Add a Directory Number (DN) for the Softphone

- a) In the Association Information panel on the left of the screen, click Line [1] - Add a new DN.



- b) Enter the following information on the Directory Number Configuration screen and click Save. You should see a new Associated Devices list appear with a single entry representing the softphone which you have just added.

**Directory Number:** 101 [or any other convenient unused extension number]

**Description:** Softphone 1

- c) Finally, click Apply Config and OK in the popup window which appears.

## Register the Softphone to CUCM

- a) It should now be possible to register a softphone to CUCM. For this example we will use the free PhonerLite application. Download the application from <http://www.phoner.de/PhonerLite.zip>, extract the contents of the zip file and run PhonerLite.exe (no installation is required). A wizard will appear prompting you to enter the details of your SIP server. Ensure manual configuration is selected in the list on the left and enter the following information, clicking the green 'next' arrow to confirm each page and the green tick to finish. In the status bar at the bottom of the

PhonerLite window you should see a green light and a message that it has registered to the CUCM server.

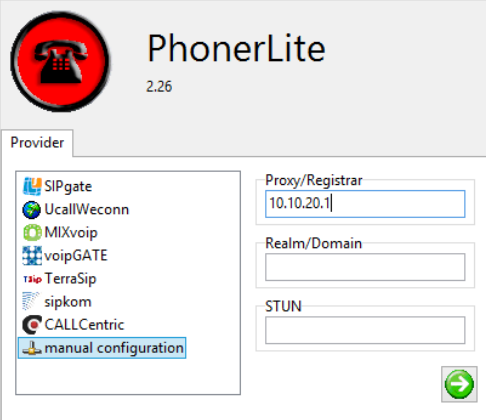
**Proxy/Registrar:** <IP address of CUCM installation>

**User name:** 101 [must match Directory Number associated with softphone Phone on CUCM above]

**Authentication name:** softphone1

**Password:** S3cr3tW0rd [must match Digest Credentials associated with softphone End User on CUCM above]

**'with the profile name':** CUCM



Setup Wizard

PhonerLite 2.26


Provider

- SIPgate
- UcallWeconn
- MIXvoip
- voipGATE
- TerraSip
- sipkom
- CALLCentric
- manual configuration

Proxy/Registrar: 10.10.20.1

Realm/Domain:

STUN:



Setup Wizard

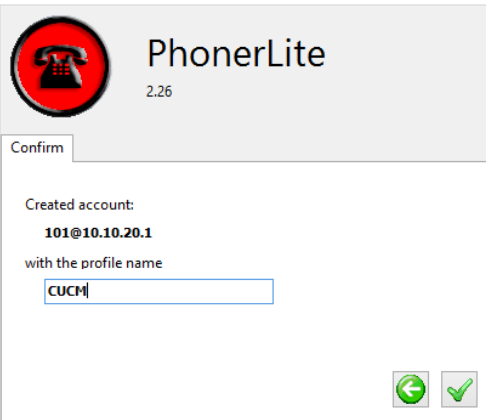
PhonerLite 2.26

User data

User name: 101  
101@10.10.20.1

Authentication name: softphone1

Password: ●●●●●●●●



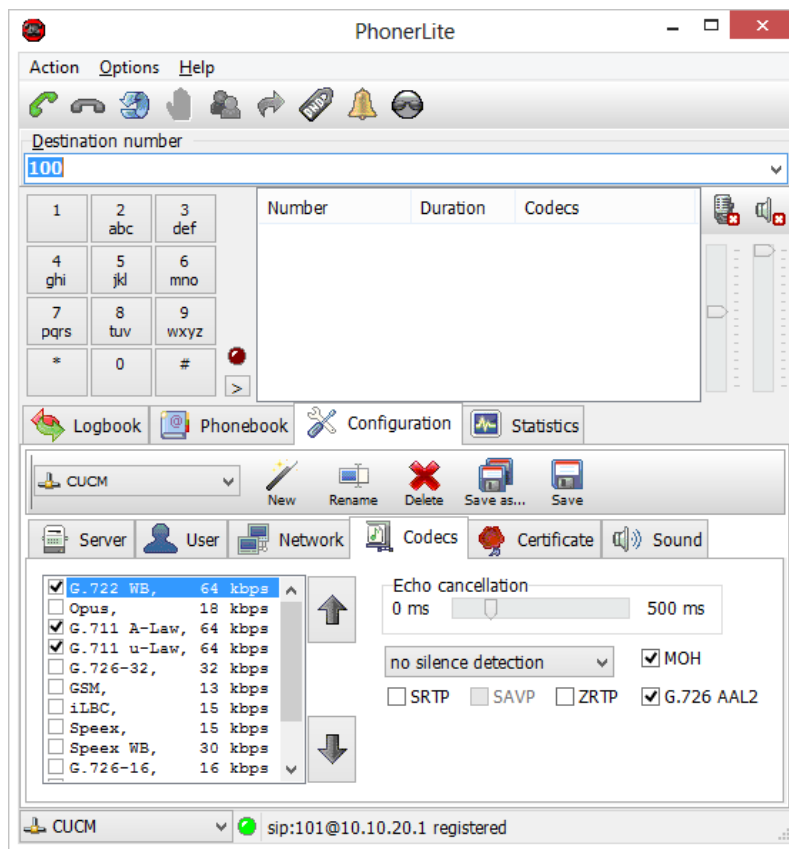
Setup Wizard

PhonerLite 2.26

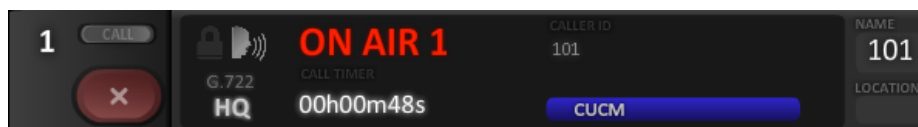
Confirm

Created account:  
**101@10.10.20.1**  
with the profile name  
CUCM

- b) Click the Configuration tab, then the Codecs tab, scroll down in the list of codecs to find G.722 WB, select it and use the up arrow to move it to the top of the list. Then uncheck all codecs in the list except G.722 WB, G.711 A-Law and G.711 u-Law. Click Save.



- c) In the Destination number field, enter 100 (or whichever Directory Number you associated with STAC-VIP on CUCM above) and press Enter. STAC-VIP should indicate an incoming call on the first of the lines associated with CUCM. Answer the call by putting it on air to confirm that everything is working as expected.



- d) To test the conferencing of multiple calls on STAC-VIP via CUCM, repeat the following steps using a different PC as the softphone client:

- Add an End User for the Softphone (using a different User ID and Last name)
- Add a Phone for the Softphone (using a different MAC Address, Description, Owner User ID [CUCM 9.1 and above only] and Digest User)
- Add a Directory Number (DN) for the Softphone (using a different Directory Number and Description)
- Register the Softphone to CUCM (using the different User name and Authentication name)

## NOTES

- Each Third-party SIP Device configured in CUCM requires a different Digest User, so a different End User must be set up for each, as we have done above.
- The crucial difference between CUCM's Basic and Advanced Third-party SIP Device is that the former allows a maximum of two concurrent calls, while the latter allows up to 16. It is for this reason that the Advanced version has to be used for STAC-VIP. Note that the Basic Device consumes three Device Licence Units and the Advanced Device consumes six Device Licence Units.
- If you wish to register to the server a SIP device which does not provide support for separate username and authentication user, the User ID of the End User (which is selected as the Digest User in the Phone Configuration screen) can be made the same as the Directory Number assigned to the device.
- If you wish to associate a caller ID name with the STAC-VIP or a softphone to be displayed to the callee, you should configure this on CUCM. In the top menu, click Device, Phone. In the 'Find Phone where' filter, select Description, begins with, enter the description set on the STAC-VIP / relevant softphone in the empty field and click Find. Click the link in the Device Name(Line) column, then in the Association Information panel on the left of the screen, click Line [1] - ... . Enter the name in the Display (Caller ID) [CUCM 10.5] or Display (Internal Caller ID) [CUCM 9.1 and below] field on the Directory Number Configuration screen in the 'Line 1 on Device SEP...' section. Click Save, then Apply Config and OK in the popup window which appears.

Line 1 on Device SEP0123456789AB	
Display (Caller ID)	<input type="text"/> Display text
ASCII Display (Caller ID)	<input type="text"/>
External Phone Number Mask	<input type="text"/>
Monitoring Calling Search Space	< None > ▼

- *[CUCM 9.1 and below only]*  
If you wish to associate a caller ID number with the STAC-VIP to be displayed to the callee, you should configure this on the STAC-VIP. Browse to the IP address of your STAC-VIP Mainframe, log in, click Configure and log in again. Click Line Configuration, VoIP Providers, CUCM, Show Advanced, Outgoing Caller ID Number, enter the number and click Save Setting. Note that the Outgoing Caller ID Name setting does not apply for registrations to CUCM (it is overridden by the Display (Internal Caller ID) setting in CUCM; see above). Click Back, then Restart (Apply Changes) and OK in the confirmation dialog.

## SIP Settings

Outgoing Caller ID Name	
Outgoing Caller ID Number	100

- To associate different Directory Numbers with different lines on the STAC-VIP, repeat the following sections:

Add an End User for STAC-VIP (using a different User ID and Last name)

Add a Phone for STAC-VIP (using a different MAC Address, Description, Owner User ID [CUCM 9.1 and above only] and Digest User)

Add a Directory Number (DN) for STAC-VIP (using a different Directory Number and Description)

Register STAC-VIP to CUCM (using a different Name, the different Account username and Auth Username and different lines for the new Provider in Line Assignments)

- We have configured CUCM to authenticate the SIP devices above (STAC-VIP and softphones) using a username and password. CUCM's default behaviour, however, is not to check the password, since Cisco proprietary IP phones send their MAC address when registering and this is validated by the server. If security is of no concern, when configuring STAC-VIP on CUCM, you can omit the creation of a secure SIP profile (the entirety of the Add a Phone Security Profile for STAC-VIP section), omit Digest Credentials and Confirm Digest Credentials in the Add an End User for STAC-VIP section and use Device Security Profile: Third-party SIP Device Advanced - Standard SIP Non-Secure Profile in the Add a Phone for STAC-VIP section. Any value can then be used for the STAC-VIP Account password setting in the Register STAC-VIP to CUCM section (but note that STAC-VIP will fail to register unless some value is entered). Similarly, when configuring the softphone on CUCM, you can omit the creation of a secure SIP profile (the entirety of the Add a Phone Security Profile for the Softphone section), omit Digest Credentials and Confirm Digest Credentials in the Add an End User for the Softphone section and use Device Security Profile: Third-party SIP Device Basic - Standard SIP Non-Secure Profile in the Add a Phone for the Softphone section. Any value can then be used for the PhonerLite Password setting in the Register the Softphone to CUCM section (but note that PhonerLite will fail to register unless some value is entered).
- Occasionally, after clicking Apply Config and OK on the Directory Number Configuration screen for either STAC-VIP or a softphone, the device whose configuration has been changed will not be able to make a call via CUCM - the call will proceed to the SIP 'ringing' stage and then be cleared around half a second later by the server, which reports the problem as 'Temporary failure'. The solution is simply to wait a couple of minutes and retry.

## REFERENCES

- [1] Cisco Systems, Inc., "Installing Cisco Unified Communications Manager, Release 8.6(1)," [Online]. Available: [http://www.cisco.com/c/en/us/td/docs/voice\\_ip\\_comm/cucm/install/8\\_6\\_1/install/cmins861.pdf](http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/install/8_6_1/install/cmins861.pdf).
- [2] Cisco Systems, Inc., "Installing Cisco Unified Communications Manager, Release 9.1(1)," [Online]. Available: [http://www.cisco.com/c/en/us/td/docs/voice\\_ip\\_comm/cucm/install/9\\_1\\_1/CUCM\\_BK\\_I05CD008\\_00\\_installing-cucm-91.pdf](http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/install/9_1_1/CUCM_BK_I05CD008_00_installing-cucm-91.pdf).
- [3] Cisco Systems, Inc., "Installing Cisco Unified Communications Manager, Release 10.0(1)," [Online]. Available: [http://www.cisco.com/c/en/us/td/docs/voice\\_ip\\_comm/cucm/install/10\\_0\\_1/CUCM\\_BK\\_I95AD2FE\\_00\\_installing-cucm-100.pdf](http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/install/10_0_1/CUCM_BK_I95AD2FE_00_installing-cucm-100.pdf).
- [4] Cisco Systems, Inc., "Cisco Unified Communications Manager Administration Guide, Release 8.6(1)," [Online]. Available: [http://www.cisco.com/c/en/us/td/docs/voice\\_ip\\_comm/cucm/admin/8\\_6\\_1/ccmcf/bccm-861-cm.pdf](http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/8_6_1/ccmcf/bccm-861-cm.pdf).
- [5] Cisco Systems, Inc., "Cisco Unified Communications Manager Administration Guide, Release 9.1(1)," [Online]. Available: [http://www.cisco.com/c/en/us/td/docs/voice\\_ip\\_comm/cucm/admin/9\\_1\\_1/ccmcf/CUCM\\_BK\\_A34970C5\\_00\\_admin-guide-91.pdf](http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/9_1_1/ccmcf/CUCM_BK_A34970C5_00_admin-guide-91.pdf).
- [6] Cisco Systems, Inc., "Cisco Unified Communications Manager Administration Guide, Release 10.0(1)," [Online]. Available: [http://www.cisco.com/c/en/us/td/docs/voice\\_ip\\_comm/cucm/admin/10\\_0\\_1/ccmcf/CUCM\\_BK\\_C95ABA82\\_00\\_admin-guide-100.pdf](http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmcf/CUCM_BK_C95ABA82_00_admin-guide-100.pdf).
- [7] Comrex Corporation, "Comrex Device Manager," [Online]. Available: <http://www.comrex.com/downloads/software/ComrexDeviceManagerInstall-2.0.121-2014-08-27.exe>.