

▶ Deployment Guide for the
Polycom® SoundStructure VoIP
Interface for Cisco® Unified
Communications Manager (SIP)

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About This Guide

This guide explains how to deploy the Polycom® SoundStructure VoIP Interface in a Cisco® Unified Communications Manager (CUCM) environment for voice calls.

In this guide, you will find everything you need to set up the SoundStructure VoIP Interface in an environment running CUCM version 8.5. While CUCM versions 6 and 7 are compatible, the instructions and screen captures were created with CUCM version 8.5.

The following related documents SoundStructure VoIP Interface are available:

- SoundStructure Design Guide, which describe how to create SoundStructure projects with the SoundStructure VoIP Interface
- Quick Upgrade Guide, which describes how to upgrade an existing SoundStructure-based TEL1 or TEL2 system to SoundStructure VoIP Interface system
- UC Software 4.0.1 Administrator's Guide, which describes how to configure, customize, manage, and troubleshoot Polycom SoundPoint® IP, SoundStation® IP, and VVX phone systems
- Technical Bulletins, which describe workarounds to existing issues and provide expanded descriptions and examples
- Release Notes, which describe the new and changed features and fixed problems in the latest version of the software

For support or service, please contact your Polycom® reseller or go to Polycom Technical Support at <http://www.polycom.com/support/voice/>.

Polycom recommends that you record the phone model numbers and software (both the BootROM and Polycom® UC Software) for future reference.

SoundStructure VoIP Interface MAC Address: _____

UC Software version: _____

Partner Platform: _____

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Overview

This guide provides information for system administrators who want to deploy the SoundStructure VoIP Interface in a Cisco® Unified Communications Manager environment.

The SoundStructure VoIP Interface is compatible with the following: Cisco Unified Communications Manager versions 6, 7, and 8.5. This guide was developed using version 8.5.

The topics in this guide include:

- [Configuring Cisco Unified Communications Manager to manage the SoundStructure VoIP Interface phone. Refer to *Configuring Cisco Unified Communications Manager* on page 2-1.](#)
- [Setting up the SoundStructure VoIP Interface phone and changing the configuration files to work with Cisco Unified Communications Manager. Refer to *Setting Up the SoundStructure VoIP Interface* on page 3-1.](#)

The topics in this chapter include:

- [Supported Phone Features](#)
- [Unsupported Phone Features](#)
- [Unsupported Cisco Features](#)
- [Topics Not Mentioned in This Guide](#)

Supported Phone Features

The following features are supported on the SoundStructure VoIP Interface:

- 2 to 16 call appearances
- Place a call/Receive a call
- Hold/Resume
- Consultative and blind transfer
- Incoming call forward

- Message Waiting Indicator
- Three-way audio conference with management options (join, split, mute)
- Server Redundancy

Unsupported Phone Features

The following features are not supported on the SoundStructure VoIP Interface due to proprietary Cisco extensions:

- Call Pickup, Group Call Pickup, Directed Call Pickup
- Call Park
- Presence and buddy lists
- Instant Messaging
- Automatic Call Distribution (ACD) Login/Logout
- Secure Real-Time Transport Protocol (SRTP)
- Last Call Return
- Barge-In
- Busy Lamp Field
- Conveying microphone mute status between endpoints

Unsupported Cisco Features

The following Cisco features are not supported on the SoundStructure VoIP Interface:

- Cisco XML Applications
- Cisco Phone directory
- Cisco ad-hoc conferencing
- Cisco TFTP software/configuration file download

Topics Not Mentioned in This Guide

The following topics are not covered in this guide:

- Configuration of other SIP registrars or proxy servers
- Cisco Unified Presence

Configuring Cisco Unified Communications Manager

The SoundStructure VoIP Interface, managed by Cisco Unified Communications Manager, is designed to be used like a traditional telephone on a public switched telephone network (PSTN).

This chapter provides basic instructions for setting up the SoundStructure VoIP Interface in a Cisco Unified Communications Manager (CUCM) v8.5 environment.

Changing Cisco Unified Communications Manager

The following settings must be verified or adjusted before connecting the SoundStructure VoIP Interface phone to Cisco Unified Communications Manager:

- Region settings should allow for an audio codec of G.722 for the best audio experience.
- Each SoundStructure VoIP Interface requires six Device License Units.

Adding the SoundStructure VoIP Interface to Cisco Unified Communications Manager requires three major tasks:

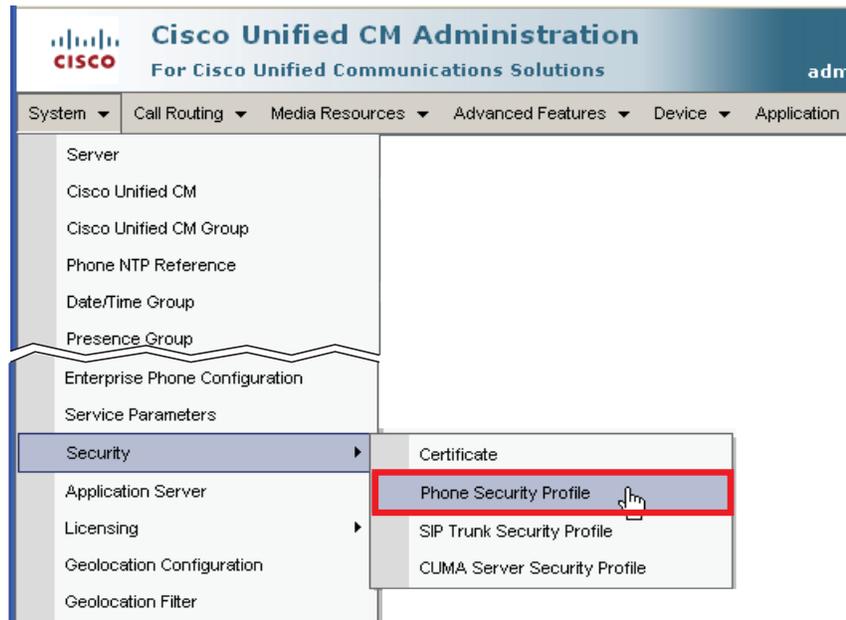
- Setting the phone security configuration
- Setting the phone user configuration
- Setting the phone device configuration

The following sections provide step-by-step instructions for adding a SoundStructure VoIP Interface to a Cisco Unified Communications Manager.

Part I: Phone Security Settings

To create the security profile that is used by the phone, follow these steps:

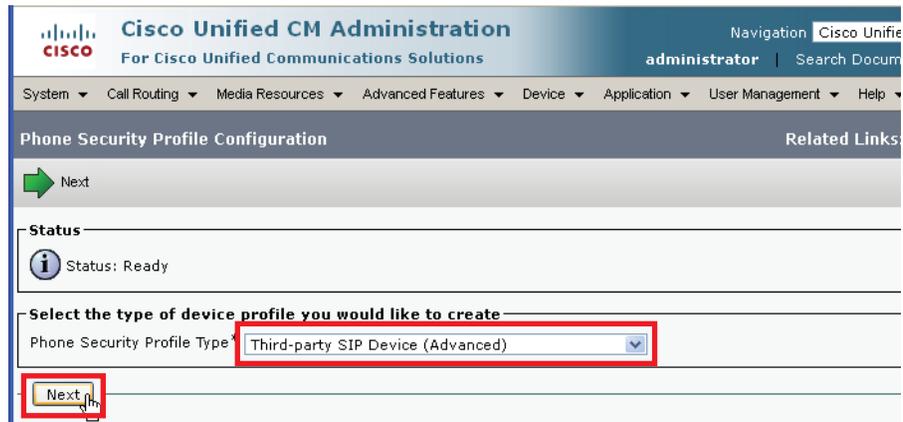
1. Open a Cisco Unified Communications Manager web administration session.
You will be prompted to enter your user name and password.
2. Select **System > Security > Phone Security Profile**.



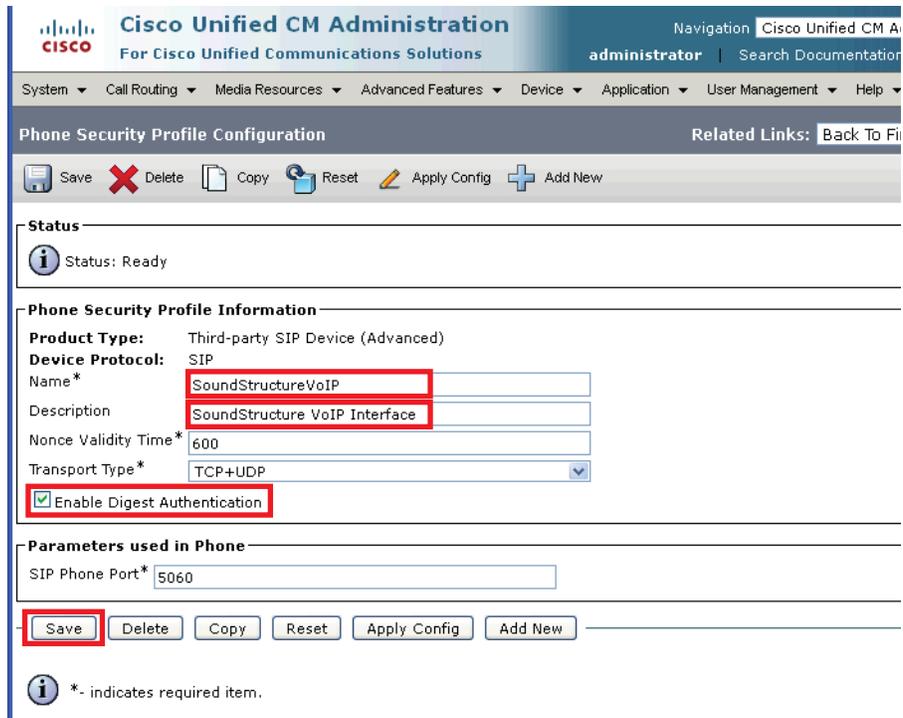
3. Click **Add New**.



4. Select **Third-party SIP Device (Advanced)** and click **Next**.



5. Enter the Phone Security Profile Information. In the **Name** text box, enter a profile name appropriate for the system, optionally enter a Description, and enable the **Enable Digest Authentication** check box.



6. Click the **Save** button.

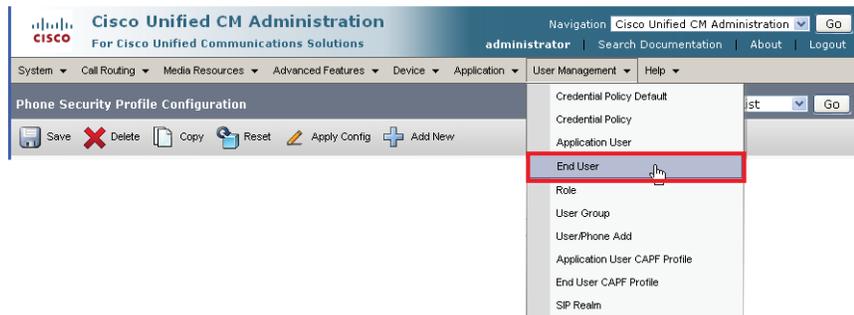
In the status bar near the top of the page, “Update Successful” should appear as shown next.



Part II: Adding a Phone User

Once the phone security settings have been created, you need to create a user for the SoundStructure VoIP Interface as detailed in the following steps.

1. To create a user, select **User Management > End User**



and click **Add New** as shown in the following figure.



- In the **User ID** text box, enter a user ID according to system and account policies. In this example, the user name is set to **sstvoipuser**.

Status	
 Status: Ready	
User Information	
User ID*	<input type="text" value="sstvoipuser"/>
Password	<input type="text"/>
Confirm Password	<input type="text"/>
PIN	<input type="text"/>
Confirm PIN	<input type="text"/>

If you cannot add a user here, your system may be LDAP integrated, in which case you can use an existing user ID (essentially associating the phone to an existing user) or have your LDAP administrator create a new user ID for this phone.

- In the **Last Name** text box, enter a last name. In this example, the value entered was LastName.

Status	
 Status: Ready	
User Information	
User ID*	<input type="text" value="sstvoipuser"/>
Password	<input type="text"/>
Confirm Password	<input type="text"/>
PIN	<input type="text"/>
Confirm PIN	<input type="text"/>
Last name*	<input type="text" value="LastName"/>

4. In the **Digest Credentials** text box, enter the digital credentials for the phone. Enter the same credentials in the **Confirm Digest Credentials** field.

This password will be used with the User ID as the authentication password in the phone's configuration file or when entering the line registration information with the Web Configuration Utility.

User Information	
User ID*	<input type="text" value="sstvoipuser"/>
Password	<input type="text"/>
Confirm Password	<input type="text"/>
PIN	<input type="text"/>
Confirm PIN	<input type="text"/>
Last name*	<input type="text" value="LastName"/>
Middle name	<input type="text"/>
First name	<input type="text"/>
Telephone Number	<input type="text"/>
Mail ID	<input type="text"/>
Manager User ID	<input type="text"/>
Department	<input type="text"/>
User Locale	< None > <input type="button" value="v"/>
Associated PC	<input type="text"/>
Digest Credentials	<input type="password" value="....."/>
Confirm Digest Credentials	<input type="password" value="....."/>

5. Click the **Save** button.
In the status bar near the top of the page, "Update Successful" appears.

Part III: Adding a Device Entry

Once the user information has been entered, you may add the device information for the SoundStructure VoIP Interface.

1. Select **Device > Phone** and click **Add New**.



2. Select **Third-party SIP Device (Advanced)** as shown below,

The screenshot shows the 'Add a New Phone' configuration page in Cisco Unified CM Administration. The page title is 'Cisco Unified CM Administration For Cisco Unified Communications Solutions'. The user is logged in as 'administrator'. The navigation menu includes System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, and Help. The 'Add a New Phone' section has a 'Next' button with a green arrow. Below this, the 'Status' is 'Ready'. The 'Select the type of phone you would like to create' section shows 'Phone Type*' set to 'Third-party SIP Device (Advanced)'. A 'Next' button is highlighted with a red box.

and click **Next**.

3. Enter the Device Information as shown in the following screen and described next.

Note that the data shown in this section is only an example.

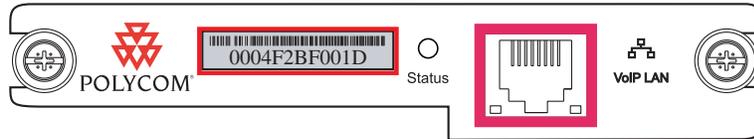
The screenshot shows the 'Device Information' configuration page. A warning icon indicates 'Device is not trusted'. The fields are as follows:

MAC Address*	0004F2BF001D
Description	Example SST VoIP Interface in Conference Room
Device Pool*	Default
Common Device Configuration	< None >
Phone Button Template*	Third-party SIP Device (Advanced)
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
Owner User ID	< None >
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 >

Checkboxes at the bottom:

- Use Device Pool Calling Party Transformation CSS
- Retry Video Call as Audio
- Ignore Presentation Indicators (internal calls only)
- Logged Into Hunt Group
- Remote Device

4. In the **MAC Address** text box, enter the MAC Address of the SoundStructure VoIP Interface. The MAC address is on the rear panel of the SoundStructure VoIP Interface as shown in the following figure and can also be found on the SoundStructure Studio Wiring page.



Device Information	
General	
Device status:	ok
Device type:	c16
Bus ID:	1
Ethernet MAC:	00:04:f2:bf:01:3e
Plug-in card:	voip
Uptime:	7d 21h 44m 50s
VoIP Interface	
VoIP Status:	ok
IP Address:	10.240.3.141
UC Software Version:	Mink 4.0.1.10052 03-Nov-11 22:31 file
BootROM Version:	5.0.1.8147 03-Nov-11 23:25
Bootblock Version:	3.0.3.0012 (33215-001) 07-Oct-11 08:39
Board Information:	3111-33215-001 Rev=2 Region=0, MAC=00:04:F2:BF:00:1D

5. (Optional) In the **Description** text box, enter a description.
6. From the **Device Pool** list, select the device pool appropriate for your Cisco Unified Communications Manager system phones.
7. From the **Phone Button Template** list, select **Third-party SIP Device (Advanced)**.
8. (Optional) From the **Calling Search Space** list, select an appropriate calling search space for the phone.
9. From the **Location** list, select an appropriate location for the phone.

- From the **Device Security Profile** list, select the profile created in step 5 of [Part I: Phone Security Settings](#).

The following screen appears. Note that the data shown in this section is only an example.

- In the **SIP Profile** field, enter the desired SIP profile.
- In the **Digest User** field, select the user created in step 2 of [Part II: Adding a Phone User](#). In this example, the user is sstvoipuser.
- Click the **Save** button.

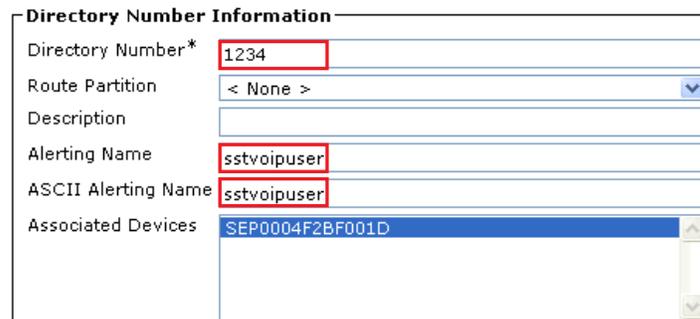
In the status bar near the top of the page, "Update Successful" appears.

- Click **Apply Config**. The resulting status message will appear. Click **OK** to continue.

- In the **Association Information** area on the left side of the window, click on the **Line [1] – Add a new DN** link.



- In the **Directory Number** text box, enter an appropriate phone extension. In this example, the extension 1234 is used.



- From the **Route Partition** list, select an appropriate route partition.
- In the **Alerting Name** text box, enter an appropriate alerting name. In this example, sstvoipuser is used.
- In the **ASCII Alerting Name** text box, enter an appropriate ASCII alerting name. In this example, sstvoipuser is used.

20. Set the Voice Mail Profile according to the Cisco Unified Communications Manager system requirements. The default settings were used in the following figure.

Directory Number Settings

Voice Mail Profile	< None >	(Choose <None> to use system)
Calling Search Space	< None >	
Presence Group*	Standard Presence group	
User Hold MOH Audio Source	< None >	
Network Hold MOH Audio Source	< None >	

21. Set the Call Forward and Call Pickup Settings to appropriate values for your system. In this example, no settings were changed.

Call Forward and Call Pickup Settings

	Voice Mail	Destination	Calling Search Space
Calling Search Space Activation Policy			Use System Default
Forward All	<input type="checkbox"/> or		< None >
Secondary Calling Search Space for Forward All			< None >
Forward Busy Internal	<input type="checkbox"/> or		< None >
Forward Busy External	<input type="checkbox"/> or		< None >
Forward No Answer Internal	<input type="checkbox"/> or		< None >
Forward No Answer External	<input type="checkbox"/> or		< None >
Forward No Coverage Internal	<input type="checkbox"/> or		< None >
Forward No Coverage External	<input type="checkbox"/> or		< None >
Forward on CTI Failure	<input type="checkbox"/> or		< None >
Forward Unregistered Internal	<input type="checkbox"/> or		< None >
Forward Unregistered External	<input type="checkbox"/> or		< None >
No Answer Ring Duration (seconds)			
Call Pickup Group			< None >

22. In the Display (Internal Caller ID) text box, enter an appropriate caller ID. In this example, Conference Room was entered. This text will be shown on the recipient's phone when a call is received from the SoundStructure VoIP Interface.

Line 1 on Device SEP0004F2BF001D

Display (Internal Caller ID)	Conference Room	Display text for a line appearance is internal text such as a name instead of a directory number for internal calls. If you specify a number, the call may not see the proper identity of the caller.
ASCII Display (Internal Caller ID)	Conference Room	
External Phone Number Mask		
Monitoring Calling Search Space	< None >	

23. In the **ASCII Display (Internal Caller ID)** text box, enter an appropriate caller ID. In this example, **Conference Room** was entered.

24. In the **Maximum Number of Calls** text box, enter an appropriate value for your system.

Multiple Call/Call Waiting Settings on Device SEP0004F2BF001D

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

Maximum Number of Calls*	<input type="text" value="2"/>
Busy Trigger*	<input type="text" value="2
Calls)"/>

Currently the maximum supported value for the SoundStructure VoIP Interface is 24.

25. In the **Busy Trigger** text box, enter an appropriate value for your system.

Currently the maximum supported value for the SoundStructure VoIP Interface is 24.

26. Click the **Save** button.

In the status bar near the top of the page, "Update Successful" appears.

You have completed the configuration of Cisco Unified Communications Manager.

Setting Up the SoundStructure VoIP Interface

This chapter provides instructions for setting up the SoundStructure VoIP Interface to register to the Cisco® Unified Communications Manager settings configured in the previous chapter.

Because of the large number of optional installations and configurations that are available, this chapter focuses on one particular way that the SoundStructure VoIP Interface may be configured in your network.

For more information on configuring the SoundStructure VoIP Interface phone, refer to the *Polycom UC Software Administrators Guide 4.0.1*, which is available at

http://supportdocs.polycom.com/PolycomService/support/global/documents/support/setup_maintenance/products/voice/UC_Software_Admin_Guide_v4_0_1.pdf.

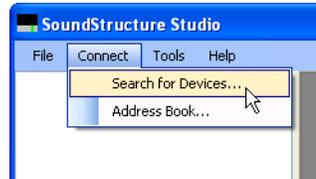
Using the Web Configuration Utility

One approach to configuring the SoundStructure VoIP Interface for operation with a Cisco Unified Communications Manager is to use the Web Configuration Utility.

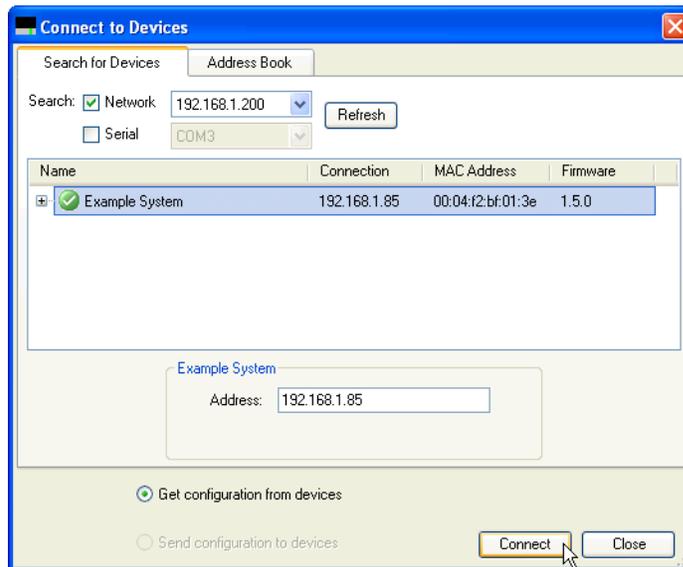
This section summarizes the steps required to register the SoundStructure VoIP Interface with the Call Manager settings that were created in [Configuring Cisco Unified Communications Manager](#).

To configure the SoundStructure VoIP Interface, follow these steps:

1. Connect to your SoundStructure System by selecting **Search for Devices** from the Connect menu as shown in the following figure.

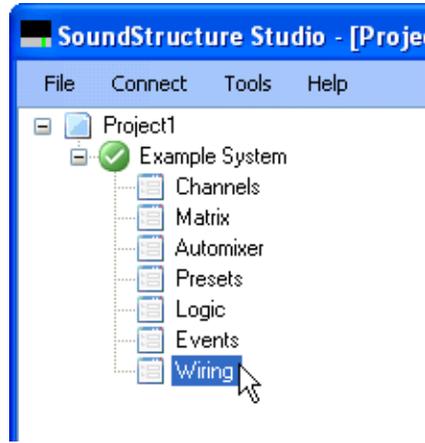


2. If the desired system was discovered, then select the system, highlight the **Get configuration from devices** option and click **Connect** as shown below. If your system is not present, you may either enter the IP address

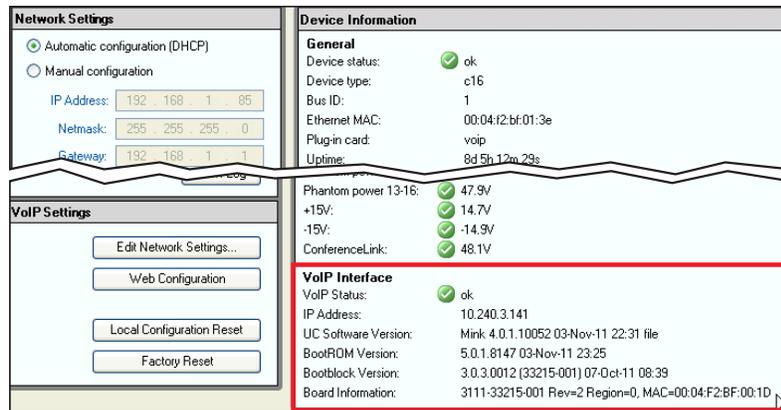


manually or use the Address Book tab to find systems that are stored in your network.

3. Navigate to the Wiring Page by left clicking on the Wiring entry as shown in the following figure.

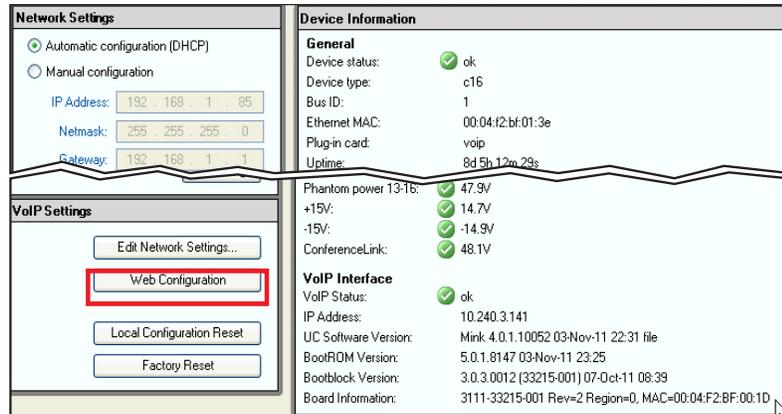


4. Confirm the SoundStructure VoIP Interface is installed in this system by reviewing the information on the wiring page. You should see that there

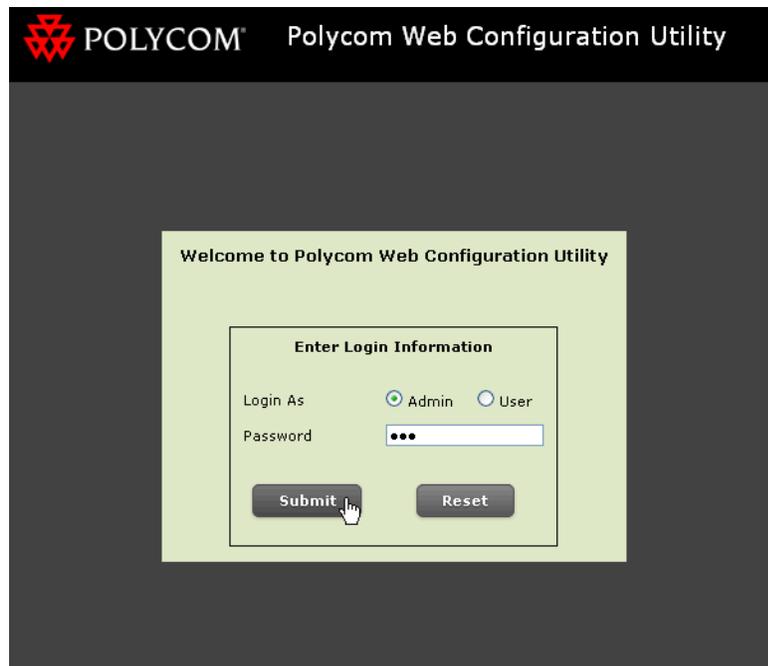


is a SoundStructure VoIP Interface listed with an ok status and a valid IP address.

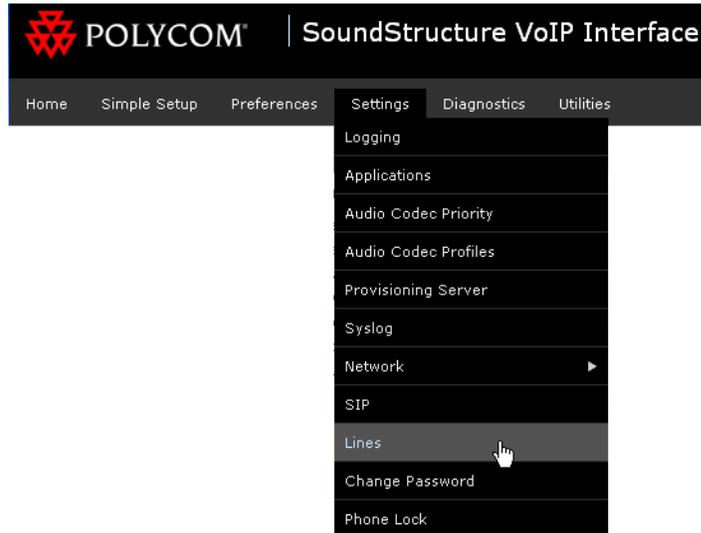
5. If the interface is found, then click on **Web Configuration** button to launch the Web Configuration Utility in your PC's browser.



6. Log into the Web Configuration Utility by selecting **Admin** and entering the default password of **456** and click **Submit**.



- Navigate to the **Settings > Lines** Page.



- Select the desired line to configure. By default this will be **Line 1**. The screen below will display.

 The screenshot displays the configuration page for 'Line 1'. The page has a light green background. At the top, it says 'Line 1'. Below this is a section titled 'Identification' with a minus sign icon. The fields in this section are: Display Name (text input), Address (text input), Authentication User ID (text input), Authentication Password (text input), Label (text input), Type (radio buttons for Private and Shared, with Private selected), Third Party Name (text input), Number of Line Keys (text input with value 1), Calls Per Line (text input with value 24), and Ring Type (dropdown menu with 'Silent Ring' selected). Below the Identification section are several other sections, each with a minus sign icon: Outbound Proxy, Server 1, Server 2, Call Diversion, and Message Center. A mouse cursor is visible over the 'Calls Per Line' field.

- Enter the **Display Name** to be displayed on the dialing page. In this example, enter **1234** as that is the extension that was defined in step 16 of [Part III: Adding a Device Entry in Configuring Cisco Unified Communications Manager](#).
- Enter the **Address**. In this example, enter **1234** to match the Display Name.

11. Enter the **Authentication User ID**. This is the **User ID** field that was entered as **sstvoipuser** in step 2 of [Part II: Adding a Phone User in Configuring Cisco Unified Communications Manager](#).
12. Enter the **Authentication Password**. This is the **Digest Credentials** value that was entered in step 4 of [Part II: Adding a Phone User in Configuring Cisco Unified Communications Manager](#).
13. Enter the **Label** that will appear on the Phone Settings page within SoundStructure Studio. In this example, enter **1234**. The resulting settings are shown in the following figure.

Line 1	
Identification	
Display Name	1234
Address	1234
Authentication User ID	sstvoipuser
Authentication Password	••••
Label	1234
Type	<input checked="" type="radio"/> Private <input type="radio"/> Shared
Third Party Name	
Number of Line Keys	1
Calls Per Line	24
Ring Type	Low Trill

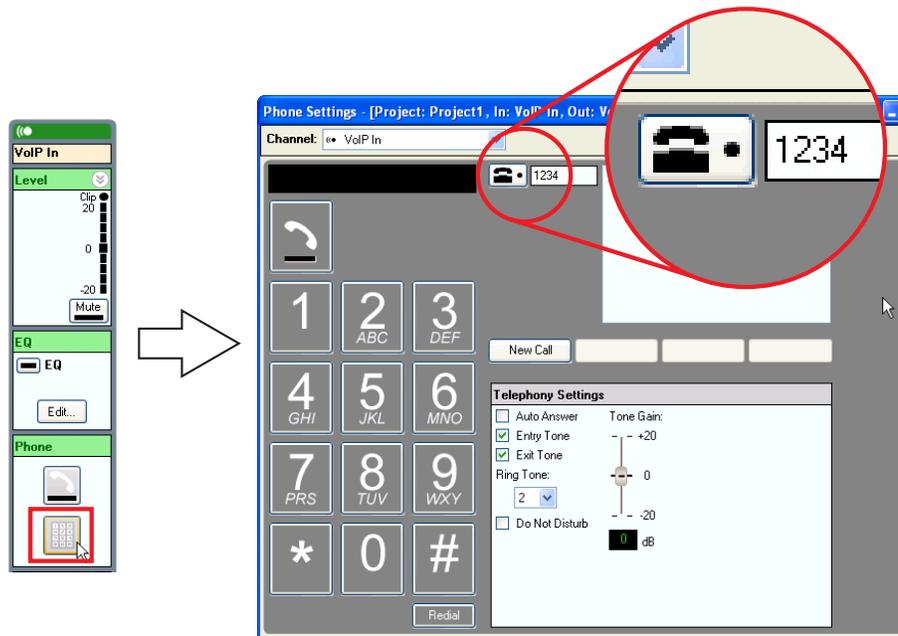
14. Next, expand the **Server 1** control and enter the IP address or hostname of the Cisco Unified Collaborations Manager. In this example the CUCM has an IP address of 172.24.144.119.

Server 1	
Address	172.24.144.119
Port	0
Transport	DNSnaptr
Expires (s)	3600
Register	<input checked="" type="radio"/> Yes <input type="radio"/> No
Retry Timeout (ms)	0
Retry Maximum Count	3
Line Seize Timeout (s)	30

15. Click **Save** to apply the settings.



16. Navigate to the Channels page in SoundStructure Studio and open the Phone Settings control by clicking on the phone icon. If the line has been successfully registered, the phone icon will appear solid as shown in the following figure.



Troubleshooting the SoundStructure VoIP Interface

This chapter contains general troubleshooting information to help you solve problems you might encounter when using a SoundStructure VoIP Interface in a Cisco® Unified Communications Manager environment.

For detailed information on configuring the SoundStructure VoIP Interface, see the SoundStructure Design Guide.

Review the latest *Release Notes* for the UC Software and for the SoundStructure VoIP Interface for known problems and possible workarounds. For the latest *Release Notes* for the SoundStructure products, go to http://http://www.polycom.com/support/voice/soundstructure/c_series.html.

Line Registration Issues

If you do not see the registered Line Icon in the Phone Settings UI within SoundStructure Studio, confirm that the Authentication User ID and Authentication Password match the User ID and Digest Password entered when configuring the Cisco Unified Communications Manager.

If the credentials are correct but the SoundStructure VoIP Interface is still not registering, confirm the IP address or hostname of the CUCM.

If the SoundStructure VoIP Interface is still not registering, check the registration status on the Phone configuration page of the CUCM system as shown in the following figure. If the phone is unregistered the system can appear as in the following figure.

The screenshot displays the Cisco Unified CM Administration interface for a phone configuration. The status is 'Ready'. Under 'Association Information', there are 8 lines listed, with the first line being 'Line [1] - 1234 (no partition)'. The 'Phone Type' section shows 'Product Type: Third-party SIP Device (Advanced)' and 'Device Protocol: SIP'. The 'Device Information' section shows 'Registration: Unregistered' (highlighted with a red box), 'IP Address: 10.240.3.141', 'Active Load ID: Unknown', 'Device is Active' (checked), 'Device is not trusted' (warning icon), 'MAC Address*: 0004F2BF001D', 'Description: Example SST VoIP Interface in Conference Room', 'Device Pool*: Default', 'Common Device Configuration: < None >', 'Phone Button Template*: Third-party SIP Device (Advanced)', and 'Common Phone Profile*: Standard Common Phone Profile'.

Once the phone is properly registered, the Phone Configuration display will appear as shown in the following figure.

The screenshot displays the Cisco Unified CM Administration interface for Phone Configuration. The status is 'Ready'. The device is a 'Third-party SIP Device (Advanced)' with protocol 'SIP'. The registration status is 'Registered with Cisco Unified Communications Manager IDC-C'. The IP address is 10.240.3.141. The device is active but not trusted. The MAC address is 0004F2BF001D. The description is 'Example SST VoIP Interface in Conference Room'. The device pool is 'Default', common device configuration is '< None >', phone button template is 'Third-party SIP Device (Advanced)', and common phone profile is 'Standard Common Phone Profile'.

Association Information	
1	Line [1] - 1234 (no partition)
2	Line [2] - Add a new DN
3	Line [3] - Add a new DN
4	Line [4] - Add a new DN
5	Line [5] - Add a new DN
6	Line [6] - Add a new DN
7	Line [7] - Add a new DN
8	Line [8] - Add a new DN

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

Device Information	
Registration	Registered with Cisco Unified Communications Manager IDC-C
IP Address	10.240.3.141
Active Load ID	Unknown
<input checked="" type="checkbox"/> Device is Active	
Device is not trusted	
MAC Address*	0004F2BF001D
Description	Example SST VoIP Interface in Conference Room
Device Pool*	Default View D
Common Device Configuration	< None > View D
Phone Button Template*	Third-party SIP Device (Advanced)
Common Phone Profile*	Standard Common Phone Profile

