



SOLUTION GUIDE

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# Deploying Polycom<sup>®</sup> SoundStation<sup>®</sup> IP Conference Phones with Cisco<sup>®</sup> Unified Communications Manager (CUCM)



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# About This Guide

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This guide uses a number of conventions that help you to understand information and perform tasks.










## Conventions Used in this Guide

This guide contains terms, graphical elements, and a few typographic conventions. Familiarizing yourself with these terms, elements, and conventions will help you successfully perform tasks.

## Information Elements

This guide may include any of the following icons to alert you to important information.

### Icons Used in this Guide

<i>Name</i>	<i>Icon</i>	<i>Description</i>
Note		The Note icon highlights information of interest or important information needed to be successful in accomplishing a procedure or to understand a concept.
Administrator Tip		The Administrator Tip icon highlights techniques, shortcuts, or productivity related tips.
Caution		The Caution icon highlights information you need to know to avoid a hazard that could potentially impact device performance, application functionality, or successful feature configuration.
Warning		The Warning icon highlights an action you must perform (or avoid) to prevent issues that may cause you to lose information or your configuration setup, and/or affect phone or network performance.
Web Info		The Web Info icon highlights supplementary information available online such as documents or downloads on <a href="http://support.polycom.com">support.polycom.com</a> or other locations.
Timesaver		The Timesaver icon highlights a faster or alternative method for accomplishing a method or operation.
Power Tip		The Power Tip icon highlights faster, alternative procedures for advanced administrators already familiar with the techniques being discussed.
Troubleshooting		The Troubleshooting icon highlights information that may help you solve a relevant problem or to refer you to other relevant troubleshooting resources.
Settings		The Settings icon highlights settings you may need to choose for a specific behavior, to enable a specific feature, or to access customization options.

# Typographic Conventions

A few typographic conventions, listed next, are used in this guide to distinguish types of in-text information.

## Typographic Conventions

<i>Convention</i>	<i>Description</i>
<b>Bold</b>	Highlights interface items such as menus, soft keys, file names, and directories. Also used to represent menu selections and text entry to the phone.
<i>Italics</i>	Used to emphasize text, to show example values or inputs, and to show titles of reference documents available from the Polycom Support Web site and other reference sites.
<a href="#">Blue Text</a>	Used for cross references to other sections within this document and for hyperlinks to external sites and documents.
<code>Courier</code>	Used for code fragments and parameter names.

## What's in This Guide?

This guide is organized sections that show you how to deploy Polycom® SoundStation® IP conference phones with Cisco Unified Communications Manager CUCM.

**Getting Started** This section contains overview information you need to get started deploying SoundStation IP series phones with Cisco Unified Communications Manager.

**Setting Up Cisco Unified Communications Manager** This section shows you how to set up Cisco Unified Communications Manager for use with Polycom SoundStation IP phones.

**Deploying SoundStation IP Phones with Cisco Unified Communications Manager** This section shows you how to integrate SoundStation IP phones with Cisco Unified Communications Manager.

**Troubleshooting the SoundStation IP Phones** This chapter lists troubleshooting problems and solutions common when deploying SoundStation IP phones.

**Getting Help** In this chapter, you'll find links to Polycom, partner, and third-party documents and web sites. In particular, you'll find links to the Polycom Community, a number of discussion forums you can use to share ideas with your colleagues.

# Getting Started

This guide shows you how to deploy the Polycom® SoundStation® IP conference phones in a Cisco® Unified Communications Manager (CUCM) environment. Note that CUCM environments differ and this guide does not account for a particular CUCM environment. This guide illustrates deployment of SoundStation IP phones in a CUCM environment version 8.6 or later. You can use this guide to deploy SoundStation IP phones in CUCM environment versions 6 and 7, however, the instructions and figures in this guide refer CUCM version 8.6.

Note that you can deploy SoundStation IP conference phones with Cisco Business Edition 6000 and Cisco Unified Communications Manager Express; however, Polycom has not performed interoperability tests.

Polycom currently supports the following SoundStation IP conference phones.

**Table 1: SoundStation IP Conference Phones**

<b>SoundStation IP 5000</b>	<b>SoundStation IP 6000</b>
	
	

## Hardware and Software Dependencies

Polycom recommends using the latest version of UC software. When deploying SoundStation IP phones in CUCM environments, Polycom supports CUCM deployments using UC software release 4.0.4 and above (Except for software versions identified for use with Microsoft® Lync™ Server.). However, older versions of UC Software are compatible. Note that if you are using SIP software version 3.2.x or previous, you must use a provisioning server and Polycom configuration files. The deployment scenarios outlined in this guide are compatible with previous versions of UC software listed in Table 2. Use Table 2 to match a phone with a compatible UC software release.

**Table 2: Compatible Phones and UC Software**

<i>Phone</i>	<i>UC Software Release</i>
SoundStation IP 5000	4.0.4
	3.3.x
	3.2.3 - 3.2.7 Requires use of a provisioning server
SoundStation IP 6000	4.0.4
	3.3.x
	3.2.x Requires use of a provisioning server
SoundStation IP 7000	4.0.4
	3.3x
	3.2.x
SoundStation Duo	4.0.4



**Note: CUCM Does Not Support UC Software 4.1.x**

UC software versions 4.1.x are for use with Microsoft Lync Server. Do not use UC software 4.1.x with CUCM.

## Supported Phone Features

The following table indicates which features the SoundStation IP phones supports when deployed when deployed with CUCM.

**Table 3: CUCM Features on SoundStation IP Phones**

<i>Feature</i>	<i>Supported / Not Supported</i>
Place and receive calls	Supported
On-hook dialing	Supported

<i>Feature</i>	<i>Supported / Not Supported</i>
Do not disturb	Supported
Call hold and resume	Supported
Call Waiting	Supported
Call Appearances (Number of simultaneous calls on a single registration)	CUCM supports up to two call appearances on third-party SIP devices.
Caller ID display	Supported
Speed Dial	Supported
Three-way audio conference with management options	Supported Polycom phones provide conferencing from the phone itself. Cisco phones conference using the CUCM server.
Voice Hunt group	Supported
Incoming Call Forwarding	Supported
Call forward busy	Supported
Call forward no answer	Supported
Call transfer – blind and consultative	Supported
Clock Display	Supported
Music on Hold (MoH)	Supported Polycom phones can receive MoH when placed on hold by another Polycom phone, but CUCM does not support streaming MoH to Cisco phones when the Polycom phone is put on hold.
Message Waiting Indicator (MWI)	Supported
<b>Additional Services</b>	
Busy Trigger	Supported
Missed/Placed/Received calls	Supported Polycom phones enable you to view and dial missed, placed, and received calls from the phone interface.
Directory-Service directory listing	Not Supported



<i>Feature</i>	<i>Supported / Not Supported</i>
Call Park	Not Supported
Call Group Pickup, Hunt group sequential, Hunt group Parallel	Not Supported
Busy Lamp Field/ Monitor (BLF)	Not Supported
Barge-In	Not Supported
Conveying microphone mute status between endpoints	Not Supported
<b>Provisioning and Management</b>	
Configuration file compatibility with CUCM	Not Supported Configuration requires use of Polycom configuration files, or you can apply parameters on a per phones basis using the Web Configuration Utility.
Server Redundancy	Not Supported
Digest Authentication	Supported
Phone Authentication	Supported
SNMP Support	Not Supported
Secure Real-Time Transport Protocol (SRTP)	Not supported
<b>Codec Support</b>	
G.711ulaw, G.722	Supported
<b>Unsupported CUCM Features</b>	
Presence and Buddy Lists	Not supported
Instant Messaging	Not supported
Cisco XML Applications	Not supported
Cisco Phone Directory	Not supported
Cisco ad-hoc conferencing	Not supported
Cisco TFTP software/configuration file	Not supported

## Before You Begin

Before deploying your SoundStation IP phones with CUCM, ensure that you obtain the proper licenses.

**Current Licensing** As of CUCM 8.0 and 7.1.5, each SoundStation IP connected to CUCM requires one Unified Workspace Licensing (UWL) Standard, or one User Connected Licensing (UCL) Enhanced. Note that you do not require Device User Licenses (DULs). Contact your Cisco representative to clarify your licensing questions.

**Legacy Licensing** When using a CUCM version prior to 8.0 or 7.1.5, each SoundStation IP using basic features that you connect to CUCM requires up to three DLUs. Each SoundStation IP phone using advanced features such as video or multiple lines requires six DLUs.

- For best audio experience on your SoundStation IP conference phones use codec G.722.

## Getting Help and Support Resources

This guide includes a [Getting Help](#) section where you can find links to Polycom product and support sites and partner sites. You can also find information about [The Polycom Community](#), which provides access to discussion forums you can use to discuss hardware, software, and partner solution topics with your colleagues. To register with the Polycom Community, you will need to create a Polycom online account.

The Polycom Community includes access to Polycom support personnel, as well as user-generated hardware, software, and partner solutions topics. You can view top blog posts and participate in threads on any number of recent topics.

# Setting Up Cisco Unified Communications Manager

The Cisco® Unified Communications Manager (CUCM) enables you to deploy and manage SoundStation IP series conference phones. Use this section to set up your CUCM environment for SoundStation IP conference phones. For information specific to SoundStation IP conference phones, see [SoundStation IP Series](#) on the Polycom voice support site.

## Cisco Unified Communications Manager

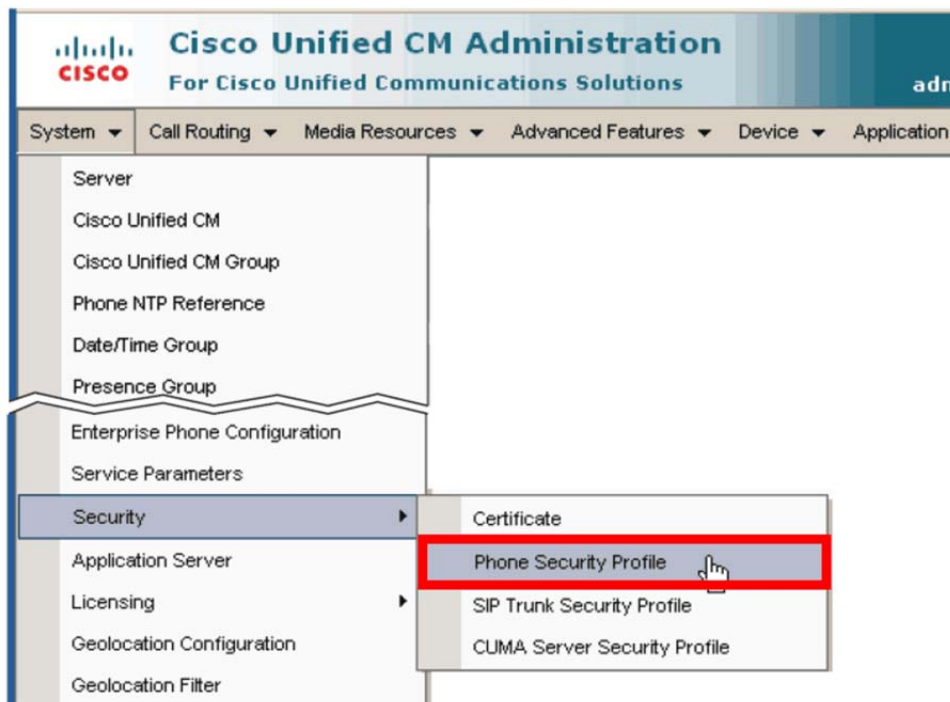
You must complete three procedures to set up Cisco Unified Communications Manager for SoundStation IP conference phones:

- Create phone security settings
- Create a user for each phone
- Add device information to the CUCM manager

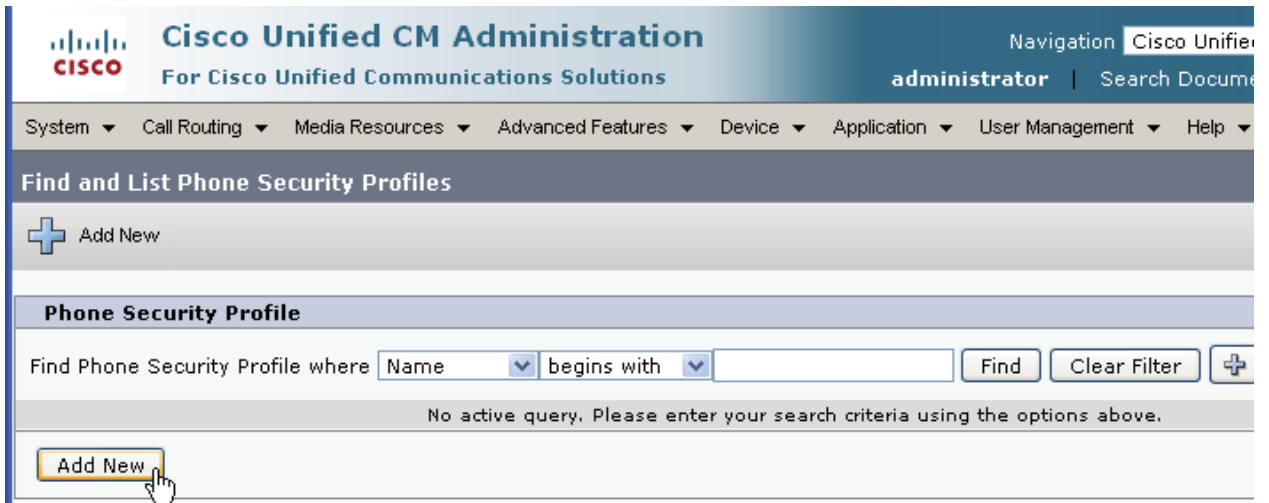
First, set conference phone security settings by creating phone security profiles.

### To create phone security profiles:

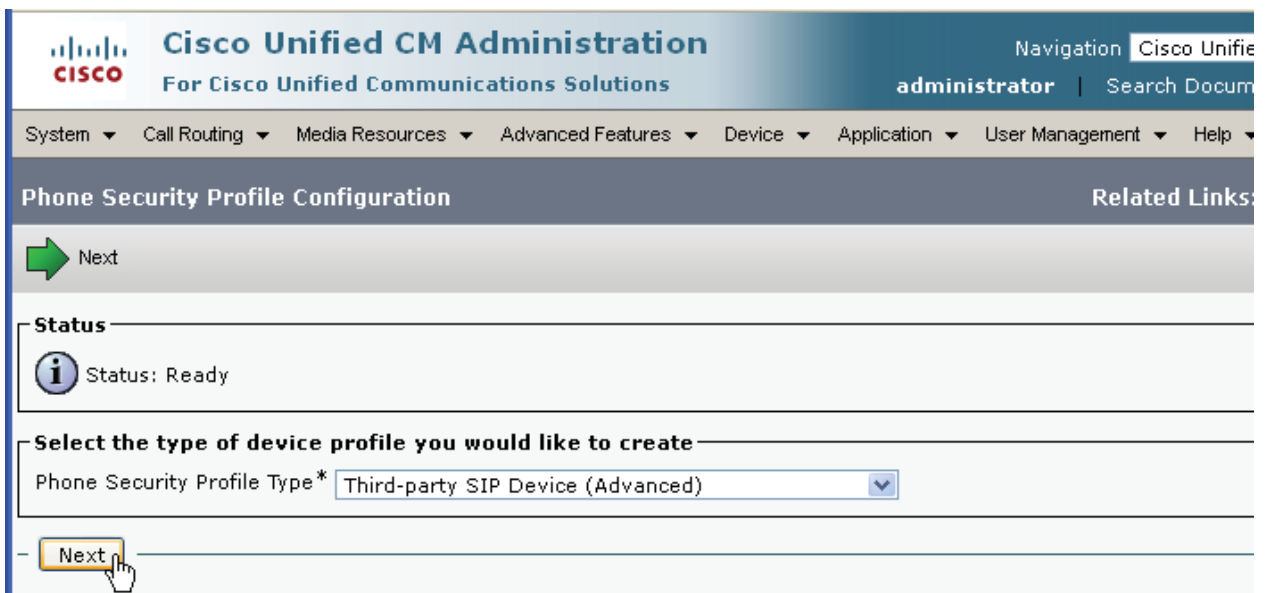
- 1 Open a Cisco Unified Communications Manager web administration session and enter your user name and password when prompted.
- 2 Select **System > Security > Phone Security Profile**.



3 Click **Add New**.



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



5 Under Phone Security Profile Information, complete the following fields.

- In **Name**, enter a profile name for your system
- (Optional) Enter a Description

➤ Check **Enable Digest Authentication**.

The screenshot shows the Cisco Unified CM Administration interface for configuring a Phone Security Profile. The page title is "Phone Security Profile Configuration". The status is "Ready". The profile information includes:

- Product Type:** Third-party SIP Device (Advanced)
- Device Protocol:** SIP
- Name\*:** SoundStation Duo
- Description:** SoundStation Duo Conference Phone
- Nonce Validity Time\*:** 600
- Transport Type\*:** TCP+UDP
- Enable Digest Authentication**

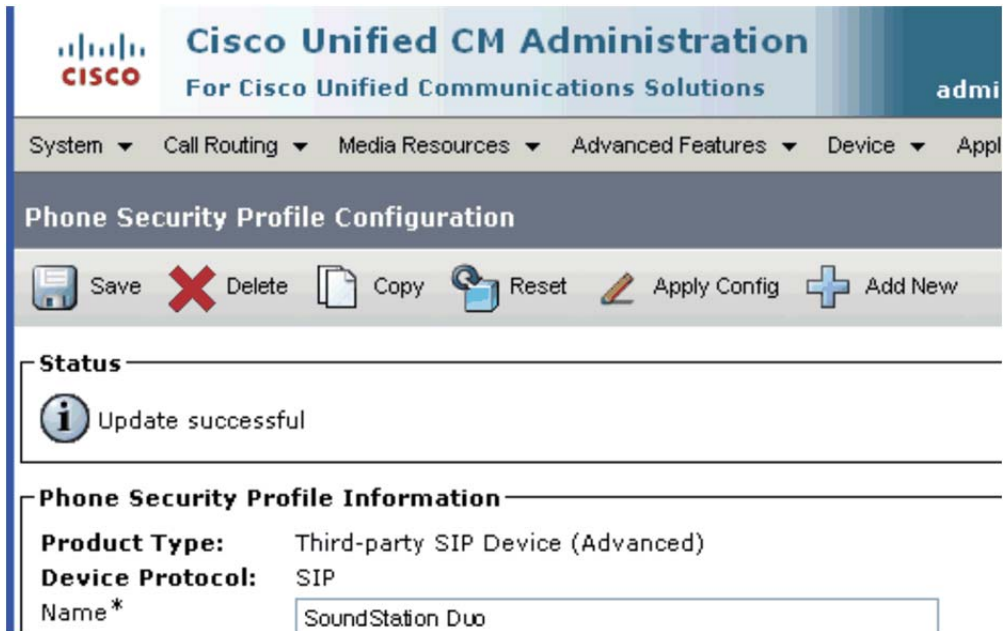
Parameters used in Phone:

- SIP Phone Port\*:** 5060

At the bottom, there is a note: **\*- indicates required item.**

6 Click **Save**.

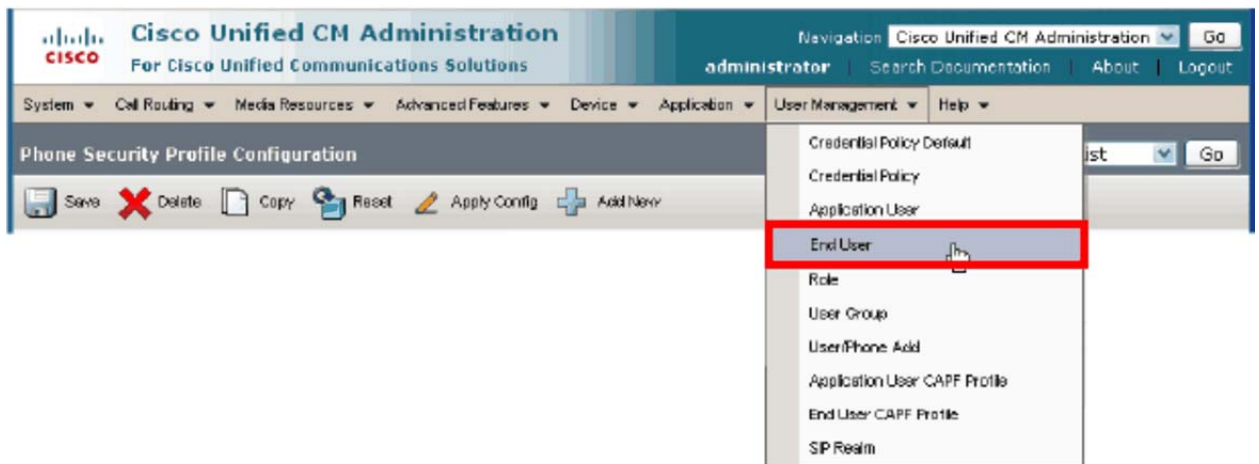
In the status bar near the top of the page, the message *Update Successful* displays, shown next.



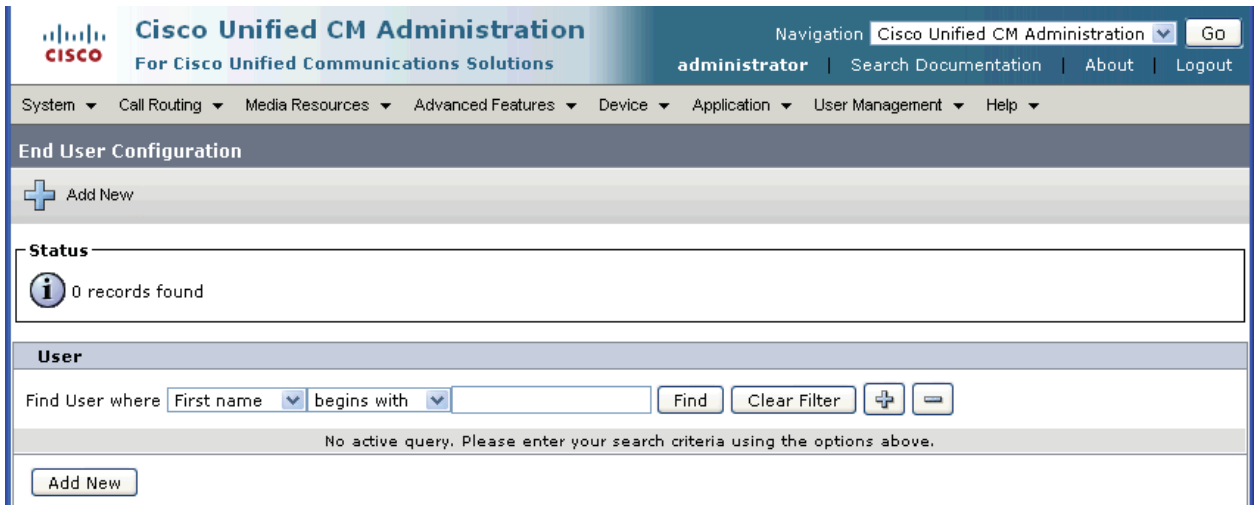
After you create phone security profiles, create a user for each SoundStation IP conference phone.

**To create a user:**

- 1 Select **User Management > End User**.



2 Click **Add New** as shown in the following figure.



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

**Status**

*i* Status: Ready

---

**User Information**

User ID*	sstvoipuser
Password	
Confirm Password	
PIN	
Confirm PIN	




**Troubleshooting: Adding a User with LDAP**

If you cannot add a user here, verify if your system is integrated with Lightweight Directory Access Protocol (LDAP). If so, use an existing user ID to associate the phone to an existing user, or create a new user ID for this phone. If your CUCM is integrated with an LDAP directory, you can add users using the LDAP directory itself.

- 4 In the **Last Name** field, enter a last name, shown next as LastName.

**Status**

 Status: Ready

---


**User Information**

User ID*	sstvoipuser
Password	
Confirm Password	
PIN	
Confirm PIN	
Last name*	LastName

- 5 In the **Digest Credentials** field and the **Confirm Digest Credentials** field, enter the digital credentials for the phone.

The 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.

**User Information**

User ID*	sstvoipuser
Password	
Confirm Password	
PIN	
Confirm PIN	
Last name*	LastName
Middle name	
First name	
Telephone Number	
Mail ID	
Manager User ID	
Department	
User Locale	< None > 
Associated PC	
Digest Credentials	●●●●●●●●
Confirm Digest Credentials	●●●●●●●●

- 6 Click **Save**.

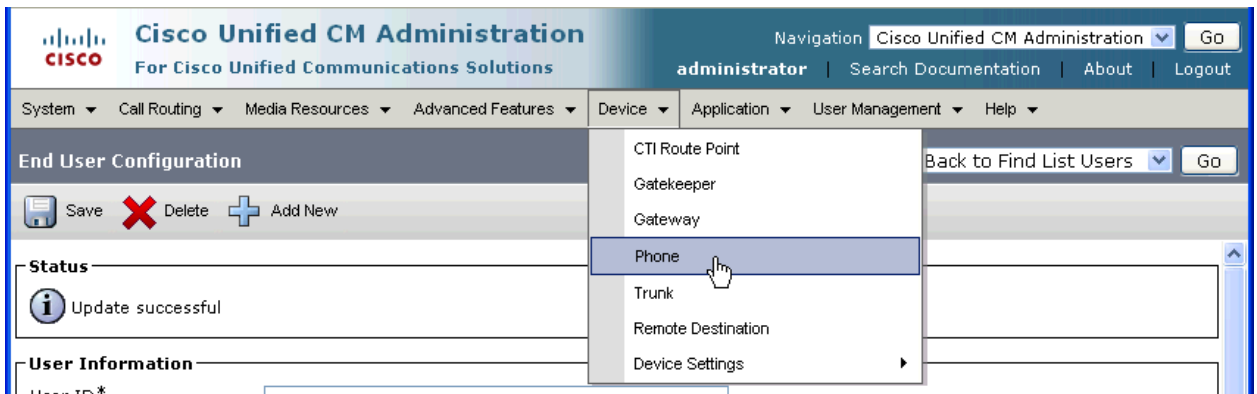
In the status bar near the top of the page, the message *Update Successful* displays.

After you create users, the next step is to add device information to CUCM.

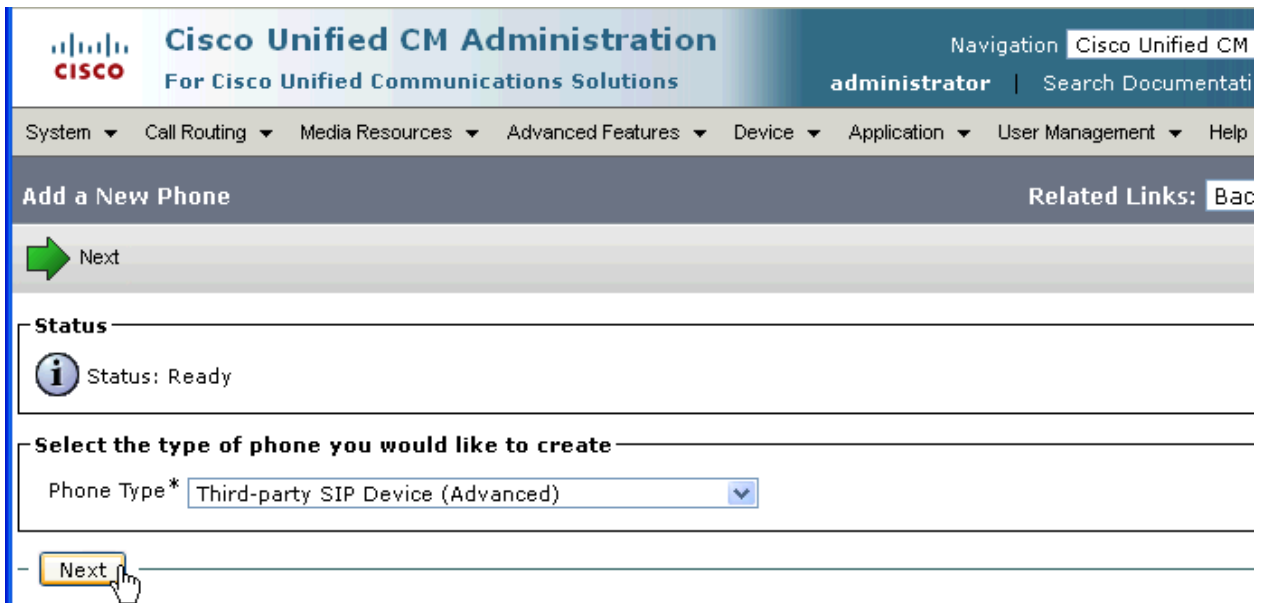


**To add the device information to the CUCM manager:**

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



- 2 Choose **Third-party SIP Device (Advanced)**, and click **Next**.



- 3 Enter the device information in fields shown on the Device Information screen. Many of the fields provide choices in a drop-down menu. Descriptions of the required fields are listed following the illustration.

Device Information	
⚠ Device is not trusted	
MAC Address*	0004F2BF001D
Description	SoundStation Duo
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 >
<input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS	
<input checked="" type="checkbox"/> Retry Video Call as Audio	
<input type="checkbox"/> Ignore Presentation Indicators (internal calls only)	
<input checked="" type="checkbox"/> Logged Into Hunt Group	
<input type="checkbox"/> Remote Device	

- In the **MAC Address** field, enter the MAC Address of the SoundStation conference phone. You can find the MAC address on a label on the bottom of the SoundStation IP conference phones. The MAC address is often referred to a serial number. MAC Address for a third-party SIP device is an arbitrary value for CUCM, however, it is recommended to use the phone's actual MAC address to ensure uniqueness and proper format.
- (Optional) In the **Description** field, enter a description.
- In **Device Pool**, choose the device pool you are using for your Cisco Unified Communications Manager system phones.
- In **Phone Button Template**, select **Third-party SIP Device (Advanced)**.
- (Optional) In **Calling Search Space**, select a calling search space for the phone.

- In **Location**, select a location for the phone.
- 4 Configure the following settings in Protocol Specific Information.

- In **Device Security Profile**, select the profile you created for your phone security settings in step 5 of the section [To create phone security profiles](#).
  - In **Rerouting Calling Search Space**, choose an option to enable call forwarding on the phone.
  - In **SIP Profile**, enter the SIP profile you want to use.
  - In **Digest User**, select the user you created in step 2 of the section [To Create a User](#). In this example, the user is `sstvoipuser`.
- 5 Click **Save**.

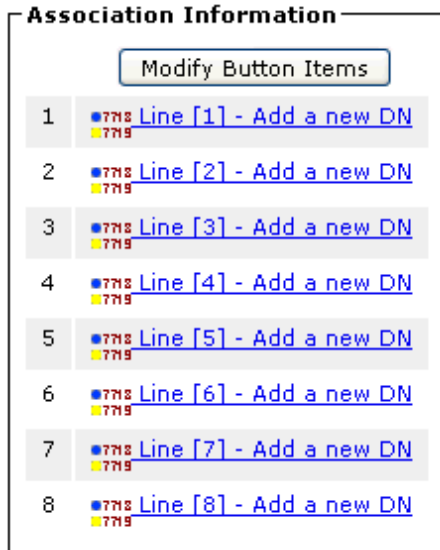
In the status bar near the top of the page, the message *Update Successful* displays.

- 6 Click **Apply Configuration**.

The following status message displays.

- 7 Click **OK** to continue.

- In the **Association Information** area on the left side of the window, add a new directory number (DN) by clicking on the **Line [1] — Add a new DN** link.



- The Directory Number Information screen displays.
- Enter the directory number information in fields shown in the Directory Number Information screen. Some of the fields provide choices in a drop-down menu. Descriptions of the required fields are listed following the illustration.

The screenshot shows a form titled "Directory Number Information". It contains the following fields:

- Directory Number\***: A text input field containing the value "4100041".
- Route Partition**: A drop-down menu showing "< None >".
- Description**: An empty text input field.
- Alerting Name**: A text input field containing the value "sstvoipuser".
- ASCII Alerting Name**: A text input field containing the value "sstvoipuser".

- In **Directory Number**, enter a phone extension. The following example uses extension 1234.
- In **Route Partition**, select a route partition.
- In **Alerting Name**, enter an alerting name. The example uses `sstvoipuser`.
- **ASCII Alerting Name** is automatically populated with the value you enter in Alerting Name. The example uses `sstvoipuser`.

**11 Set Voice Mail Profile** to the Cisco Unified Communications Manager system requirements. The following example shows the default settings.

**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 >	

**12 In the Call Forward and Call Pickup Settings screen, shown next, set values for your system.** This example shows the default screen and settings.

**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 >

- 13** In the Protocol Specific Information Screen, choose a **Rerouting Calling Search Space** value for your environment. In order for Call Forward All, Call Forward Busy, and Call Forward No Answer to work properly on a Polycom phone registered with CUCM, you must properly set the Rerouting Calling Search Space on the Device page.

**Protocol Specific Information**

Presence Group\* Standard Presence group

MTP Preferred Originating Codec\* 711ulaw

Device Security Profile\* 3rd party SIP Device Basic - Standard SIP Secure

**Rerouting Calling Search Space Unlimited**

SUBSCRIBE Calling Search Space < None >

SIP Profile\* Standard SIP Profile

Digest User sstvoipuser

Media Termination Point Required

Unattended Port

Require DTMF Reception

- 14** In **Display (Internal Caller ID)**, enter a caller ID. This example uses the caller ID Conference Room. The caller ID you enter here displays on the recipient's phone when a call is received from the SoundStation IP phone.

**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 >

- 15** In **ASCII Display (Internal Caller ID)**, enter a caller ID. This example uses the caller ID Conference Room.

**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\* 2

Busy Trigger\* 2 (Calls)

- In **Maximum Number of Calls**, enter a value for your environment?. Note that the SoundStation IP conference phones support a maximum of 24 calls.

- In **Busy Trigger**, enter a value for your environment. Busy Trigger defines the maximum number of simultaneous call appearances—active, busy, and on-hold calls—the device can support before additional calls receive a busy signal. Currently CUCM supports a maximum of two call appearances for third-party SIP devices.

**16** Click **Save**.

In the status bar near the top of the page, the message *Update Successful* displays.

You have successfully added device information to the CUCM manager.

This section has shown you how to complete three major procedures that set up your CUCM environment for SoundStation IP conference phones.

# Configuring a SoundStation IP Conference Phone with CUCM

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This section shows you how to configure settings that register SoundStation IP conference phones to the Cisco® Unified Communications Manager. Note that deployment environments differ and this guide cannot account for a particular deployment. You must complete the procedures in the section [Setting Up Cisco Unified Communications Manager](#) before setting up the SoundStation IP conference phones.

For more information on configuring the SoundStation IP conference phones, refer to the [Polycom UC Software 4.0.1 Administrators' Guide](#).

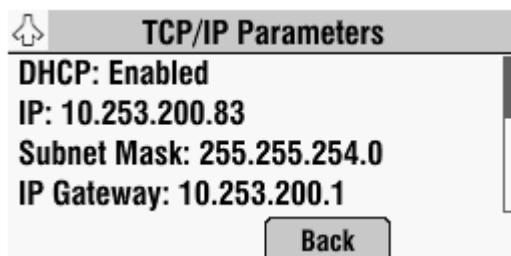
Polycom provides several ways to configure settings. This section shows you how to use the Polycom Web Configuration Utility to configure phone settings. The Web Configuration Utility is a web interface application that is particularly helpful when you are working remotely. You can use the Web Configuration Utility to provision one phone at a time.

## Configuring a SoundStation IP Conference Phone

This section shows you how to configure settings that register a SoundStation IP conference phone to Cisco Unified Communications Manager, and how to configure the phone's date and time settings. Note that illustrations of the Web Configuration Utility used in this section refer to the user interface available with UC software versions 4.0.4 and above. The user interface of the Web Configuration Utility when using UC software versions 3.x or earlier have a different user interface than shown in those shown here, but the corresponding parameter values in the earlier versions are available for configuring per the examples that follow.

### To log in to administrator settings on the Web Configuration Utility:

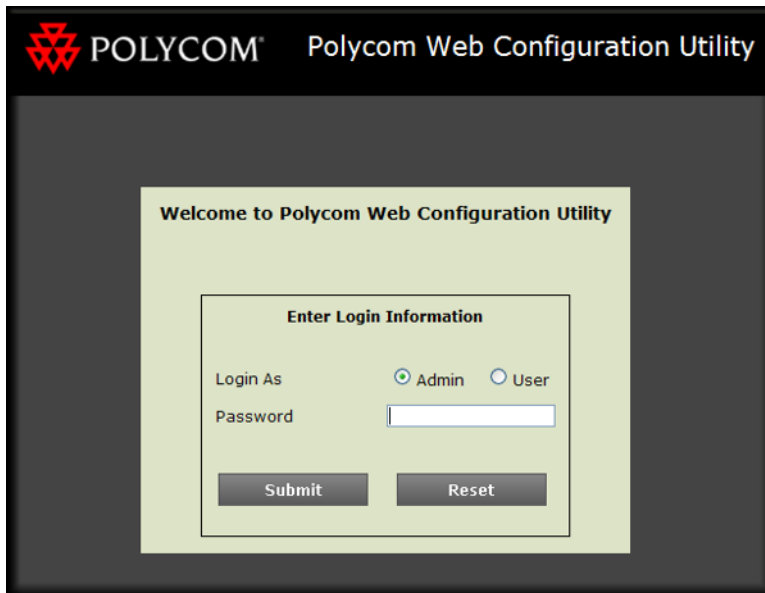
- 1 Obtain the IP address of your conference phone by navigating your phone's menu to **Menu > Status > Network > TCP/IP Parameters > IP:xxx . xxx . xxx . xxx .**





- 2 Enter the IP address to the address bar of a web browser on a computer connected to the same network as the conference phone, and press **Enter** on your keyboard.

The login screen displays, shown next.

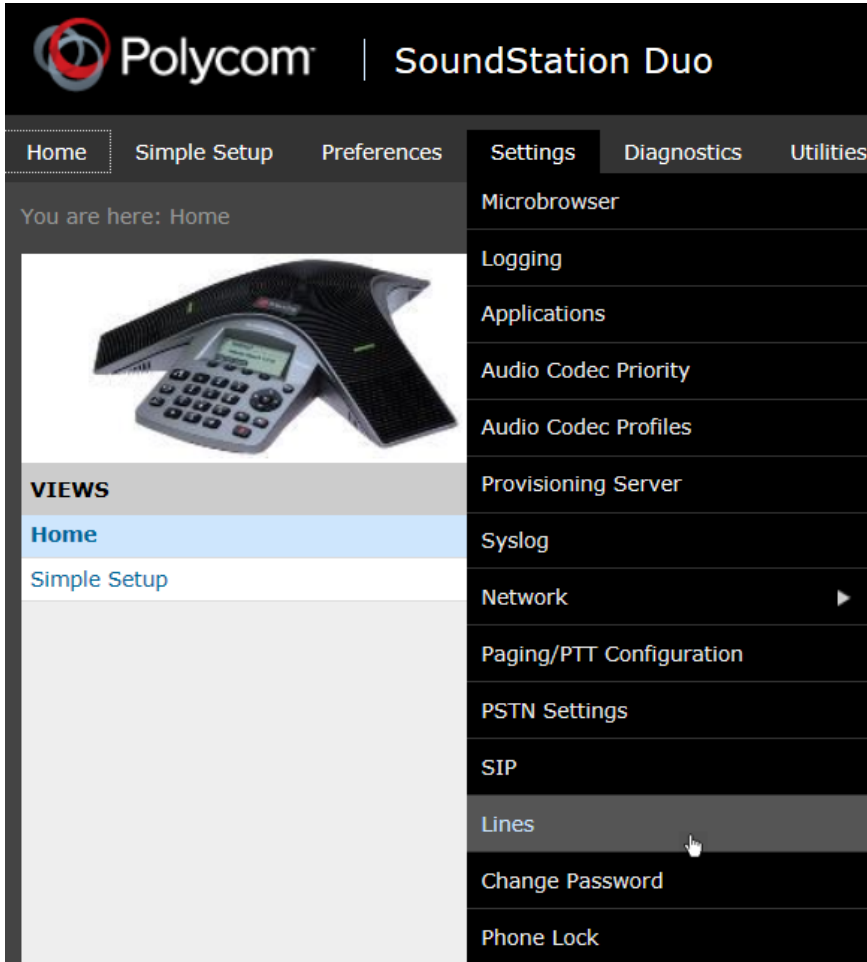


The screenshot shows the Polycom Web Configuration Utility login interface. At the top, the Polycom logo and the text "POLYCOM Polycom Web Configuration Utility" are displayed. Below this, a central box contains the text "Welcome to Polycom Web Configuration Utility". Underneath, a section titled "Enter Login Information" contains a "Login As" label with two radio button options: "Admin" (which is selected) and "User". Below the radio buttons is a "Password" label and an empty text input field. At the bottom of the form are two buttons: "Submit" and "Reset".

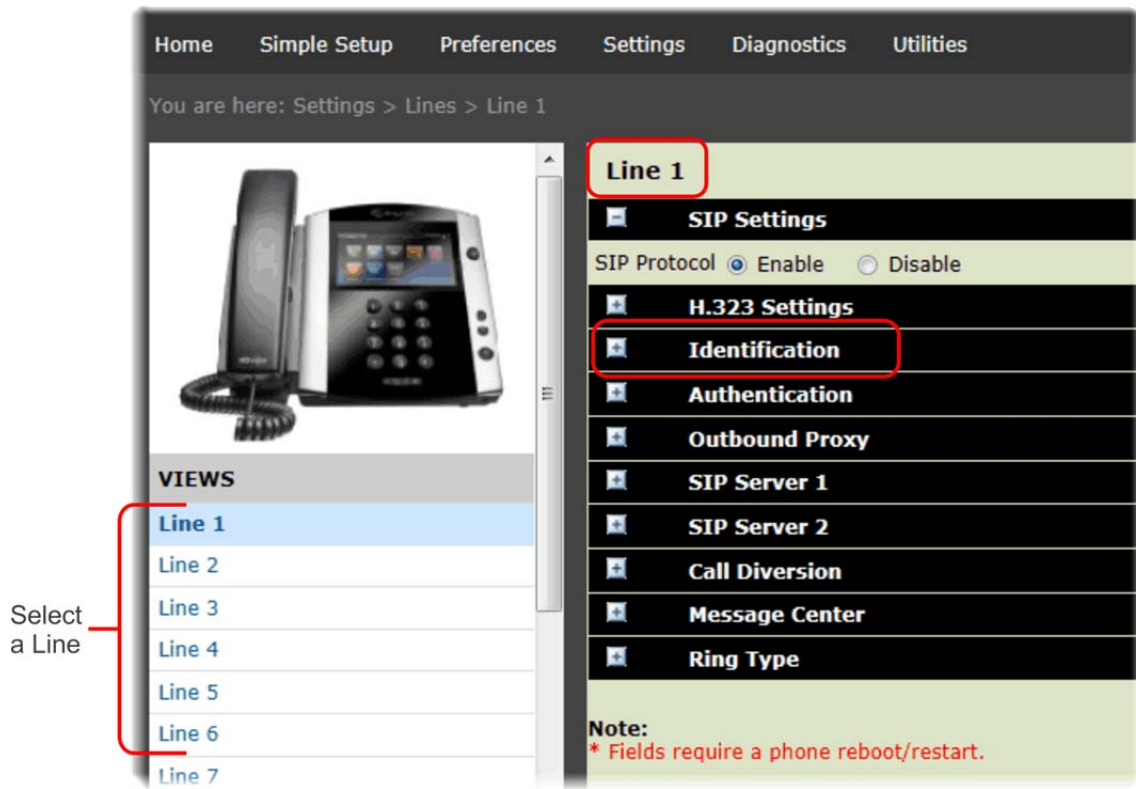
- 3 Log in to the Web Configuration Utility as an Administrator, use the default password 456, and click **Submit**.

**To configure line settings:**

- 1 Navigate to **Settings > Lines**.



- 2 Select the line you want to configure and expand Identification. Line 1 is elected by default.



- 3 Complete the following fields.

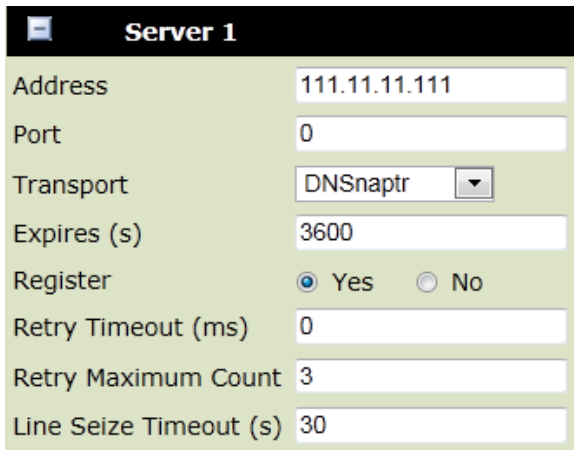
Line 1	
<b>Identification</b>	
Display Name	4100041
Address	4100041
Authentication User ID	sstvoipuser
Authentication Password	••••
Label	4100041
Type	<input checked="" type="radio"/> Private <input type="radio"/> Shared
Third Party Name	
Number of Line Keys	1
Calls Per Line	8
Ring Type	Low Trill

- Enter the **Display Name**. This example shows 4100041.
- Enter the **Address**. This example shows 4100041 to match Display Name.

- Enter the **Authentication User ID**. This example uses `sstvoipuser`.
- Enter the **Authentication Password**. This is the same value you entered in the **Digest Credentials** field when configuring digital credentials for the phone in CUCM.
- Enter the **Label** that displays on the phone. This example uses the phone extension number 4100041.

**To configure SIP server settings:**

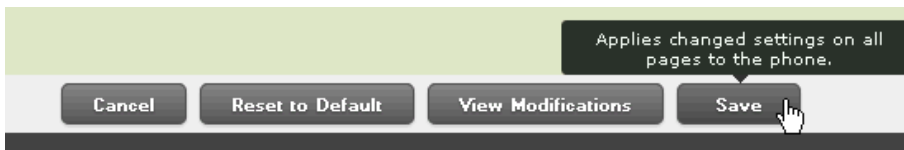
- 1 Expand **SIP Server 1**.




The screenshot shows a configuration form for 'Server 1'. The fields are: Address (111.11.11.111), Port (0), Transport (DNSnaptr), Expires (s) (3600), Register (Yes selected), Retry Timeout (ms) (0), Retry Maximum Count (3), and Line Seize Timeout (s) (30).

- In **Address**, enter the IP address or hostname of the Cisco Unified Communications Manager. In this example the CUCM has an IP address of `111.11.11.111`.
- Set **Port** to the correct port number for your environment.
- In **Transport**, choose a transport type for your environment.

- 2 Click **Save** to apply the settings.



- 3 Verify that the line has registered successfully on the conference phone display screen. When a line is successfully registered, a solid phone icon  displays on the phone screen.

## To configure data and time settings:

### 1 Navigate to **Preferences >Date & Time**

The screenshot shows the 'Date & Time' configuration page. It is divided into three sections: 'Display Format', 'Time Synchronization', and 'Daylight Savings'.  
- **Display Format:** 'Time Format' is set to '12 AM/PM' and 'Date Format' is set to 'Monday, January 1'.  
- **Time Synchronization:** 'SNTP Server' is 'north-america.pool.ntp.org', 'SNTP Resync Period (s)' is '86400', and 'Time Zone' is '(GMT -8:00) Pacific Time (US & Canada)'.  
- **Daylight Savings:** 'Daylight Savings' is 'Enable', 'Fixed Day' is 'Disable', 'Start Date' is 'Second Sunday March 02:00', and 'End Date' is 'First Sunday November 02:00'.

### 2 Change the following values:

- Under Display Format, choose a **Time Format** and **Date Format** that you want to display on the phone.
- Under Time Synchronization, choose an **SNTP server** in your region that the phone receives its time setting from.
- Select the appropriate **Time Zone**.
- Under Daylight Savings, enable or disable **Daylight Savings** time changes. When enabled, the phone's time settings automatically adjust to daylight savings time according to the settings you configure in Fixed Day, Start Date, and End Date.

# Troubleshooting the SoundStation IP

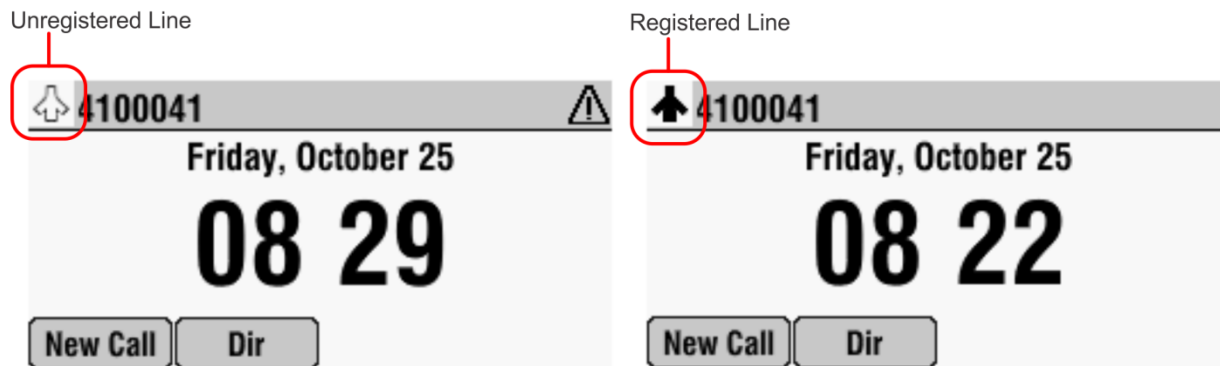
This chapter contains general troubleshooting information to help you solve problems you might encounter when using a SoundStation IP conference phones in a Cisco® Unified Communications Manager environment.

## Line Registration Issues

If you do not see the registered Line Icon on the SoundStation IP phone screen, confirm that the Authentication User ID and Authentication Password match the User ID and Digest Password you entered when you configured the Cisco Unified Communications Manager.

The following figure shows an unregistered and a registered line icon.

**Figure 1: Unregistered and Registered Line Icon**



If the credentials are correct but the SoundStation IP conference phones is still not registering, confirm the IP address or hostname of the CUCM.

If the SoundStation IP conference phone 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, CUCM shows Registration Unregistered.

**Figure 2: Phone Configuration Page of CUCM**

The screenshot shows the CUCM Phone Configuration page. At the top, the status is 'Ready'. Under 'Association Information', there are 8 lines listed, each with a 'Modify Button Items' button and a link to 'Add a new DN'. The 'Phone Type' section shows 'Product Type: Third-party SIP Device (Advanced)' and 'Device Protocol: SIP'. The 'Device Information' section is highlighted with a red box, showing 'Registration: Unregistered', 'IP Address: 10.253.200.44', 'Active Load ID: Unknown', 'Device is Active' (checked), 'Device is not trusted' (warning icon), 'MAC Address\*: 0004F217AC94', 'Description: SEP0004F217AC94', and 'Device Pool\*: Default'.

Once the phone is properly registered, the CUCM shows the device as registered.

**Figure 3: Device Registered with CUCM**

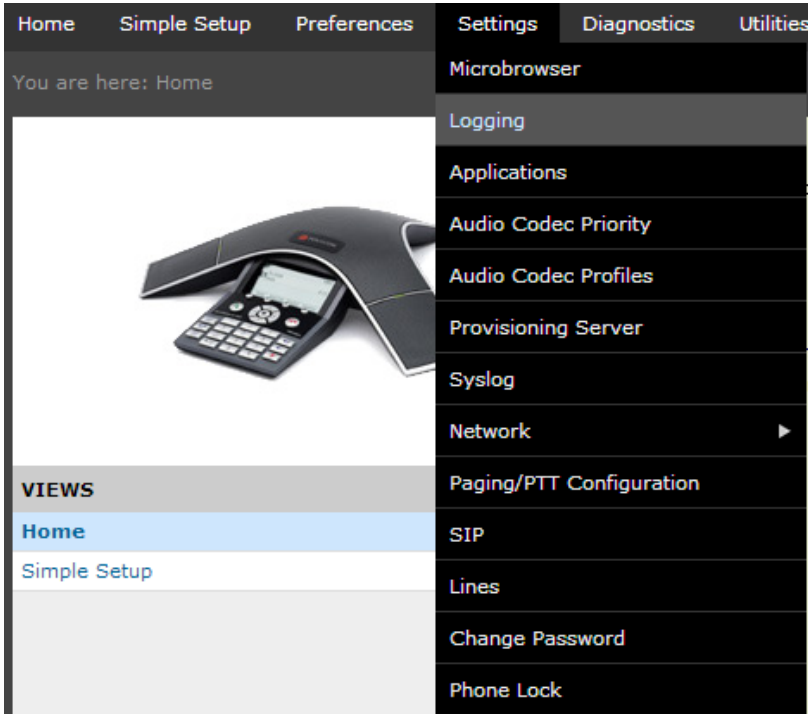
The screenshot shows the CUCM Phone Configuration page with the phone now registered. The 'Registration' status in the 'Device Information' section is highlighted with a red box and reads 'Registered with Cisco Unified Communications Manager'. Other details remain the same as in Figure 2: 'IP Address: 10.253.200.44', 'Active Load ID: Unknown', 'Device is Active' (checked), 'Device is not trusted' (warning icon), 'MAC Address\*: 0004F217AC94', 'Description: SEP0004F217AC94', and 'Device Pool\*: Default'.

# Logging

You can use the Polycom Web Configuration Utility to access phone log files.

**To access log files:**

- 1 Navigate to **Settings > Logging**.





**2 Navigate to Module Log Level Limits.**

**Logging**

- Global Settings
- Log File Upload
- Module Log Level Limits**

Application	Minor Error	LDAP	Minor Error	RAM Disk	Minor Error
ARES	Minor Error	License	Minor Error	Resource Finder	Minor Error
Buffer	Minor Error	LLDP	Minor Error	RTOS	Minor Error
Call Media Playback	Minor Error	Logging	Minor Error	Scheduled	Event 3
CDP	Minor Error	Micro Browser	Minor Error	Security	Minor Error
Configuration	Minor Error	Mobile	Minor Error	SIP	Event 3
Copy Utilities	Minor Error	Network	Minor Error	Srtp	Minor Error
CURL	Minor Error	Niche	Minor Error	SSH Client	Minor Error
DNS	Minor Error	OAI Protocol	Minor Error	SSPS	Minor Error
Dot1x	Minor Error	OCSP	Minor Error	Support Objects	Minor Error
EFK	Minor Error	PMT	Minor Error	Syslog	Minor Error
Ethernet Filter	Minor Error	Poll	Minor Error	TA	Fatal Error
HTTP Auth	Minor Error	Power Saving	Minor Error	TLS	Minor Error
HTTP Server	Minor Error	PPS	Minor Error	Util-Main	Minor Error
HTTP TA	Minor Error	Presence	Minor Error	Util-Trace	Minor Error
HW Desc	Minor Error	Presentation	Minor Error	Wapp Mgr	Minor Error
Idle Browser	Minor Error	PTT	Minor Error	Watch-dog	Minor Error
Key Observer	Minor Error	Push	Minor Error		


Cancel    Reset to Default    View Modifications    Save

**3 Change the SIP level to Event 3.**

**4 Navigate to Diagnostics > View & Download Logs.**

Home    Simple Setup    Preferences    Settings    **Diagnostics**    Utilities

You are here: Home    View & Download Logs



**Home**

- Phone Information
- Phone Model
- Part Number
- MAC Address
- IP Address
- UC Software Version
- BootROM Software Version

The following figure highlights a registration error (404 Error).

Figure 4: 404 Error

The screenshot shows a log viewer window titled "View & Download Logs". The "Log File Type" is set to "App" and "Log Level Filter" is set to "All". The log content shows SIP registration attempts. A 404 error is highlighted in yellow in the following log entry:

```
002318.702|sip|3|03|CStkDialog::SetAddressLocal new address added of 1
002318.702|sip|3|03|Reg UAC Response: code 404 new m_nExpire 79 m_nOverlap 0 ticks Trans 0x4186943c
002318.702|sip|3|03|SipStartFailOver 0
002357.995|sip|3|03|NoCall::TimeOut500ms 'Registering' m_nExpire == 0 RegisterCall -> Schedule Register listSize 0 lTimeout 0
002357.996|sip|3|03|CCallNoCall::NewCallState 'Register'-'>'Register' (0x418c310c)
002357.996|sip|3|03|CCallNoCall::NewCallState 'Register'-'>'Register' (0x418c310c)
002357.996|sip|3|03|RegClient:RegClient expire 66 overlap 0
002358.151|sip|3|03|UA Client Non-INVITE REGISTER trans state 'callingTrying'-'>'completed' by 401 resp 10 timeout(0x41867f7c)
002358.151|sip|3|03|401 challenge received
002358.180|sip|3|03|UA Client Non-INVITE REGISTER trans state 'callingTrying'-'>'proceeding' by 100 resp 6S timeout(0x41869dbc)
002358.180|sip|3|03|CTrans::InitNonInviteRetransSchedule retryCount 0
002358.374|sip|3|03|UA Client Non-INVITE REGISTER trans state 'proceeding'-'>'completed' by 200 resp 10 timeout(0x41869dbc)
002358.374|sip|3|03|NewRegisterState: 'Register' 'Registering' -> 'Registered' Expires 66 Overlap 0 for (0x418c310c)
002358.374|sip|3|03|CUser::OnRegistered Entry for call 0x418c310c with expires 240 ticks Transport 'UDP' inval Method 2 RROFO 0
002358.374|sip|3|03|SipOnEvRegistrarUpdate User 0, index 0, state 2, expire 120, working 1
002358.375|sip|3|03|SipOnEvProxyIPList 0,Total proxy 1
002358.375|appl|*|03|Ctx [0] Registered [true]
002358.375|sip|3|03|CStkDialog::SetAddressLocal localTag set to ''
```

# Getting Help

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For more information about installing, configuring, and administering Polycom products, refer to Documents and Downloads at [Polycom Support](#).

To find all Polycom partner solutions, see [Polycom Global Strategic Partner Solutions](#).

For more information on this and other Polycom partners, see [Polycom Global Strategic Partner Solutions](#).

The following related documents are available:

- [UC Software 5.0.0 Administrators' Guide](#), which shows you how to configure, customize, manage, and troubleshoot Polycom SoundPoint® IP, SoundStation® IP, and VVX business media phones.
  - For specific information on setting up a provisioning server see [Chapter 4: Setting Up the Provisioning Server](#).
  - For details on Polycom provisioning methods, see [Chapter 5: Configuration Methods](#).
- For workarounds to existing issues and expanded descriptions and examples see [Polycom Engineering Advisories and Technical Notifications](#).
- Release Notes for specific UC software releases are posted on the [Polycom UC Software Support Center](#).
- If you are familiar with Polycom provisioning methods and are looking for tips on configuring multiple phones, see [Provisioning with the Master Configuration File](#). This document is especially helpful if you are deploying multiple Polycom phone models.
- If you are updating to UC software 4.0 or later, you need to update to UC software 4.0.x using the [Polycom Upgrader 4.4.0B Utility](#). Before you download and install Polycom UC software version 4.0.0 or higher, Polycom strongly recommends that you review changes to the upgrade procedures detailed in the [Polycom UC Software 4.0.1 Administrators' Guide](#) and in [Polycom UC Software 4.0.x Upgrade and Downgrade Methods \(Engineering Advisory 64731\)](#).

# The Polycom Community

The [Polycom Community](#) gives you access to the latest developer and support information. Participate in discussion forums to share ideas and solve problems with your colleagues. To register with the Polycom Community, simply create a Polycom online account. When logged in, you can access Polycom support personnel and participate in developer and support forums to find the latest information on hardware, software, and partner solutions topics.

Learn, Share, Connect  
**The Polycom Community**

Community Home Register · Sign In · Help Contact Us

### Community Homepage

**Hello and Welcome to the Polycom Community!**  
We've created this community site so you can connect and interact with your colleagues and industry experts to exchange ideas, post questions, answers and share information. Come join the discussions! Happy Posting!

#### Support Community

- Voice
- PSTN
- VoIP
- SpectraLink
- DECT

#### Audio / Video

- Video Endpoints
- Telepresence
- Integrated Audio
- RealPresence Mobile

#### Developer Community

Click on one of the Forum links below to sign in or register and accept our SDK Agreement.

- Polycom Infrastructure Forum
- Polycom End Points Forum

#### Top Kudoed Posts

Re: Updated 4000 - now can't access?	2
Re: Updated 4000 - now can't access?	2
Re: Telepresence M100 not working	2
[FAQ] VoIP frequently asked questions	2
Re: Browser Environment error for RMX	1

[View All](#)

Use the following topics in the Polycom Community to find out more about deploying SoundStation IP conference phones.

**Topic** Using digit map features to resolve issues dialing number while off hook, or adding a second call to a conference:

- <http://community.polycom.com/t5/VoIP/FAQ-Unable-to-Dial-number-if-Off-Hook-or-on-2nd-Call-in-a/td-p/4233>

**Topic** Scripting tools that automate the creation of configuration files required to deploy large numbers of Polycom VoIP endpoints:

- <http://community.polycom.com/t5/Polycom-End-Points-Forum/SCRIPT-Mass-Deployment-Script/m-p/6009>
- <http://community.polycom.com/t5/Polycom-End-Points-Forum/SCRIPT-Automatic-Username-logon-file/m-p/6357>
- <http://community.polycom.com/t5/Polycom-End-Points-Forum/SCRIPT-Automatic-000000000000-directory-xml-from-a-CSV-File/m-p/7806>

**Topic** Modifying or removing soft keys:

- <http://community.polycom.com/t5/VoIP/FAQ-Using-Enhanced-Feature-Keys-EFK-macros-to-change-softkey/td-p/6544>
- <http://community.polycom.com/t5/VoIP/FAQ-How-can-I-limit-access-to-certain-menu-s/td-p/36382>

**Topic** Limiting access to phone menus:

- <http://community.polycom.com/t5/VoIP/FAQ-How-can-I-limit-access-to-certain-menu-s/td-p/36382>