DEPLOYMENT GUIDE

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Deploying Polycom® RealPresence Trio™ Solution, SoundStation® IP and Polycom® SoundStation® Duo Conference Phones with Cisco® Unified Communications Manager (CUCM)

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About This Guide

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

Name	*Icon*	*Description*
Note		The Note icon highlights information of interest or important information needed to be successful in accomplishing a procedure or to understand a concept.
Web Info		The Web Info icon highlights supplementary information available online such as documents or downloads on support.polycom.com or other locations.
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

Convention	*Description*
Bold	Highlights interface items such as menus, soft keys, file names, and directories. Also used to represent menu selections and text entry to the phone.
Italics	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.
Blue Text	Used for cross references to other sections within this document and for hyperlinks to external sites and documents.

Convention	*Description*
`Courier`	Used for code fragments and parameter names.

Get Started

This guide shows you how to deploy the Polycom® RealPresence Trio 8800 and Polycom® RealPresence Visual+ (together referred to as the RealPresence Trio solution), Polycom® SoundStation Duo, and 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. To illustrate registration steps, this guide uses 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, and more importantly, Polycom does not officially support anything earlier than CUCM 8.

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.

You can deploy the Polycom conference phones shown in the table Polycom Conference Phones as third-party devices with CUCM.

Polycom Conference Phones

<table>
<tr><th colspan="2">RealPresence Trio 8800
</th></tr>
<tr><td>SoundStation IP 5000
</td><td>SoundStation IP 6000
</td></tr>
<tr><td>SoundStation 7000
</td><td>SoundStation Duo
</td></tr>
</table>

Before You Begin

Before deploying your RealPresence Trio solution, SoundStation IP and SoundStation Duo devices (hereafter referred to as SoundStation IP conference phones) as third-party SIP devices with CUCM, ensure that you obtain the proper licenses. If you need to calculate the license units you require, see *Calculate CUCM License Units* in *Bulk Deployment of Polycom® SoundStation® IP and Polycom® SoundStation® Duo Conference Phones with Cisco Unified Communications Manager (CUCM)*.

Current Licensing As of CUCM 8.0, each SoundStation IP connected to CUCM requires one Unified Workspace Licensing (UWL) Standard, or one User Connected Licensing (UCL) Enhanced. You do not require Device User Licenses (DULs). Contact your Cisco representative to clarify your licensing questions. For CUCM 9 and higher, SoundStation IP uses a CUCM basic license. If registering the RealPresence Trio 8800 and RealPresence Visual+ to add video and content sharing with CUCM, a CUCM enhanced license is required.

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.

Settings: Use G.7222 code with SoundStation IP conference phones.

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

Frequently Asked Questions

Refer to the frequently asked questions (FAQs) to help answer questions you have about the solution before you begin.

What versions CUCM are tested and supported?

Polycom has tested and verified the RealPresence Trio solution, SoundStation Duo, and SoundStation IP conference phones with CUCM versions 8.6.x, 9.1.x and 10.5. Polycom has not tested or verified Polycom endpoints with any other Cisco call-control platforms including Cisco Unified Communications Manager Express and Cisco Business Edition 6000.

What models of Polycom SoundStation IP conference phones are compatible with CUCM?

The Polycom SoundStation IP 5000, 6000, 7000, and the SoundStation Duo running UC Software versions 4.0.4 and higher are compatible with CUCM. RealPresence Trio 8800 software version 5.4 is supported with CUCM 10.5 and higher.

What capabilities are supported?

See Supported Phone Features for a list of all supported and not supported features.

Are there important features that are not supported?

The following features are not supported:

- Busy lamp field, shared-line, call park, call group pickup, hunt group sequential, hunt group parallel, extension mobility, SRTP, Cisco phone directory (support is possible using LDAP), IM & Presence, Cisco XML Apps, Cisco Music on Hold, Cisco MeetMe. For a full list of supported and not supported features see Supported Phone Features.

Do Polycom SoundStation IP conference phones support Cisco Skinny Client Control Protocol (SCCP)?

Polycom IP phones do not use Cisco's proprietary SCCP. SoundStation IP phones are compliant with Internet Engineering Task Force (IETF) RFC 3261, SIP: Session Initiation Protocol and can be used with CUCM as third-party SIP devices.

Are there additional fees or licensing required on the Cisco platform?

There are no additional fees for third-party SIP devices on CUCM versions supported by Polycom. For additional information review licensing information in Before You Begin. If using CUCM 8.x, for more information on licensing, see *Calculate CUCM License Units* in *Bulk Deployment of Polycom® SoundStation® IP and Polycom® SoundStation® Duo Conference Phones with Cisco Unified Communications Manager (CUCM)*. If using CUCM 9.x and higher, audio-only devices use a CUCM basic license. If using the RealPresence Trio 8800 with RealPresence Trio Visual+ to add video and content capability, a CUCM enhanced license is required.

Is documentation available for bulk deployment of Polycom SoundStation phones in a CUCM environment?

CUCM offers features to support bulk deployment of third-party SIP endpoints. For detailed information on provisioning multiple SoundStation IP conference phones with CUCM, see *Bulk Deployment of Polycom® SoundStation® IP and Polycom® SoundStation® Duo Conference Phones with Cisco Unified Communications Manager (CUCM)*. Alternately, you can use Polycom RealPresence Resource Manager to auto-provision devices on CUCM (supported devices as of RPRM 9.0 will be the SoundStation IP, Duo and Trio 8800).

Get Help and Support Resources

This guide includes a Get 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.

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 only 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 the table Polycom Phones and UC Software. Use this table to match a phone with a compatible UC software release.

Polycom Phones and UC Software

Phone	*UC Software Release*
RealPresence Trio solution	5.4.0
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 only with Microsoft Lync Server. Do not use UC software 4.1.x with CUCM.

Supported Phone Features

The following table indicates which features the RealPresence Trio 8800 and SoundStation phones support when deployed with CUCM.

Note: RealPresence Trio solution video support

RealPresence Trio 8800 paired with RealPresence Trio Visual+ for video supports H.264 AVC up to 1080p, and if using CUCM 10.5 or higher, RealPresence Trio solution supports 1080p SVC as well.

CUCM Features on SoundStation Phones

Feature	*Supported / Not Supported*
Place and receive calls	Supported

Feature	*Supported / Not Supported*
On-hook dialing	Supported
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 provide conferencing from 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 device places a call on hold.
Message Waiting Indicator (MWI)	Supported
Additional Services	
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
Call park	Not Supported
Call group pickup, Hunt group sequential, Hunt group parallel	Not Supported

Feature	*Supported / Not Supported*
Busy Lamp Field (BLF) monitoring	Not Supported
Barge-In	Not Supported
Conveying microphone mute status between endpoints	Not Supported
Provisioning and Management	
Configuration file compatibility with CUCM	Not Supported Configuration requires the use of Polycom configuration files, or you can apply parameters on a per phone basis using the Web Configuration Utility.
Server redundancy	Supported with custom configuration on phone
Digest authentication	Supported
Phone authentication	Supported
SNMP support	Not Supported
Secure Real-Time Transport Protocol (SRTP)	Not supported
Codec Support	
G.711ulaw, G.722	Supported
Unsupported CUCM Features	
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

Set Up Cisco Unified Communications Manager

The Cisco® Unified Communications Manager (CUCM) enables you to deploy and register RealPresence Trio solution and SoundStation IP series conference phones. Use this section to set up a CUCM environment for your Trio or SoundStation conference phones. For information and documentation specific to SoundStation IP conference phones, see SoundStation IP Series on Polycom Voice Support. For information and documentation for RealPresence Trio solution, see RealPresence Trio on Polycom Support.

Web Info: Bulk Deployment of SoundStation Phones with CUCM.

CUCM offers features to support bulk deployment of third-party SIP endpoints. For detailed information on provisioning multiple SoundStation conference phones with CUCM, see *Bulk Deployment of Polycom® SoundStation® IP and Polycom® SoundStation® Duo Conference Phones with Cisco Unified Communications Manager (CUCM)*.

Cisco Unified Communications Manager

This section uses a SoundStation IP conference phone as an example. You must complete three procedures to set up Cisco Unified Communications Manager for SoundStation IP conference phones:

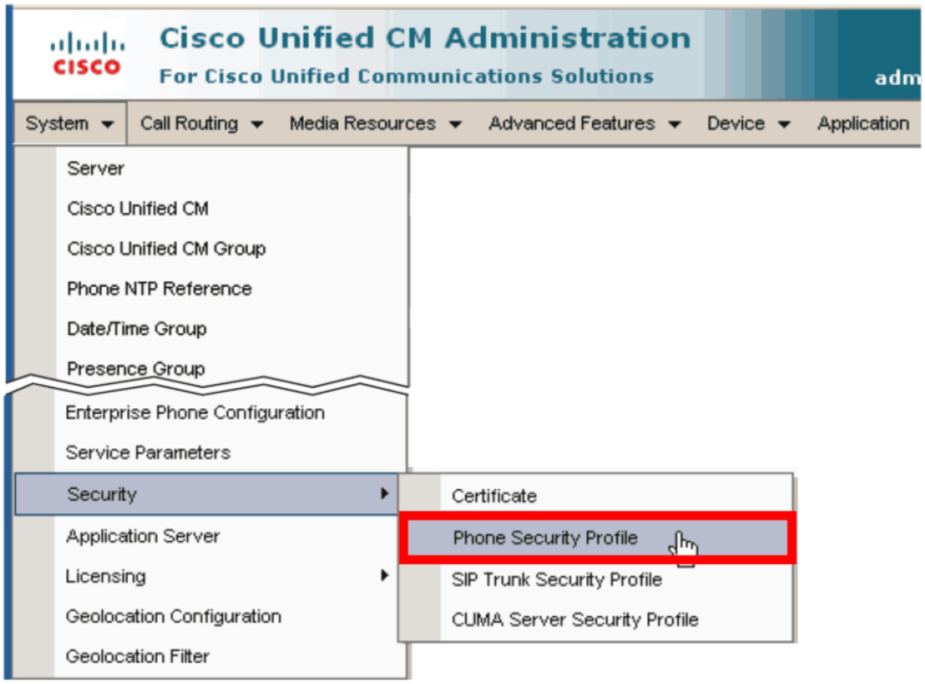
- Create phone security profiles
- Create a user for each phone
- Add device information to the CUCM manager

The first procedure, below, is to set conference phone security profiles (security profiles are optional for all Polycom devices). If security profiles are not used, the device authenticates with a username and extension only, no password.

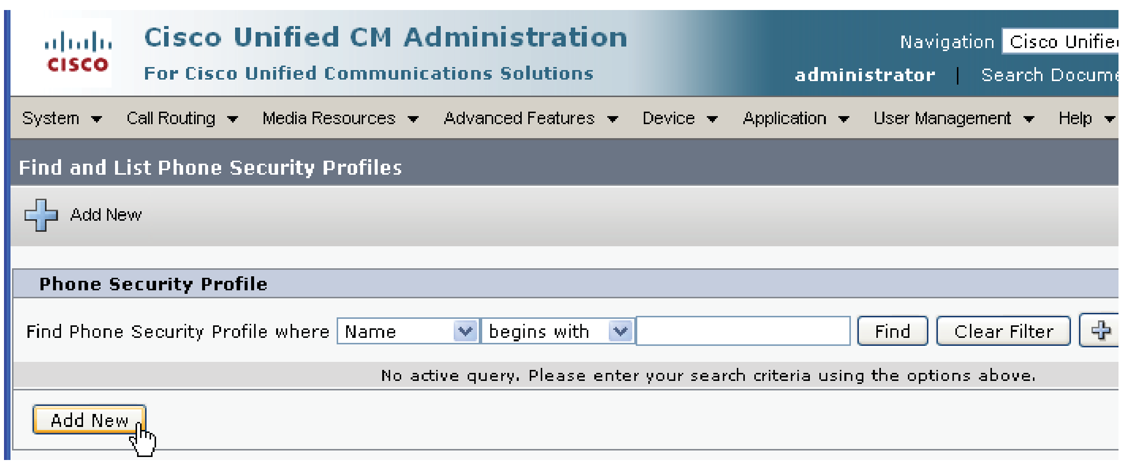
To create phone security profiles:

1. Open a Cisco Unified Communications Manager web administration session and enter your user name and password when prompted.

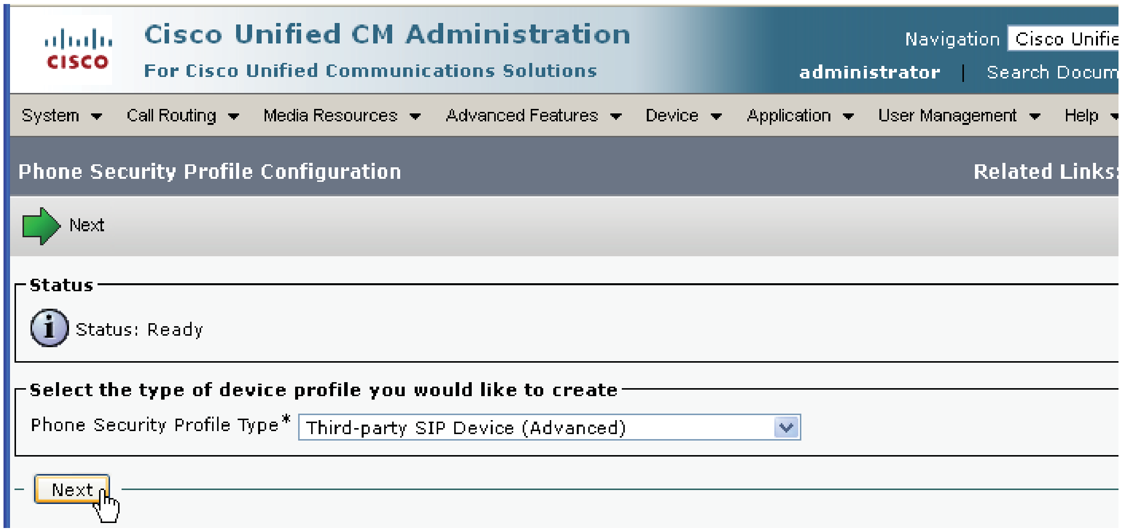
2 Select **System > Security > Phone Security Profile.**



3 Click **Add New**.

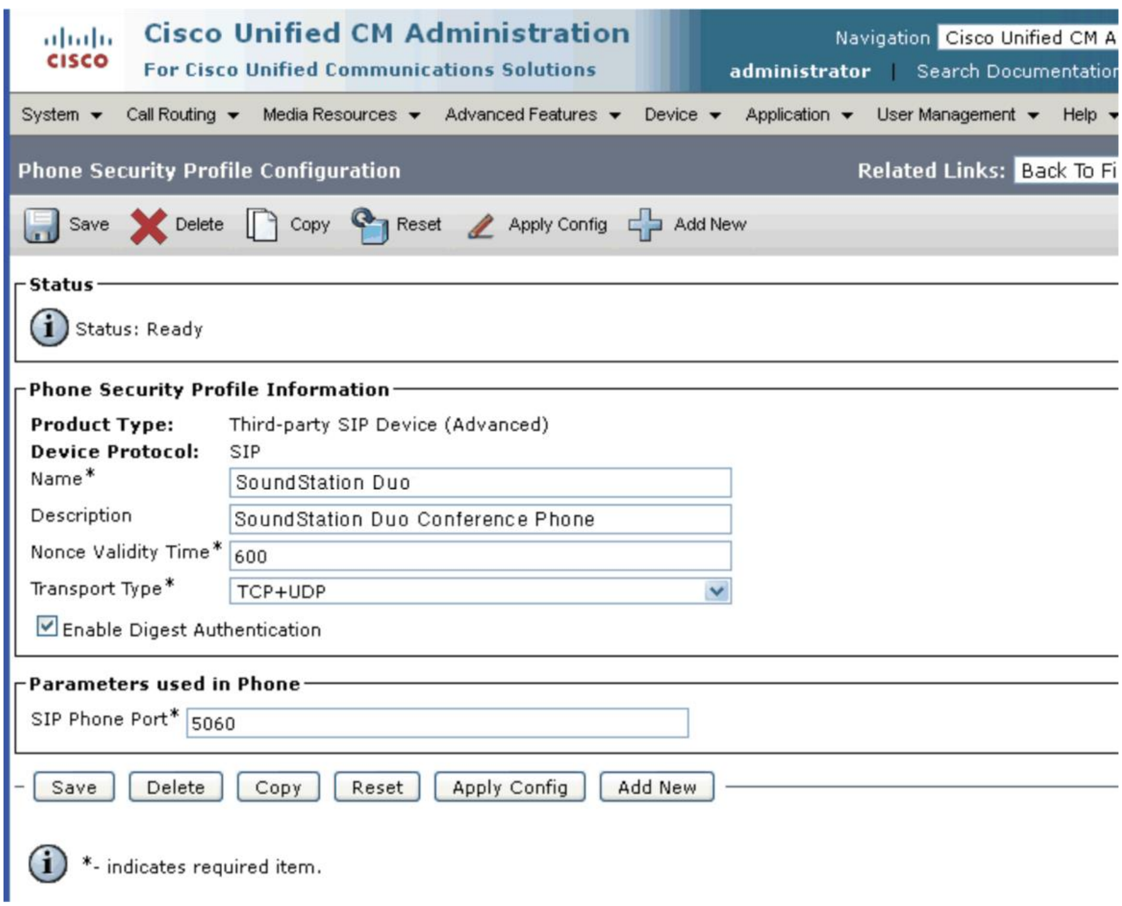


4 Choose Third-party **SIP Device (Advanced)** and click **Next**.



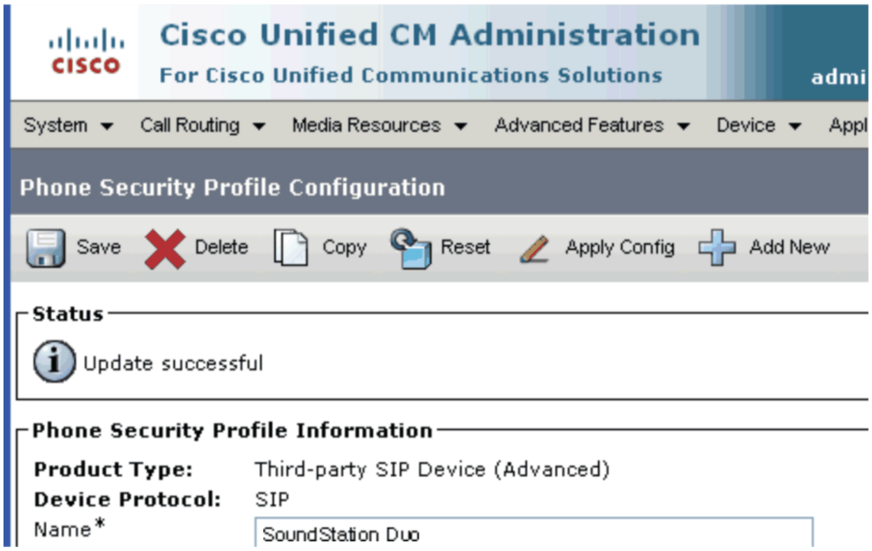
5 Under Phone Security Profile Information, complete the following fields.

- In **Name**, enter a profile name for your system
- (Optional) Enter a Description
- Check **Enable Digest Authentication**.



6 Click **Save**.

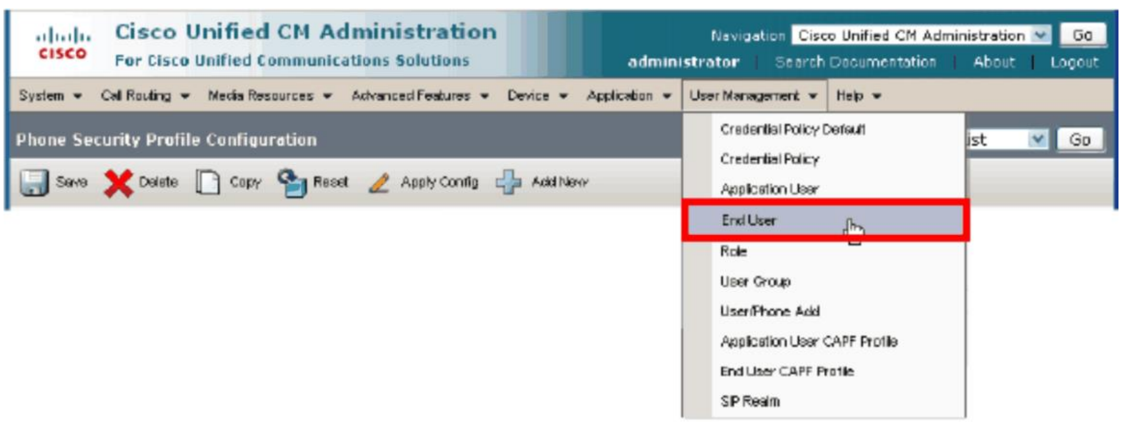
In the status bar near the top of the page, the message *Update Successful* displays, shown next.



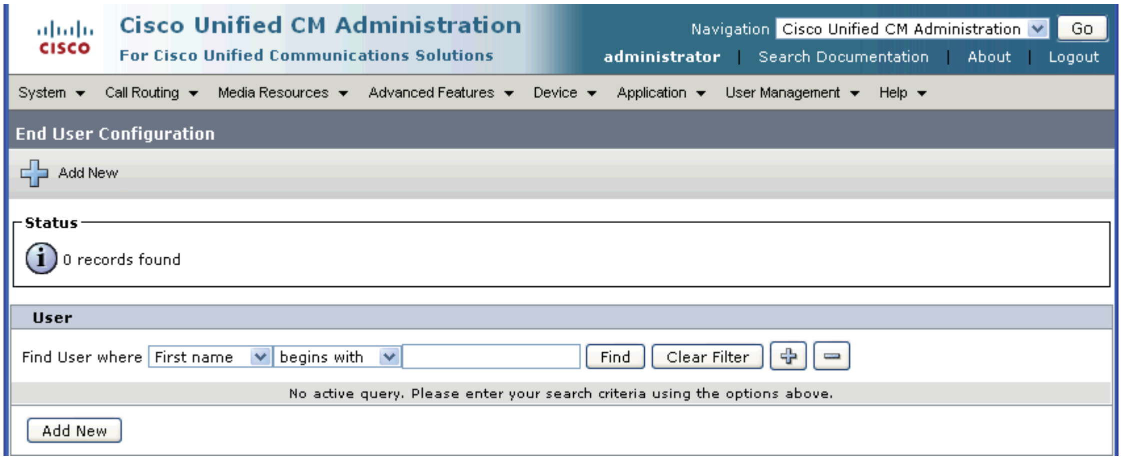
After you create phone security profiles, create a user for each SoundStation IP conference phone.

To create a user:

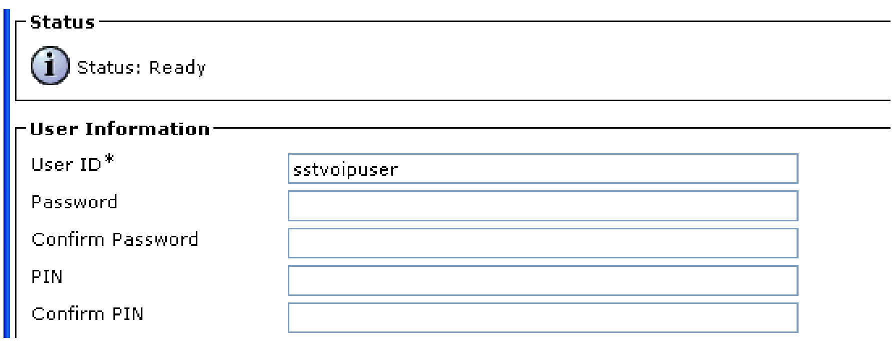
1 Select **User Management > End User.**



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



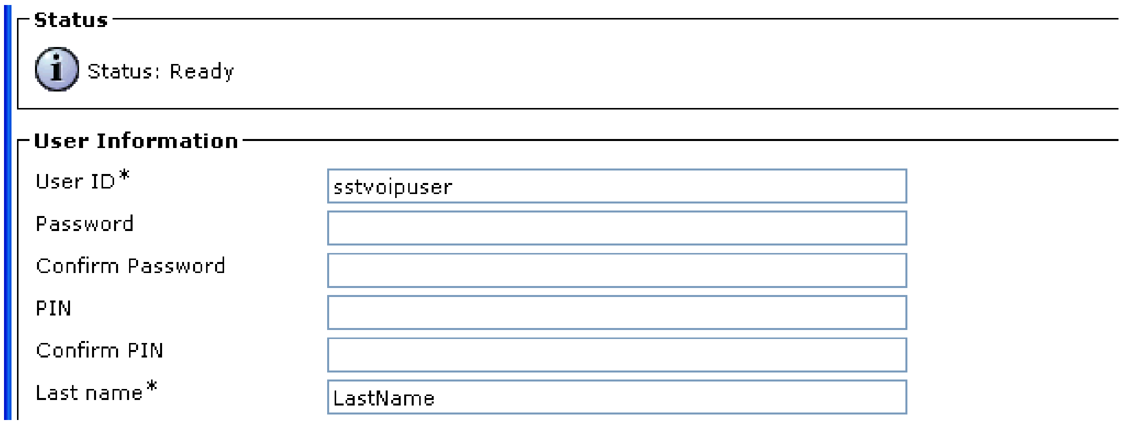
3 In the **User ID** text box, enter a user ID according to system and account policies. In this example, the user name is `sstvoipuser`.



Troubleshooting: Adding a User with LDAP

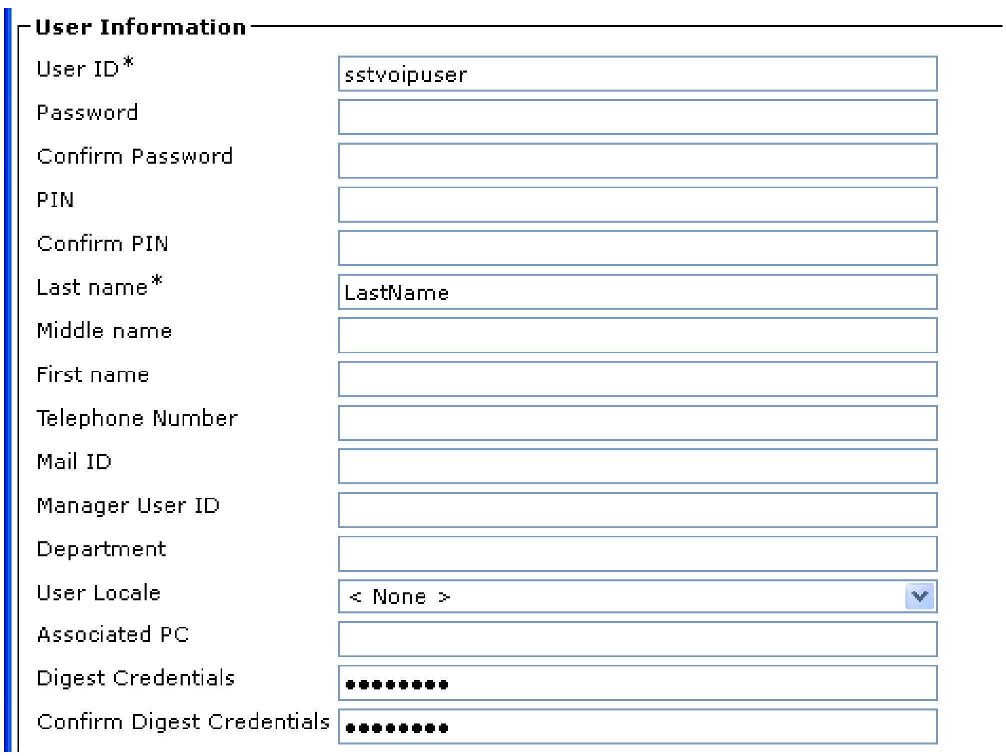
If you cannot add a user here, verify if your system is integrated with Lightweight Directory Access Protocol (LDAP). If so, use an existing user ID to associate the phone to an existing user, or create a new user ID for this phone. If your CUCM is integrated with an LDAP directory, you can add users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

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



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.



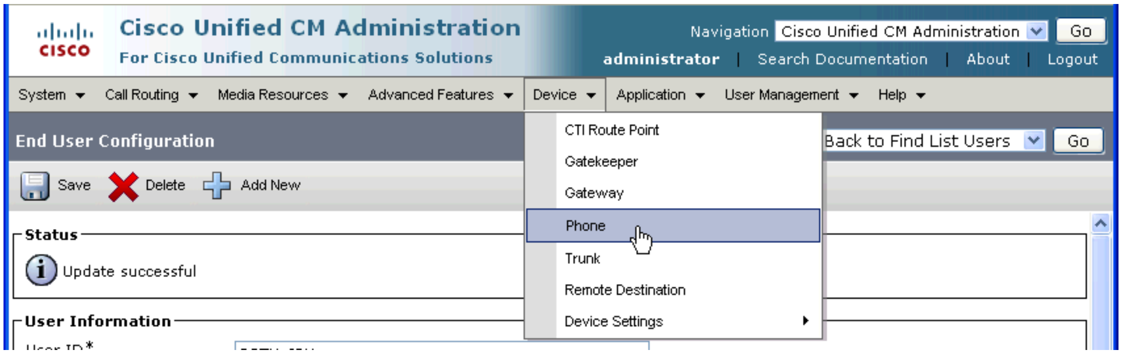
6 Click **Save**.

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

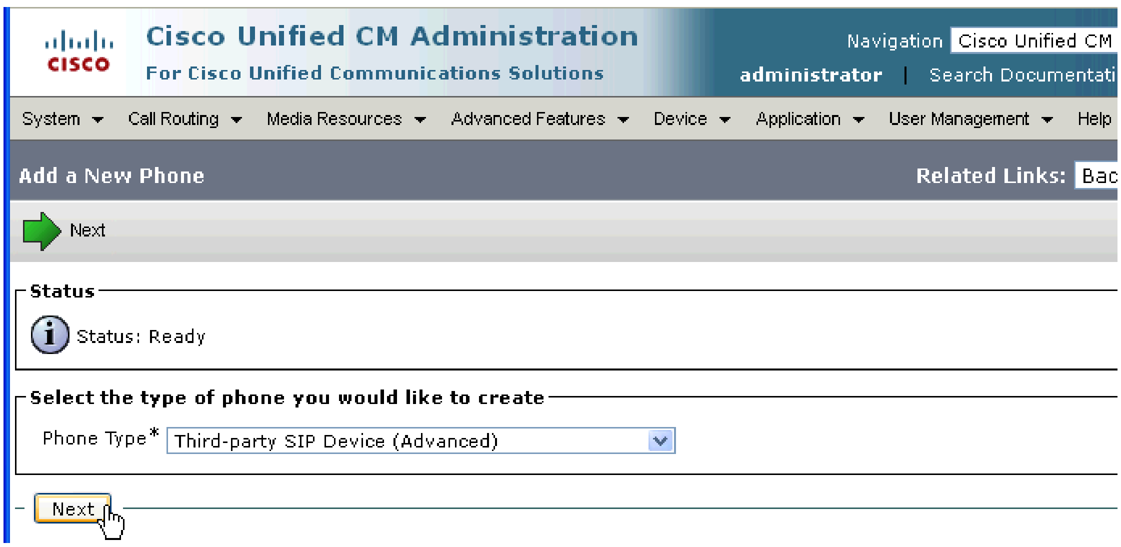
After you create users, the next step is to add device information to CUCM.

To add device information to the CUCM manager:

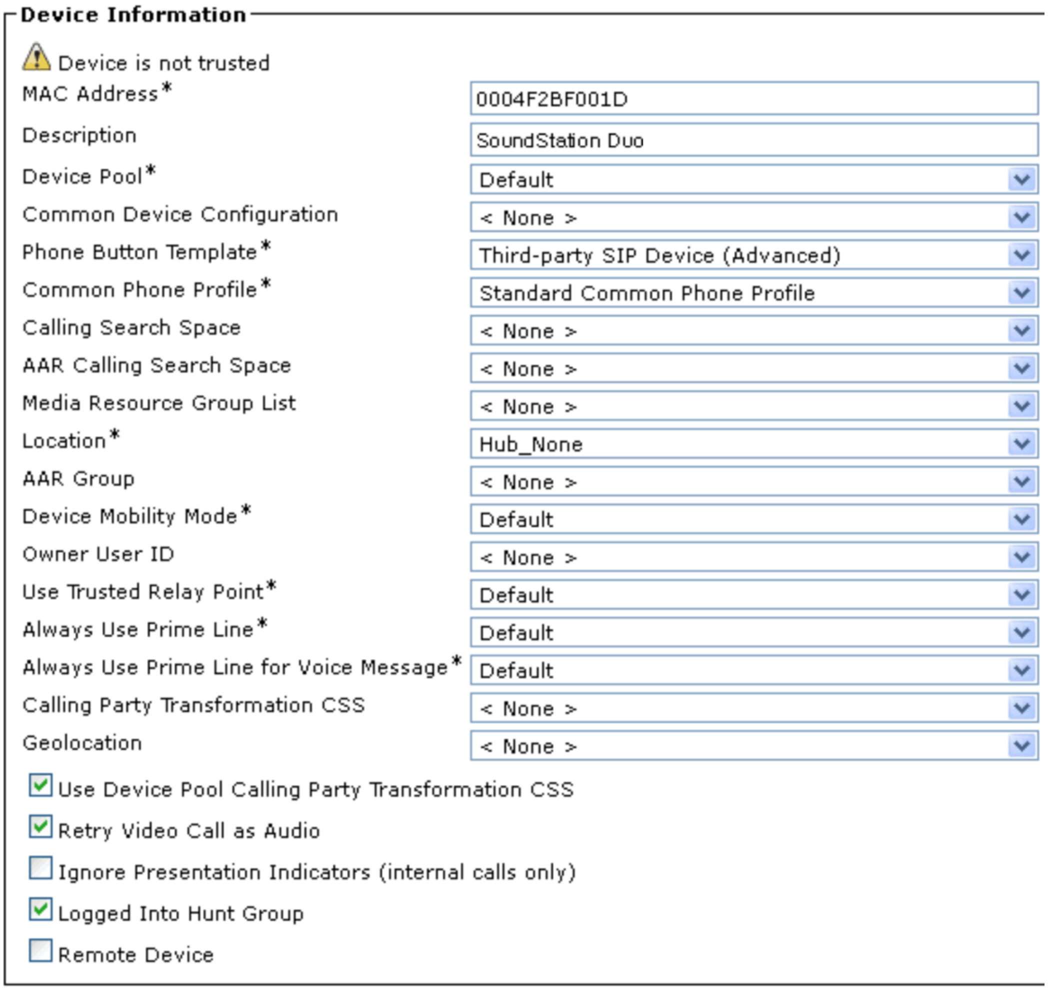
1 Select **Device > Phone** and click **Add New**.



2 In **Phone Type**, choose **Third-party SIP Device (Advanced) if using the Trio with the Visual+ and video calls**, choose **–Third-part SIP Device (Basic) if doing voice only** and click **next**.



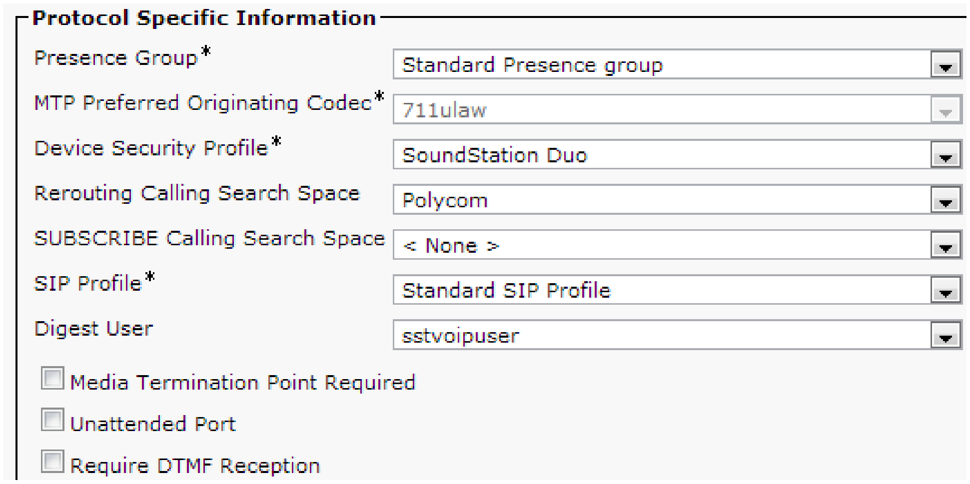
3 Enter the device information in fields shown on the Device Information screen. Many of the fields provide choices in a drop-down menu. Descriptions of the fields are listed following the illustration.



- 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. On CUCM, a MAC address is an arbitrary way to identify a third-party SIP device, however, Polycom recommends using the phone's MAC address to ensure you give each device a unique identifier and common 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) or Third-party SIP Device (Basic)** as appropriate.
- (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.



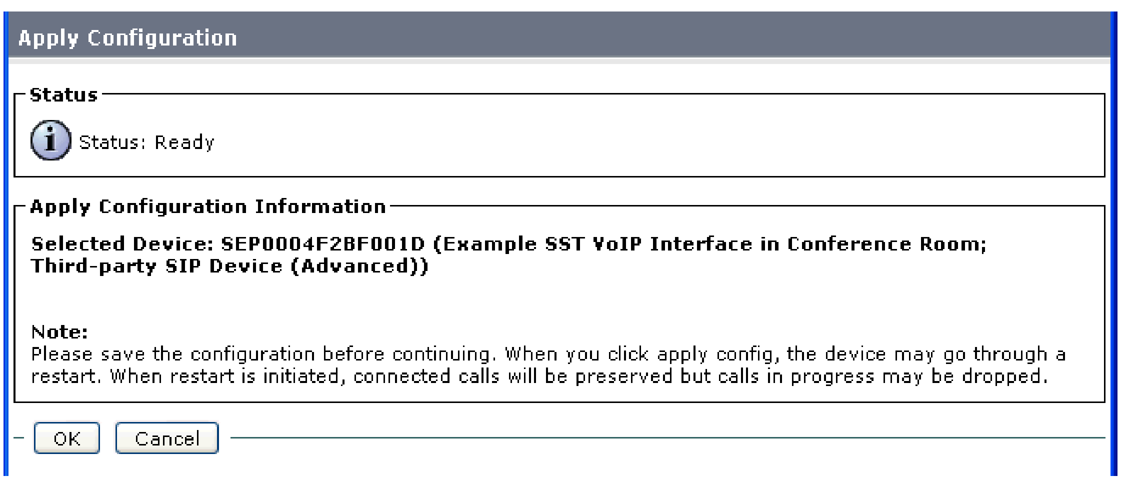
- In **Device Security Profile**, select the profile you created in step 5 of the procedure To create phone security profiles.
- In **Rerouting Calling Search Space**, choose an option to enable call forwarding on the phone.
- In **SIP Profile**, enter the SIP profile you want to use.
- In **Digest User**, select the user you created in step 2 of the procedure 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.

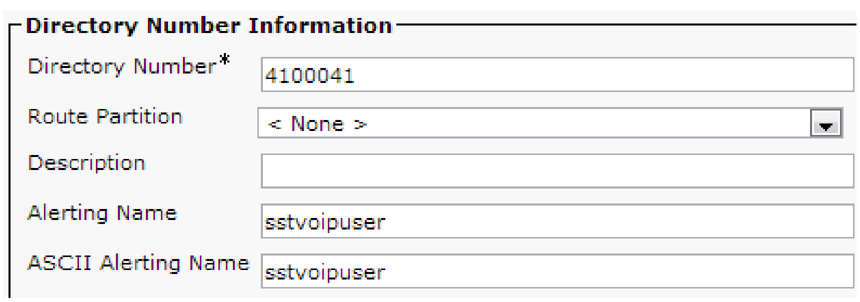


7 Click **OK** to continue.

8 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.

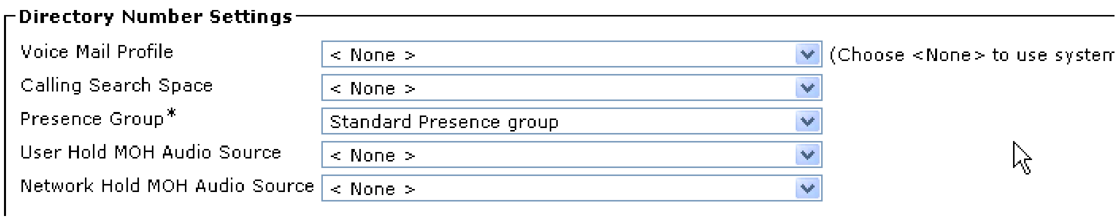
9 The Directory Number Information screen displays.

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

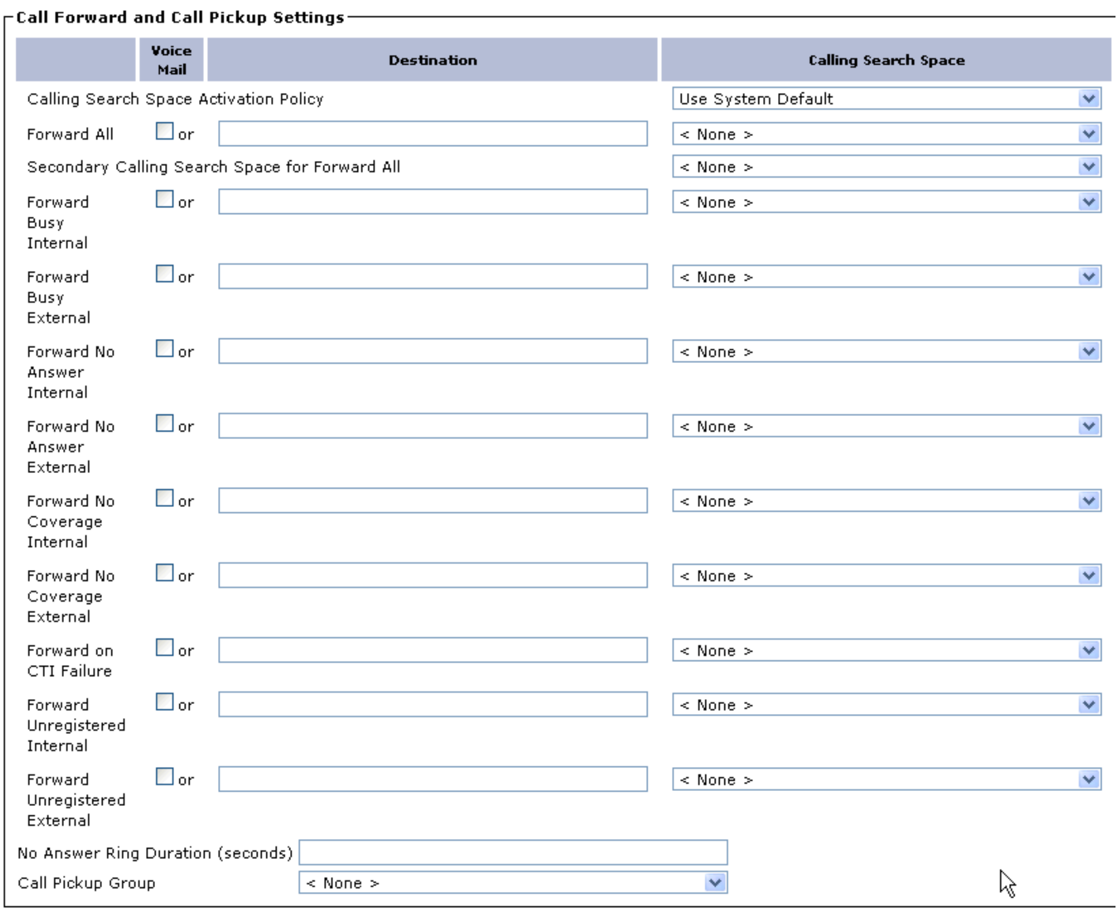


- In **Directory Number**, enter a phone extension. The following example uses extension `1234`.
- In **Route Partition**, select a route partition.
- In **Alerting Name**, enter an alerting name. The example uses `sstvoipuser`.
- **ASCII Alerting Name** is automatically populated with the value you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

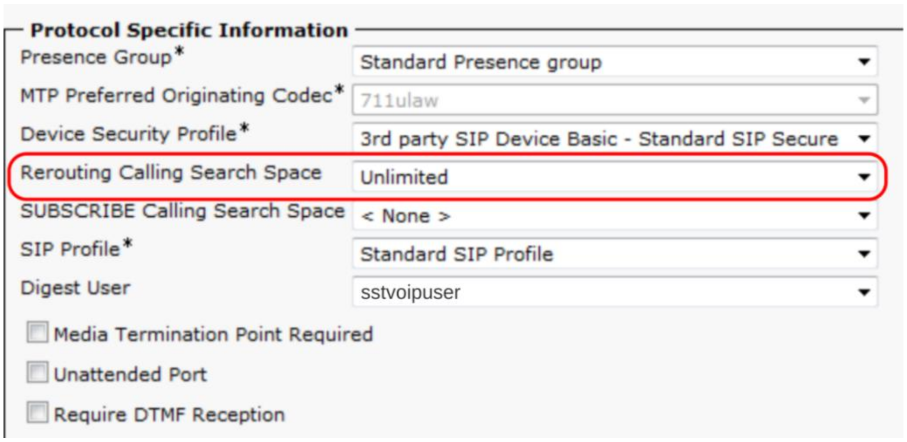
11 Set **Voice Mail Profile** to the Cisco Unified Communications Manager system requirements. The following example shows the default settings.



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.

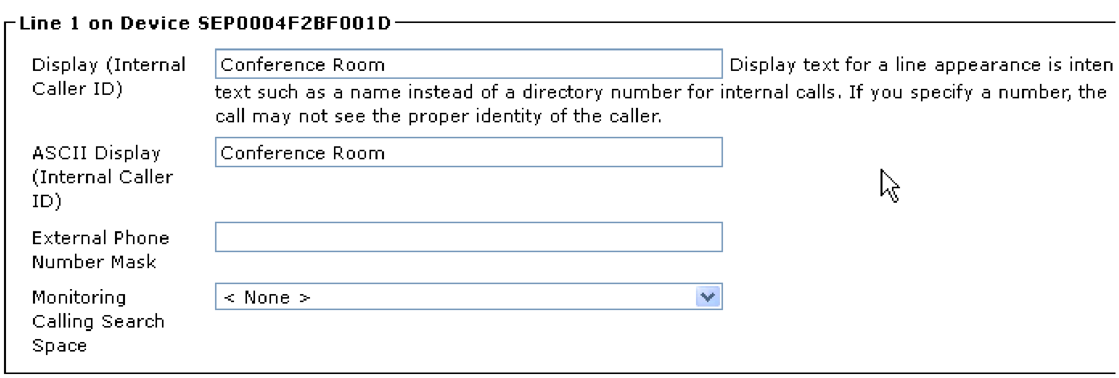


13 On the Protocol Specific Information screen, choose a **Rerouting Calling Search Space** value for your environment. In order for Call Forward All, Call Forward Busy, and Call Forward No Answer to work properly on a Polycom phone registered with CUCM, you must properly set the Rerouting Calling Search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



14 Enter the following information for Line 1 of your device.

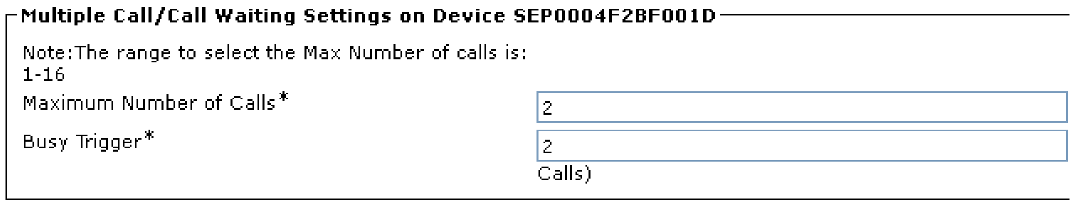
- In **Display (Internal Caller ID)**, enter a caller ID. This example uses the caller ID `Conference Room`. The caller ID you enter here displays on the recipient's phone when receiving a call from the SoundStation IP phone.
- In **ASCII Display (Internal Caller ID)**, enter a caller ID. This example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



15 Enter the following information in Multiple Call/Call Waiting Settings for your device.

- 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.

Configure RealPresence Trio Solution and SoundStation IP Conference Phones with CUCM

This section shows you how to configure settings that register the RealPresence Trio 8800 or SoundStation IP conference phones to the Cisco® Unified Communications Manager. You must complete the procedures in the section Set Up Cisco Unified Communications Manager before registering RealPresence Trio solution and SoundStation IP conference phones with CUCM. Note that deployment environments differ and this guide cannot account for a particular deployment.

For information on UC software and documentation for all Polycom voice products, see Voice Support on Polycom Support.

For documentation on RealPresence Trio solution, see RealPresence Trio on Polycom Support.

For information and documentation specific to SoundStation IP conference phones, see SoundStation IP Series on Polycom Support.

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.

Web Info: Bulk Deployment of SoundStation Phones with CUCM.

CUCM offers features to support bulk deployment of third-party SIP endpoints. For detailed information on provisioning multiple SoundStation conference phones with CUCM, see *Bulk Deployment of Polycom® SoundStation® IP and Polycom® SoundStation® Duo Conference Phones with Cisco Unified Communications Manager (CUCM)*.

Configure the Phone Using the Web Configuration Utility

This section shows you how to use the Polycom Web Configuration utility to configure settings that register a RealPresence Trio 8800 and SoundStation IP conference phone with Cisco Unified Communications Manager. Illustrations of the Web Configuration Utility in this section refer to the user interface available with UC software versions 4.0.4 and above, and use a SoundStation phones as an example. The user interface of the Web Configuration Utility for UC software versions 3.x or earlier have a different user interface than those shown in this section, however, parameter values in earlier software versions are available.

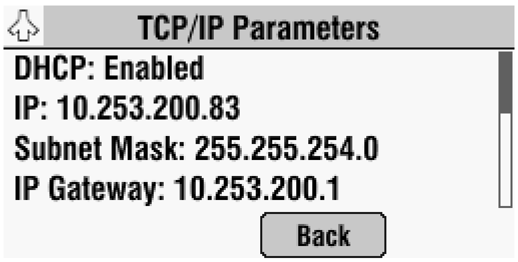
This section includes the following procedures:

- Log into a phone's Web Configuration Utility as an administrator

- Configure line settings
- Configure SIP server settings
- Configure date and time settings

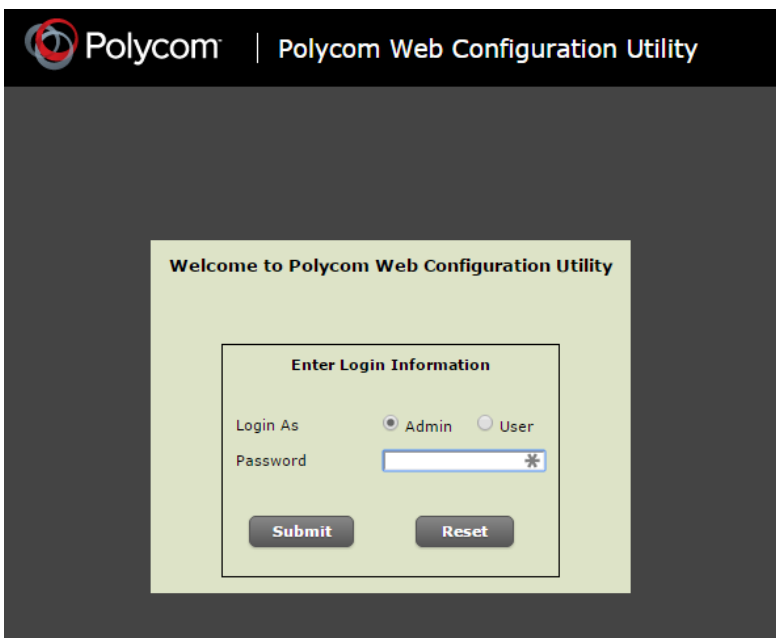
To log into the Web Configuration Utility as an administrator:

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.

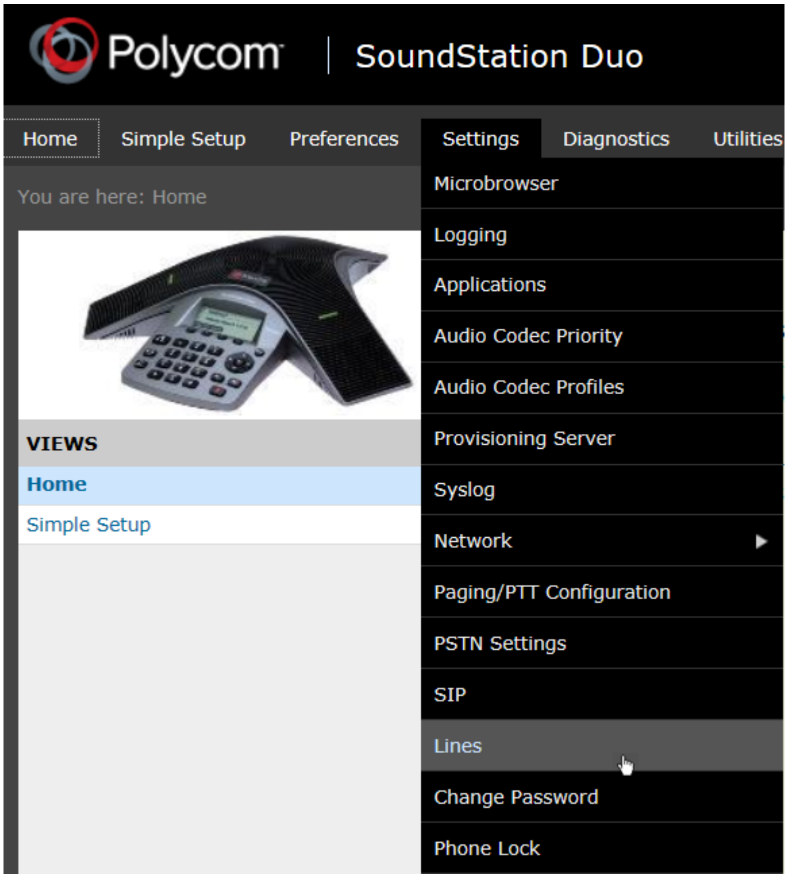
The login screen displays, shown next.



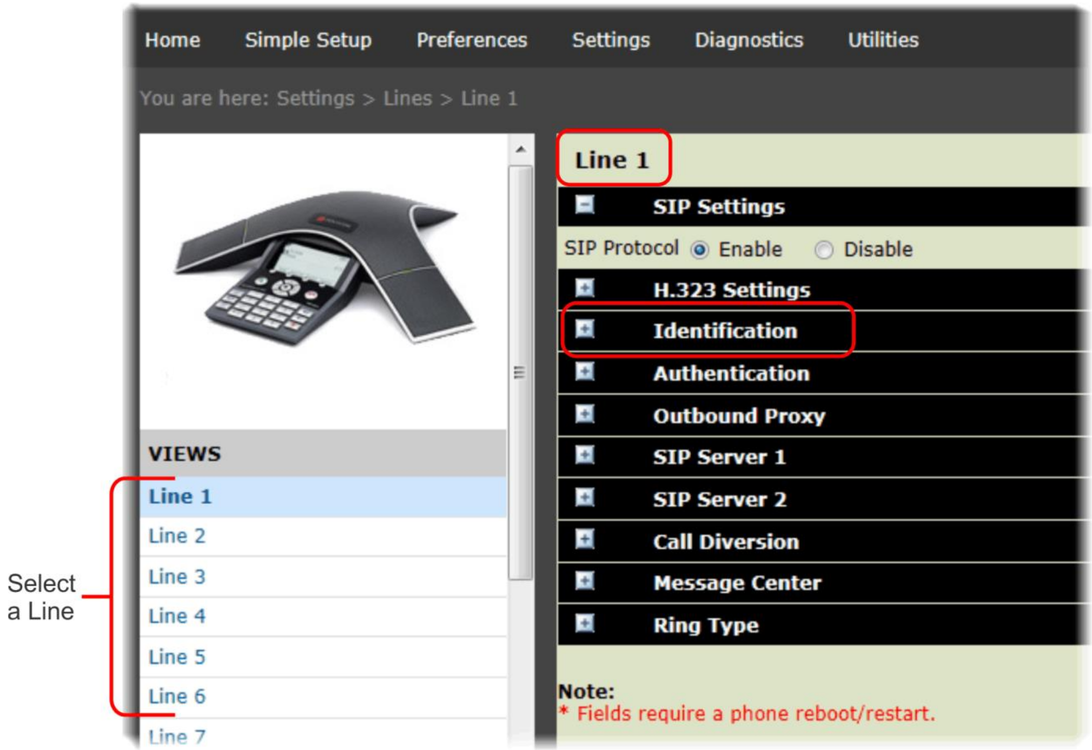
3 Log in to the Web Configuration Utility as an Administrator, use the default password 456, and click **Submit**.

To configure line settings:

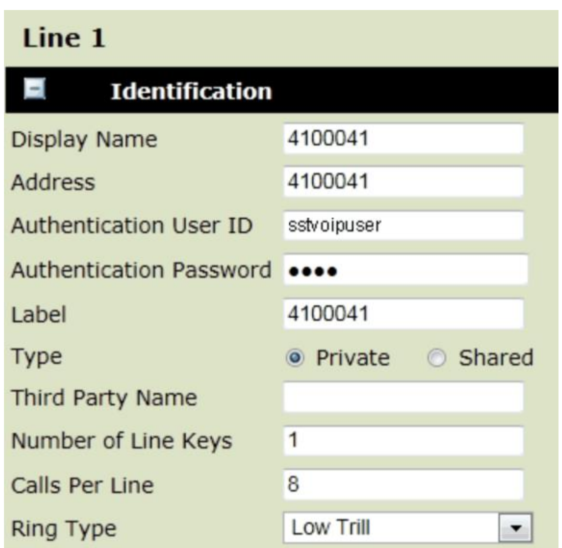
1 Navigate to **Settings** > **Lines**.



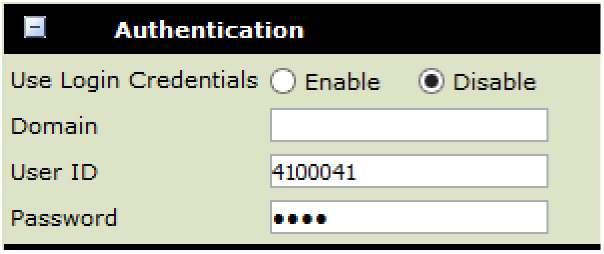
2 Select the line you want to configure and expand the **Identification** menu. Line 1 is selected by default.



3 Complete the following fields.



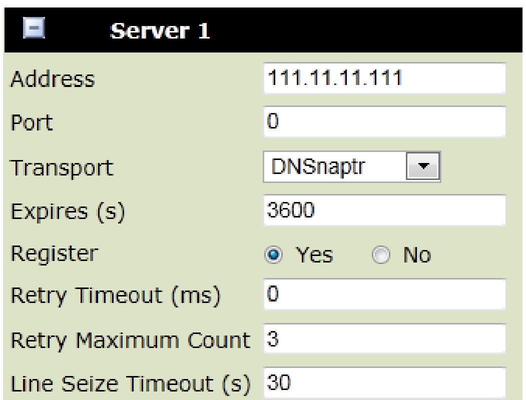
- Enter the **Display Name**. This example shows 4100041.
- Enter the **Address**. This example shows 4100041 to match Display Name. Address represents the extension created for the device in CUCM.
- Enter the **Authentication User ID**. This example uses `sstvoipuser`. In newer software The authentication user ID and password are located in a separate section called “Authentication”:



- 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`.

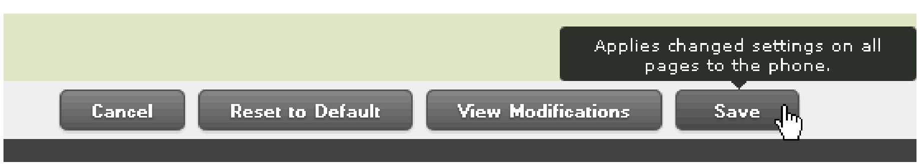
To configure SIP server settings:

1 Expand **SIP Server 1.**



- 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.

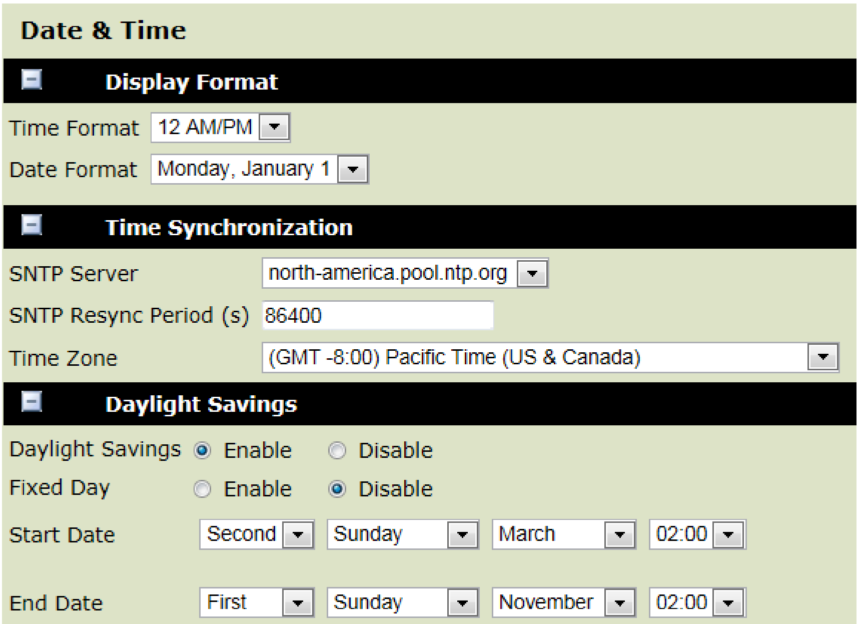
2 Click **Save** to apply the settings.



3 The line is successfully registered and a solid phone icon  displays on the phone screen. On the Trio a green circle with a check mark will appear.

To configure date and time settings:

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



2. Change the following values:
   - Under **Display Format**, choose a **Time Format** and **Date Format** that you want the phone to display.
   - Under **Time Synchronization**, choose an **SNTP server** in your region that the phone receives its time setting from and select a region in **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.

You have successfully configured settings that register a conference phone with CUCM and configured basic phone settings using the Web Configuration Utility.

Configure Fault Tolerance for RealPresence Trio 8800 or SoundStation IP for CUCM

This section shows you how to manually configure fault tolerance for the Polycom RealPresence Trio 8800 or SoundStation IP phones registered with CUCM. Configuring fault tolerance specifies how the phone re-registers if the primary CUCM fails. Fault tolerance is supported on CUCM 8.x, 9.x and 10.x, and is not supported on CME/SRST at this time.

To set up fault tolerance:

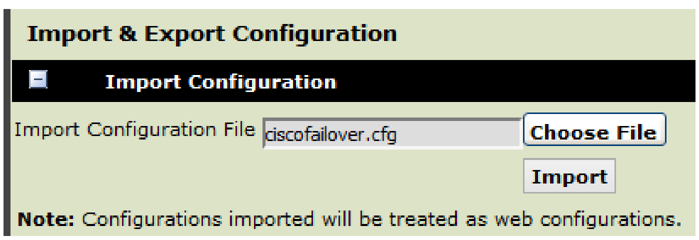
1 Copy and paste the following XML text into a configuration file with .cfg file extension name, and remove the comments.

```
<?xml version="1.0" encoding="utf-8"?>
  <polycomConfig>
    <reg
    reg.1.server.1.address="cucm.local1" <- set this to the name used in the
DNS cache section below
    reg.1.address=3301@10.223.224.6 <- set this to the extention@any valid CUCM
subscriber IP address, this
MUST be ext@IPAddress
    reg.1.displayName="Conference Room - 3301" <- Name that the phone sends
when making a call, generally overridden by
CUCM setting
    reg.1.auth.userId="3301" <- this is the username assigned to the phone on
its “device” page
    reg.1.auth.password="fea83340" <- if using digest authentication, this is
the digest password as set on the
  user
    reg.1.label="Ext-3301" <- label that shows up on the line button
                reg.1.server.1.port="5060" <- SIP port, 5060 is the default
                reg.1.server.1.transport="TCPOnly" <- set to TCPOnly
                reg.1.tcpfastfailover="1" <- tells phone to quickly fail over
in the event its current registrar is down
                reg.1.server.1.retryTimeOut="500"
                reg.1.server.1.retryMaxCount="3"
                reg.1.server.1.failOver.reRegisterOn="1" <- tells phone to
attempt to reregister to highest
 available server in the DNS cache list
                reg.1.server.1.failOver.failBack.mode="duration"
                reg.1.server.1.failOver.failBack.timeout="60"
```

```
    >
    </reg>
                <dns
                dns.cache.A.1.address="10.223.224.6" <- first subscriber IP
address, can have up to 4 in the
    list
                dns.cache.A.1.name="cucm.local1" <- fake dns name, must NOT
match any DNS name
                dns.cache.A.2.address="10.223.224.5" <- second subscriber IP
address
                dns.cache.A.2.name="cucm.local1" <- second same fake DNS name
                >
                </dns>
  </polycomConfig>
```

2 In a web browser, log into the Web Configuration Utility for your device.

3 Navigate to **Utilities > Import & Export Configuration**, click **Choose File**, and select the configuration file you created in step 1. In this example, the configuration file is named `ciscofailover.cfg`.



4 Click **Import**. If the upload fails, check your configuration file for errors.

Troubleshoot 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.

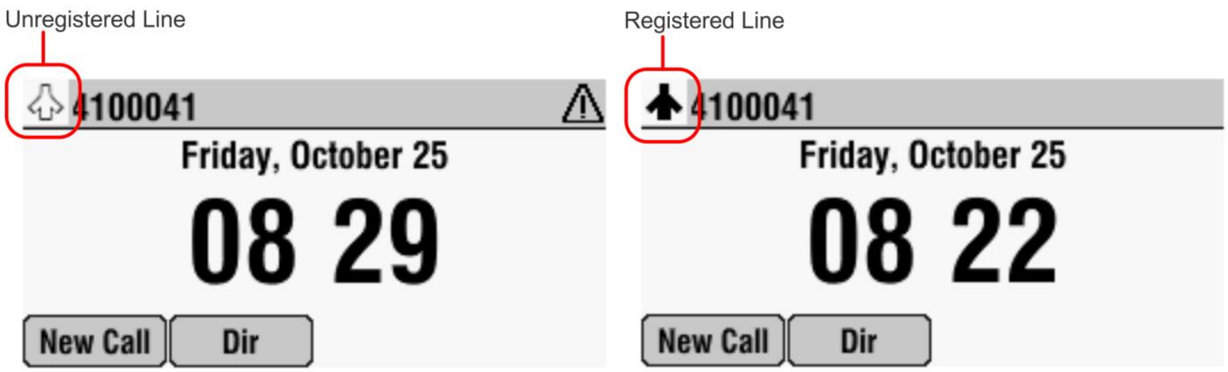
For further help with RealPresence Trio solution, see the *RealPresence Trio Administrator Guide* at RealPresence Trio on Polycom Support.

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 figure Unregistered and Registered Line Icon shows an unregistered and a registered line icon.

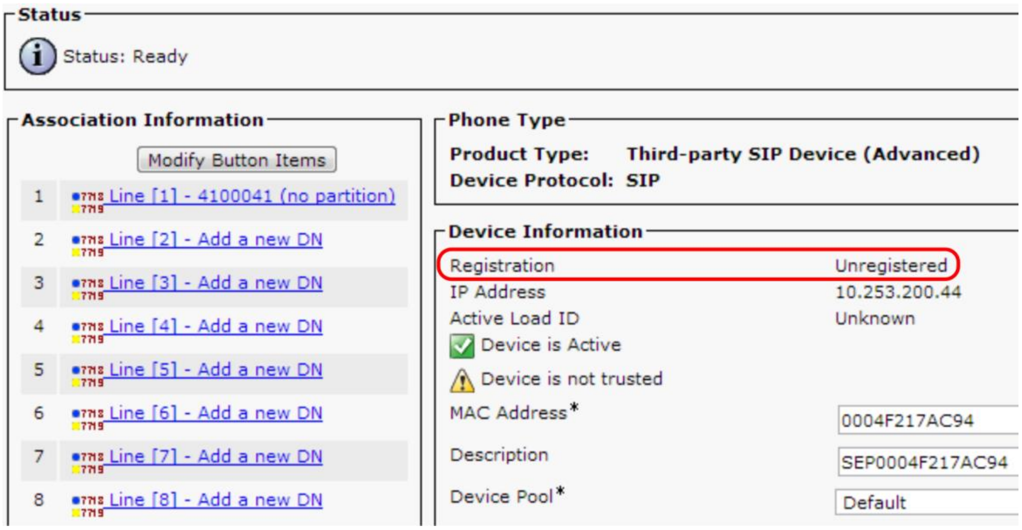
Unregistered and Registered Line Icon



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 figure CUCM Phone Configuration Page. If the phone is unregistered, CUCM shows *Registration Unregistered*.

CUCM Phone Configuration Page



Once the phone is properly registered, the CUCM shows the device as registered, shown in the figure Device Registered with CUCM.

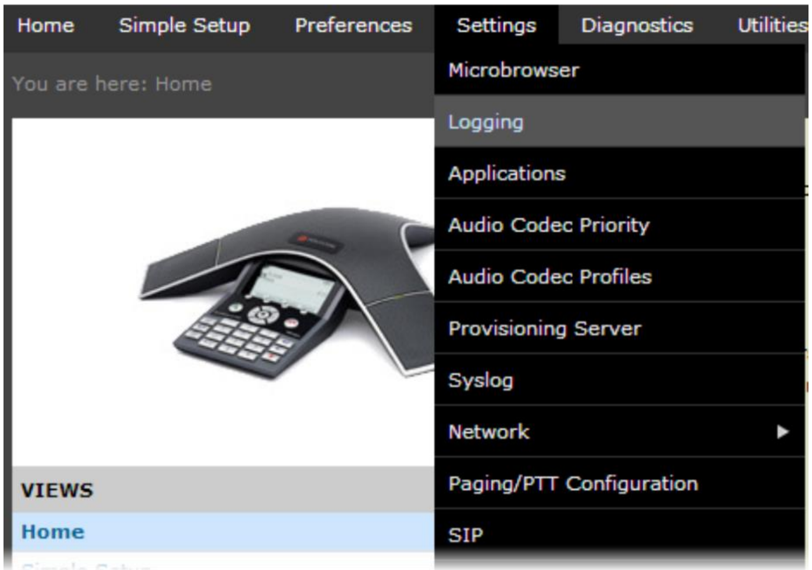
Device Registered with CUCM

Logging

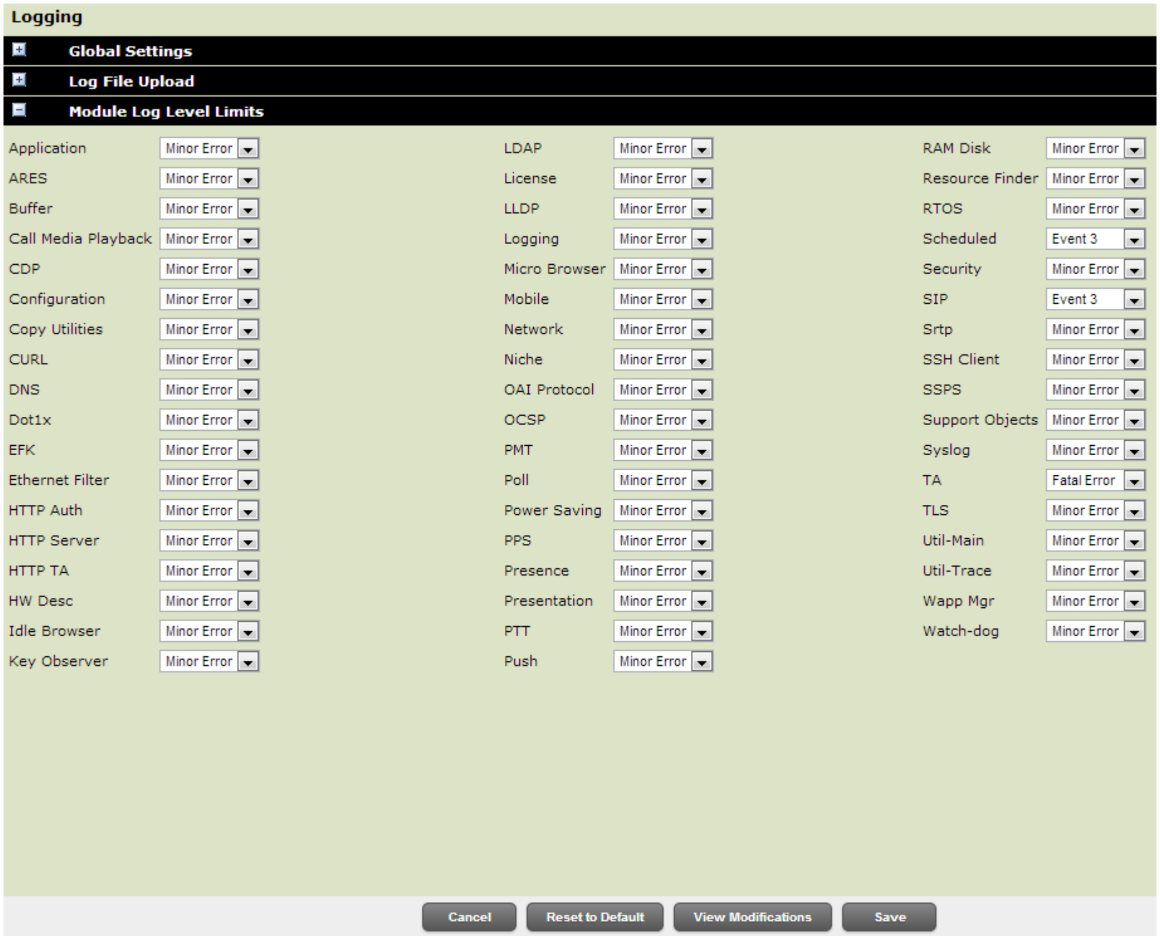
You can use the Polycom Web Configuration Utility to access phone log files.

To access log files:

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



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

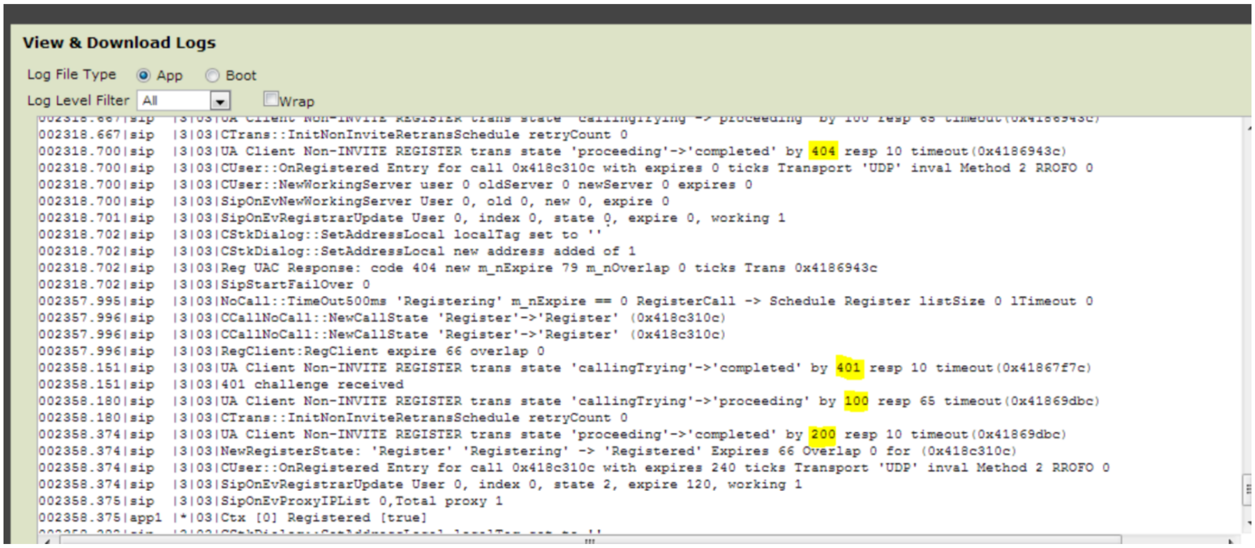


3 Change the SIP level to **Event 3**.

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

The figure 404 Error highlights examples of a registration error.

404 Error



Get Help

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.

For information on UC software and voice product documentation, see Voice Support on the Polycom Support site.

If you are deploying a large number of Polycom SoundStation IP conference phones with CUCM, see *Bulk Deployment of Polycom® SoundStation® IP and Polycom® SoundStation® Duo Conference Phones with Cisco Unified Communications Manager (CUCM)*.

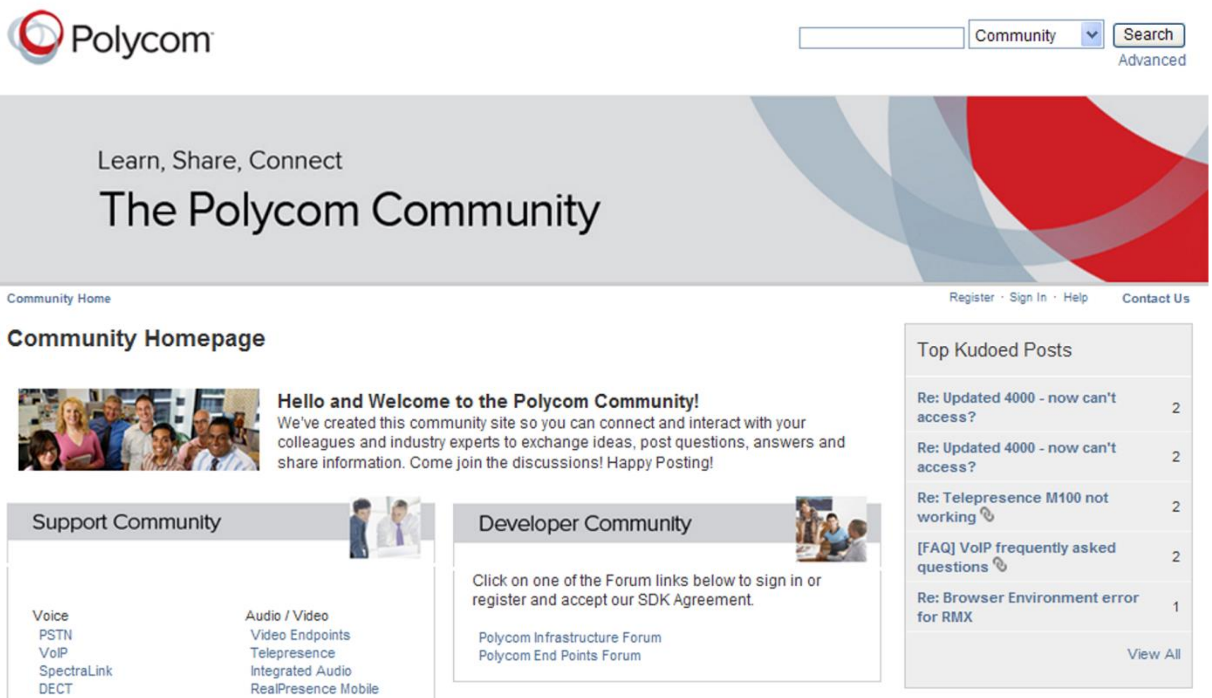
For workarounds to frequent issues, see Polycom Engineering Advisories and Technical Notifications.

Release Notes for specific UC software releases are posted on the Polycom UC Software Support Center.

If you are 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 *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, 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.



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