SHORETEL APPLICATION NOTE

for

Polycom SoundStation IP6000 and IP7000

Date:	December 18, 2017
App Note Number:	TC-17065
For use with:	Polycom SoundStation IP Phones
Product:	ShoreTel Connect ONSITE
System:	ST Connect 21.82.2128.0

Contents

ShoreTel tests and validates the interoperability of the Member's solution with ShoreTel's published software interfaces. ShoreTel does not test, nor vouch for the Member's development and/or quality assurance process, nor the overall feature functionality of the Member's solution(s). ShoreTel does not test the Member's solution under load or assess the scalability of the Member's solution. It is the responsibility of the Member to ensure their solution is current with ShoreTel's published interfaces.

The ShoreTel Technical Support organization will provide Customers with support of ShoreTel's published software interfaces. This does not imply any support for the Member's solution directly. Customers or reseller partners will need to work directly with the Member to obtain support for their solution.

Introduction

This document describes the configuration procedures for integrating the Polycom SoundStation IP Phones as SIP extensions on the ShoreTel Connect Onsite system.

Polycom

Polycom helps organizations unleash the power of human collaboration. More than 400,000 companies and institutions worldwide defy distance with secure video, voice and content solutions from Polycom to increase productivity, speed time to market, provide better customer service, expand education and save lives. Polycom and its global partner ecosystem provide flexible collaboration solutions for any environment that deliver the best user experience, the broadest multi-vendor interoperability and unmatched investment protection.

Make it feel like everyone's together in the same room. Polycom conference phones are the standard because they deliver the clearest sound to every participant in every location. Our advanced audio technology allows each conference phone to intelligently adapt to different room environments, so everyone can hear and be heard, even when more than one person talks at a time. You'll eliminate confusion and enhance productivity. Not a single word—or opportunity—gets missed.

Features

- Unparalleled clarity – Polycom HD Voice makes your conference calls sound amazingly clear and lifelike
- More productive conference calls – Our patented Polycom Acoustic Clarity technology provides you with the best conference phone experience and no compromises
- Conferencing technology that is ideal for midsize rooms – Its 12-foot (3.5-meters) microphone pickup is designed for small and midsize conference rooms accommodating up to 12 people

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Polycom® SoundStation® IP 6000

The SoundStation IP 6000 is an advanced IP conference phone that delivers superior performance for small to midsize conference rooms. With advanced features, broad SIP interoperability and remarkable voice quality, the SoundStation IP 6000 offers a price/performance breakthrough for SIP-enabled IP environments.

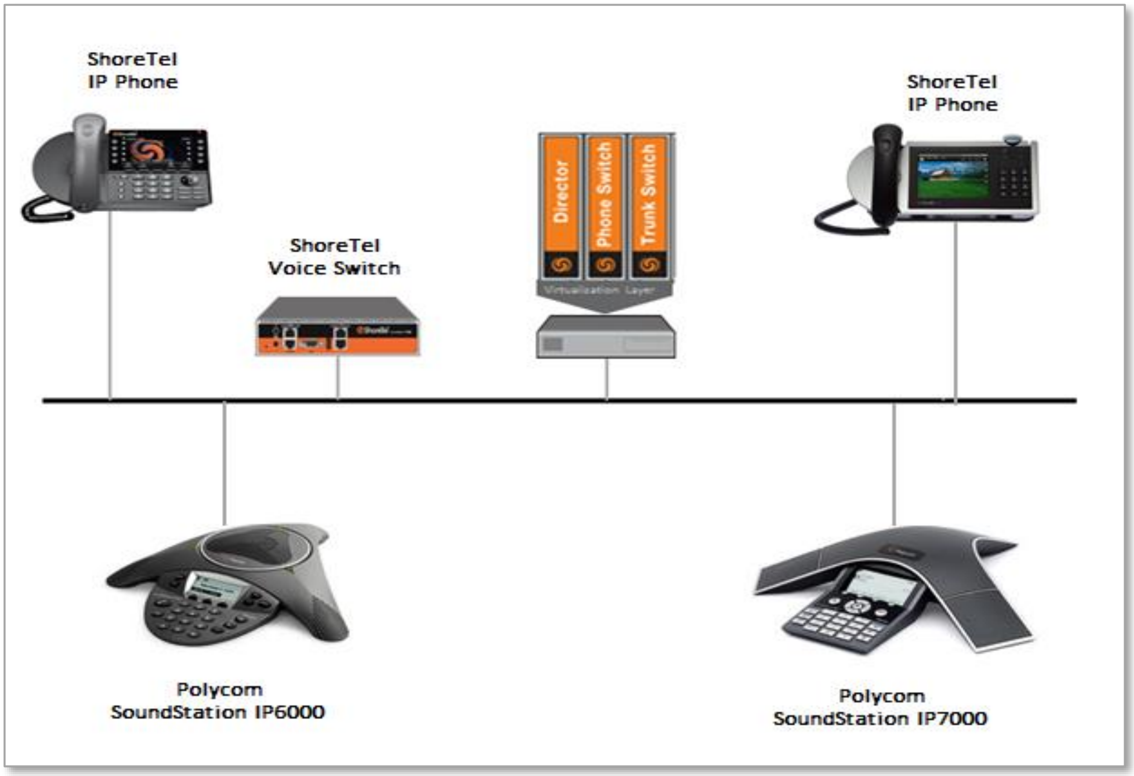
The SoundStation IP 6000 features Polycom® HD Voice™ technology, boosting productivity and reducing listener fatigue by turning ordinary conference calls into crystal-clear interactive conversations. It delivers high-fidelity audio from 220 Hz to 14 kHz, capturing both the deeper lows and higher frequencies of the human voice for conference calls that sound as natural as being there.

Polycom® SoundStation® IP 7000 SIP-Based IP Conference Phone

The Polycom® SoundStation® IP 7000 is a breakthrough conference phone that delivers outstanding performance and a robust feature set for SIP-based VoIP platforms. It is the most advanced conference phone ever developed, and is ideal for executive offices, conference rooms, and boardrooms.

The SoundStation IP 7000 features Polycom® HD Voice™ technology, boosting productivity and reducing listener fatigue by turning ordinary conference calls into crystal-clear interactive conversations. It delivers high-fidelity audio from 160 Hz to 22 kHz, capturing both the deeper lows and higher frequencies of the human voice for conference calls that sound as natural as being there.

Network Topology



Test Environment

- ShoreTel Connect ONSITE Server
- ShoreTel Virtual Phone Switch
- ShoreTel Voice Switch
- ShoreTel IP Phones
- Polycom SoundStation IP Phones (UC Software Version 4.0.12.0926)

NOTE: This Application Note assumes the setup, configuration and licensing of the Virtual/Physical Switches has already been completed. If you require additional information, please refer to the ShoreTel Connect Onsite Planning and Installation guide at the following location.

[ShoreTel Connect Onsite Planning and Installation Guide]

Special Notes

The following considerations must be taken when implementing Polycom SoundStation IP Phones as SIP Extensions on the ShoreTel Connect ONSITE system.

ShoreTel Extension License

Extension Licenses are required for each Polycom SoundStation IP Phone.

ShoreTel SIP Phone License

Deployment of SIP Extensions require a SIP Phone License. One SIP Phone License is required for each Polycom SoundStation IP Phone SIP Extension.

Polycom Digitmap

For the Digitmap parameters, please be sure to manipulate the Digitmap to conform to your ShoreTel Dial Plan settings. Failure to do so will result in Off-Hook dialing failures, where the phone number dialed may be incomplete. Please refer to Polycom's documentation for additional information, see Technical Bulletin 11572.

NOTE: For additional information on SIP Endpoints with a ShoreTel Connect ONSITE system, please refer to Chapter 19 of the ShoreTel Connect Onsite System Administration Guide.

ShoreTel Configuration

This section describes the detailed steps required on the ShoreTel Connect ONSITE system to configure Polycom SoundStation IP Phones as SIP extensions.

Call Control Options

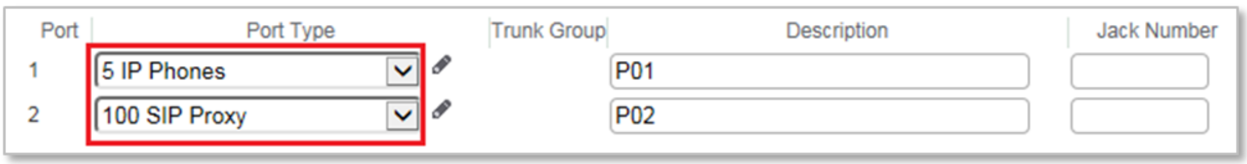
This section describes the SIP settings required on the ShoreTel system to work with Polycom SoundStation IP Phones. This is accomplished from ShoreTel Connect Director.

1. Navigate to Administration > Features > Call Control > Options
2. Verify the parameters located under the **SIP** section
3. **Realm**: The realm is used in authenticating all SIP devices. Changing this value will require a reboot of switches serving as SIP extensions. It is not necessary to modify this parameter
4. **Enable SIP Session Timer**: Ensure this parameter is checked
5. **Session interval**: Session interval value indicates the SIP session registration period. There is no need to modify the default value of 1800 seconds.
6. **Refresher**: The refresher setting decides if user agent client or user agent server refreshes the session. There is no need to modify the default value of “Caller (UAC).”
7. Click **SAVE**

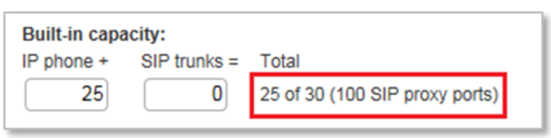
SIP Proxy Settings – Allocating Ports for SIP Extensions

This section describes the Switch configuration required on the ShoreTel system to work with the Polycom SoundStation IP Phones. Depending on the switch type, ShoreTel Voice Switches, and Virtual Phone Switches support variable numbers of SIP Proxies and IP Phones, and can be verified on the Switch Edit page of ShoreTel Connect Director.

ShoreTel ShoreGear Switches with processing resources that support Digital and Analog ports can be reallocated to support 100 SIP Proxies. The ShoreTel Administrator can define one of the "Port Type" settings from the available ports to "100 SIP Proxy", as well as sufficient "IP Phone" ports to support the total number of Polycom SoundStation IP Phones. The following example shows Port allocation designated on a ShoreTel SG-90 for IP Phones and SIP Proxy resources



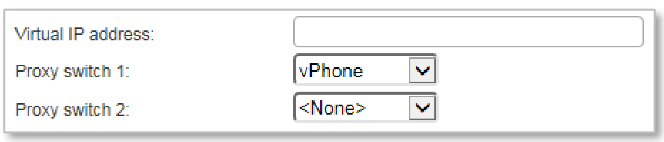
If the ShoreTel ShoreGear Switch that you have selected has "built-in" capacity (i.e., ShoreGear 50/90/220T1/E1, etc.) for IP phones and SIP trunks, you can also remove 5 ports from the total number available to provide the "100 SIP Proxy" configuration necessary. Every 5 ports you remove from the total available will result in "100 SIP Proxy" ports being made available. The following example shows 5 ports removed from total available resulting in 100 SIP Proxy ports being available.



Site Settings

The next settings to address are the administration of Sites. The ShoreTel Administrator can designate up to two Proxy switches per site for redundancy and reliability: one switch is assigned as the primary Proxy server, and the other switch acts as the backup Proxy server in case the primary fails. A Virtual IP Address is the IP Address of the switch that is configured as the SIP Proxy server for the Site. The Virtual IP Address must be static. If you choose not to define a "Virtual IP Address," you can only define one proxy switch, and there will be no redundancy or failover capabilities. The switches available in the "Proxy Switch 1 / 2" will only be shown if proxy resources have been enabled on the switch. This is accomplished from ShoreTel Connect Director.

1. Navigate to Administration > System > Sites
2. Select the name of the Site in which SIP Proxies will be assigned
3. In the General Tab, set **Proxy switch 1:** Select the ShoreTel switch configured with SIP Proxies for the Site
4. Click **SAVE**

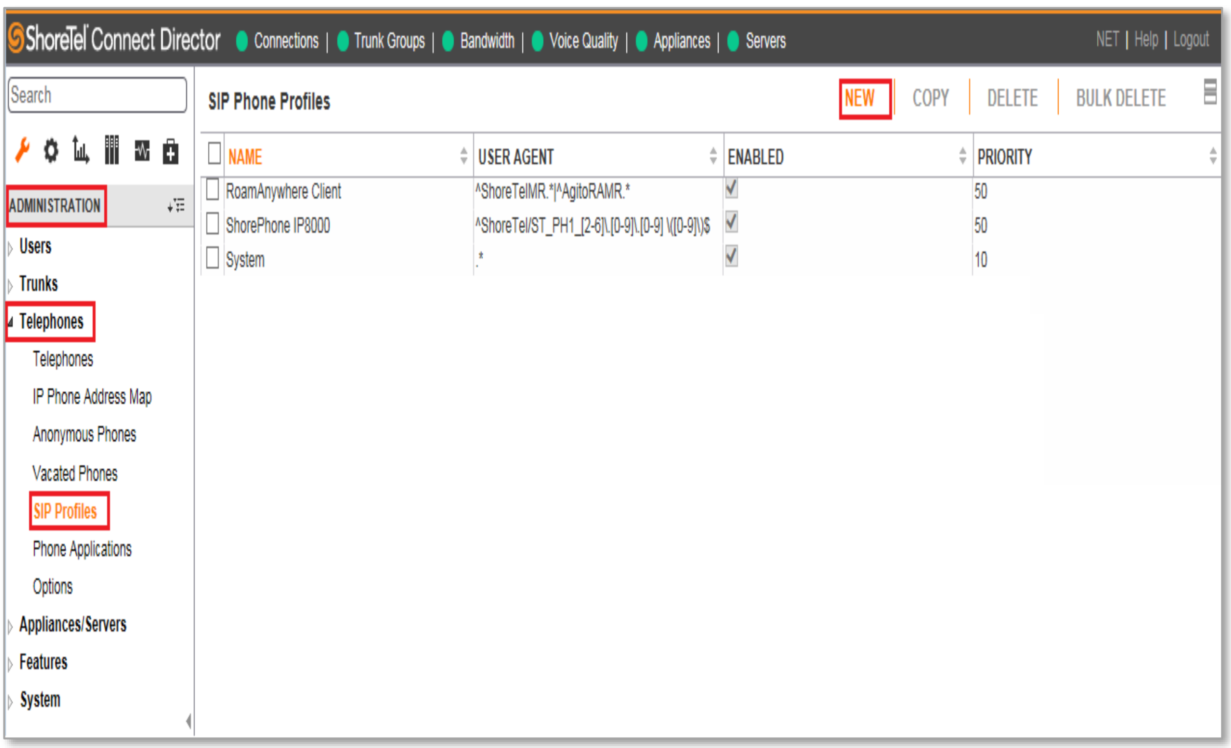


NOTE: Once the ShoreTel switch has been selected to support SIP Proxies, please note the IP Address of the switch as it will be used later in the Polycom Web Configuration Utility under the Line configuration.

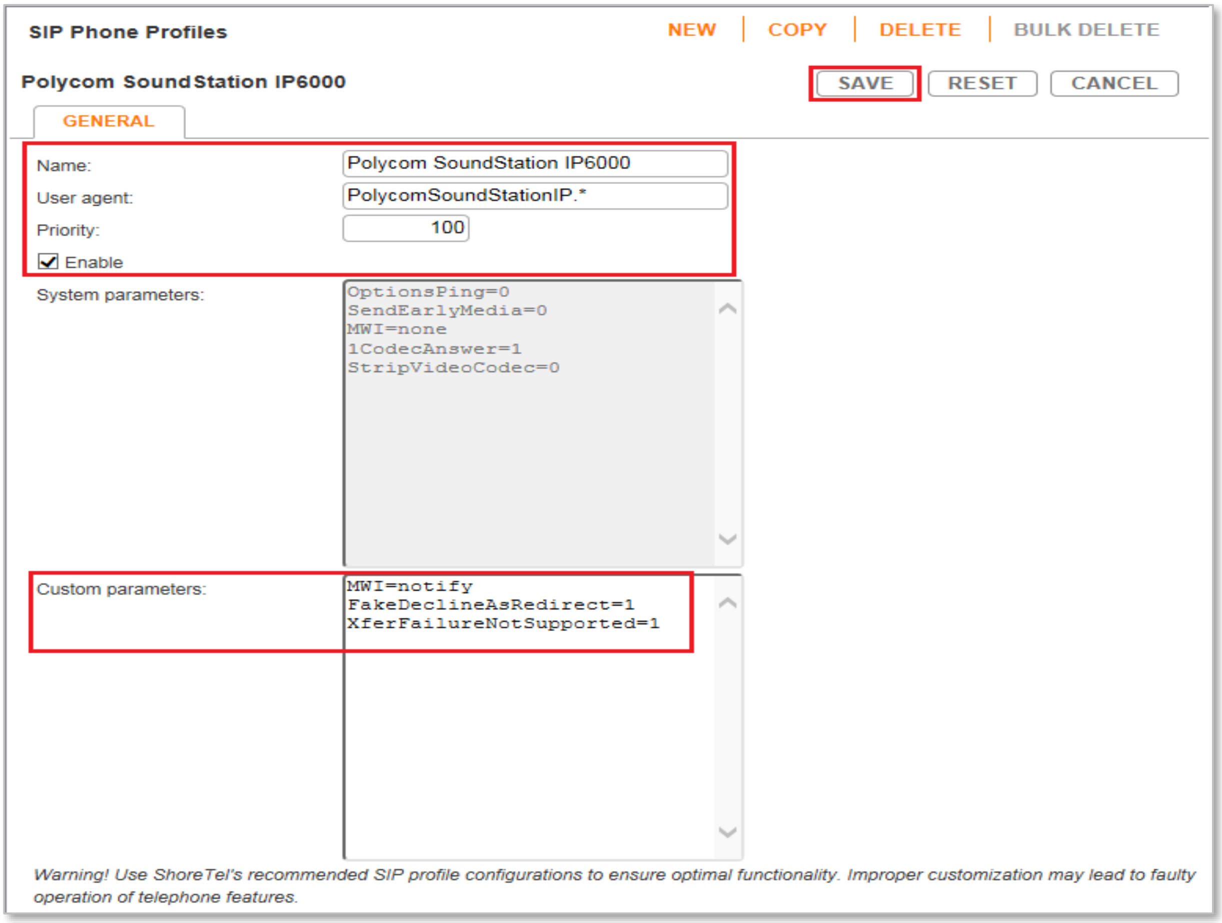
Configure a SIP Profile

This section describes the steps required to configure the “SIP Profiles” for the Polycom SoundStation IP Phones. By default, the Polycom SoundStation IP Phones utilize the “System” profile. In order to optimize the functionality, you will need to add a custom profile. This is accomplished from ShoreTel Connect Director.

1. Navigate to Administration > Telephones > SIP Profiles
2. Click **New**, to create a new SIP Profile



3. In the General Tab, define a **Name:** we recommend a name that describes the SIP endpoint.
4. For the parameter **User agent:**, enter “PolycomSoundStationIP.*” (without quotes, make sure to include the period followed by the asterisk)
5. The parameter “**Priority:**” defaults to 100, no change is required.
6. Enable the profile by checking (enabling) the **Enable** option.



7. In the “**Custom Parameters:**” section, add the following entries:

MWI=notify
FakeDeclineAsRedirect=1
XferFailureNotSupported=1

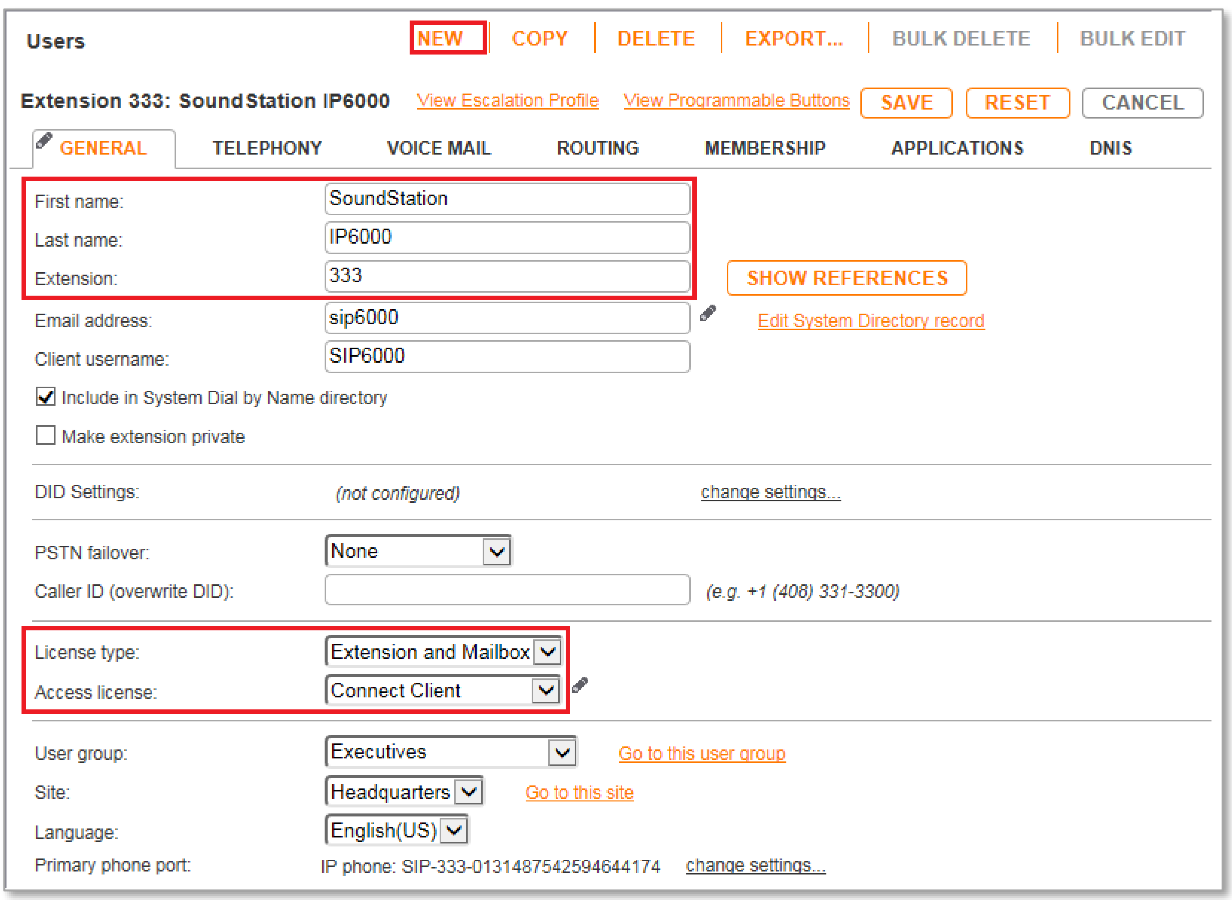
8. Click **SAVE**

Configure a Polycom SoundStation IP Phone as a SIP Extension

This section describes the steps required to configure a Polycom SoundStation IP Phone as SIP Extension on the ShoreTel system. This is accomplished from ShoreTel Connect Director.

1. Navigate to Administration > Users > Users
2. Click **New**, to create a new user
3. Define the **First name:** and **Last name:** Enter the appropriate user information
4. Define an **Extension:** ShoreTel Connect Director will automatically assign the next available extension number, but it can also be modified to any available extension number
5. Define the **License type:** and **Access license:** In our example, we chose “Extension and Mailbox”, and “Connect Client” for Access license

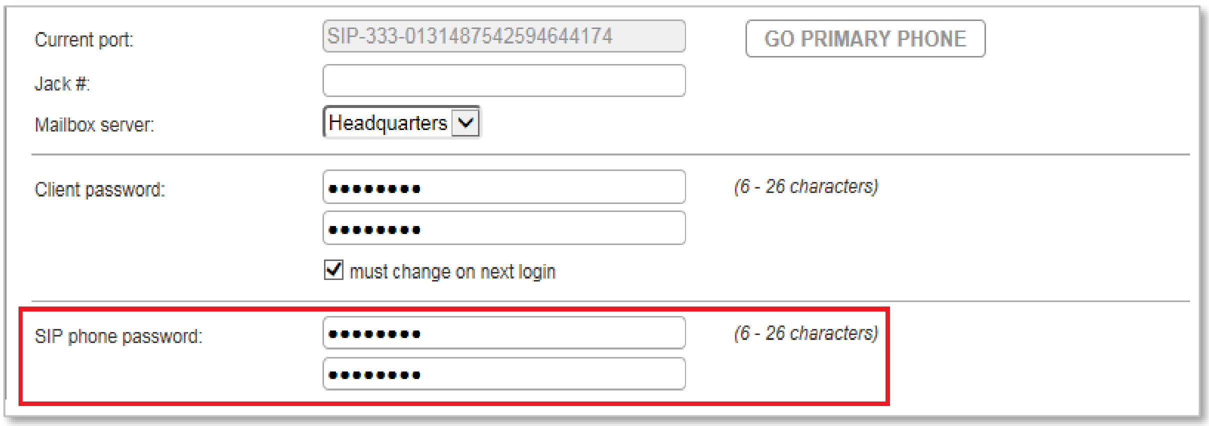
NOTE: If the “License type” is configured as “Extension-Only”, then “Any IP Phone” cannot be selected, but instead must be set to “SoftSwitch”.



6. Define a **SIP phone password:** There is no default SIP phone password configured, it is masked with the appearance that there is a default password, and must be defined by the ShoreTel Director Administrator. Make certain to type the password in both fields.

NOTE: Please note the "SIP phone password" configured for the user as it will be used later for the Polycom SoundStation IP Phones under the Line Identification configuration.

7. Click **SAVE**

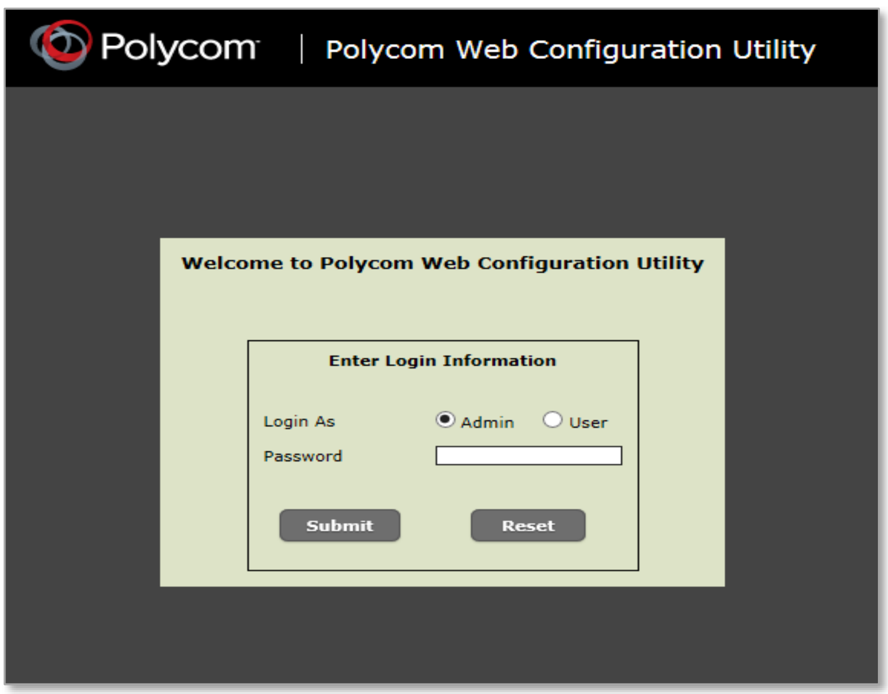


Polycom SoundStation IP Phones Configuration

The following steps detail the configuration process using the Polycom Web Configuration Utility for the Polycom SoundStation IP Phones to register as SIP extensions onto a ShoreTel Connect Onsite system.

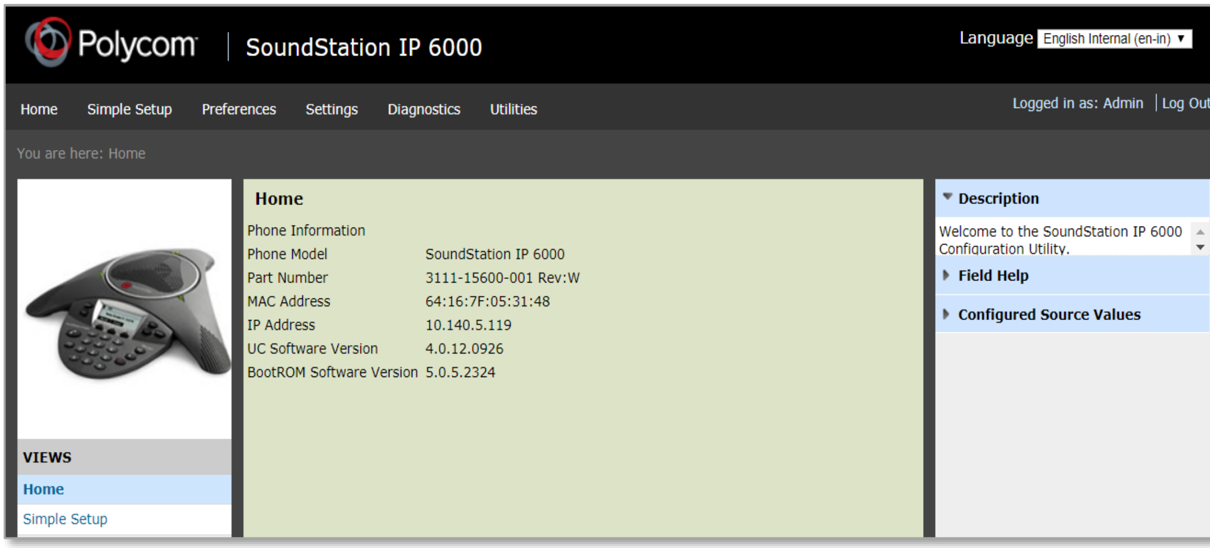
NOTE: The SoundStation IP Phones were powered up using the Ethernet LAN port, via an Ethernet cable, connected to a Power over Ethernet (PoE) switch. DHCP is enabled by default on the SoundStation IP Phones. A DHCP server was used for the network parameters, and then manually provisioned the minimum configuration parameters required for validation with the ShoreTel Connect Onsite system.

1. To access the Polycom Web Configuration Utility, the phone's IP Address will be required. The IP Address on the phone can be found through the phone's user interface, press the following keys **Menu > Status > Network > TCP/IP Parameters**.
2. Access the Polycom Web Configuration Utility by entering the phone's IP Address in the web browser's address bar, then press the **Enter** key.
3. The Polycom Web Configuration Utility login screen will be displayed.



4. For the option **Login As**, select **Admin**.
5. Enter the Admin password. The default Admin password is **456**.
6. Click the **Submit** button.

7. The Home screen will be displayed.



NOTE: We used the Web Configuration Utility method to configure the parameters on the Polycom SoundStation IP Phones. If your deployment involves provisioning more than 10 to 20 SoundStation IP Phones, Polycom recommends using configuration files hosted by a centralized server as the provisioning method

Configure the Date and Time parameters for the SoundStation IP Phones

This section describes the steps required to configure the Date and Time parameters on the Polycom SoundStation IP Phones.

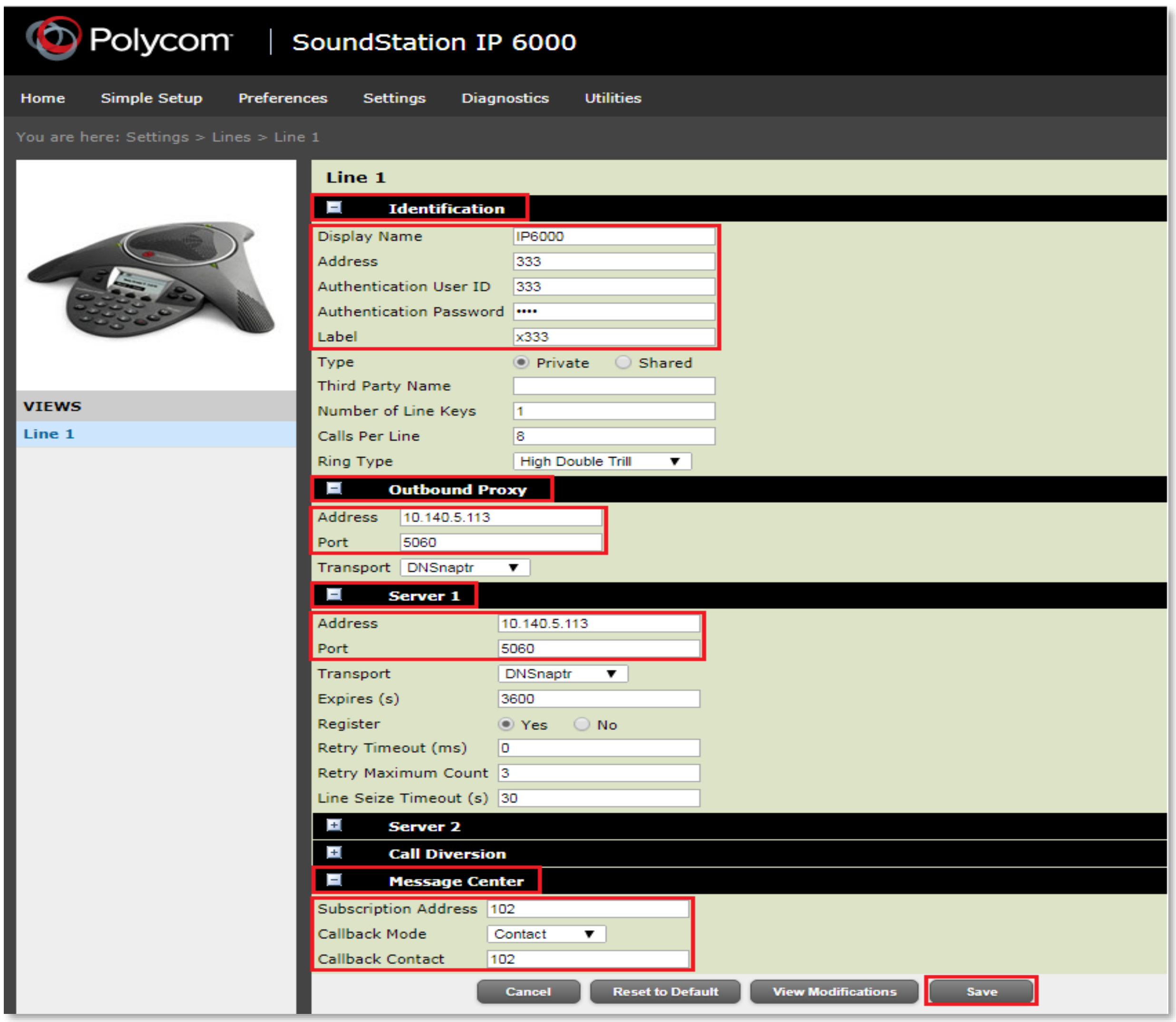
1. Set the Date and Time by selecting **Preferences > Date & Time** from the menu bar.

2. Expand the headings by clicking on the "**+**" symbol next to **Display Format**, **Time Synchronization** and **Daylight Savings**.
3. The **Display Format** parameters were not modified from their default values.
4. For the **Time Synchronization** parameters, select the **SNTP Server** where the phone will obtain its time. In our example, we selected the public time server **north-america.pool.ntp.org**. Set the **Time Zone** parameter associated to the phones GMT offset location.
5. The **Daylight Savings** parameters were not modified from their default values configured for North America.
6. Click the **Save** button

Configure the Line parameters for the SoundStation IP Phones

This section describes the steps required to configure the Line parameters on the Polycom SoundStation IP Phones.

1. Set the Line parameters by selecting **Settings > Lines** on the menu bar.
2. Expand the headings by clicking on the "**+**" symbol next to **Identification**, **Outbound Proxy**, **Server 1**, and **Message Center**.



3. Under the heading **Identification**, for **Display Name**, enter the desired display name (example: IP6000).

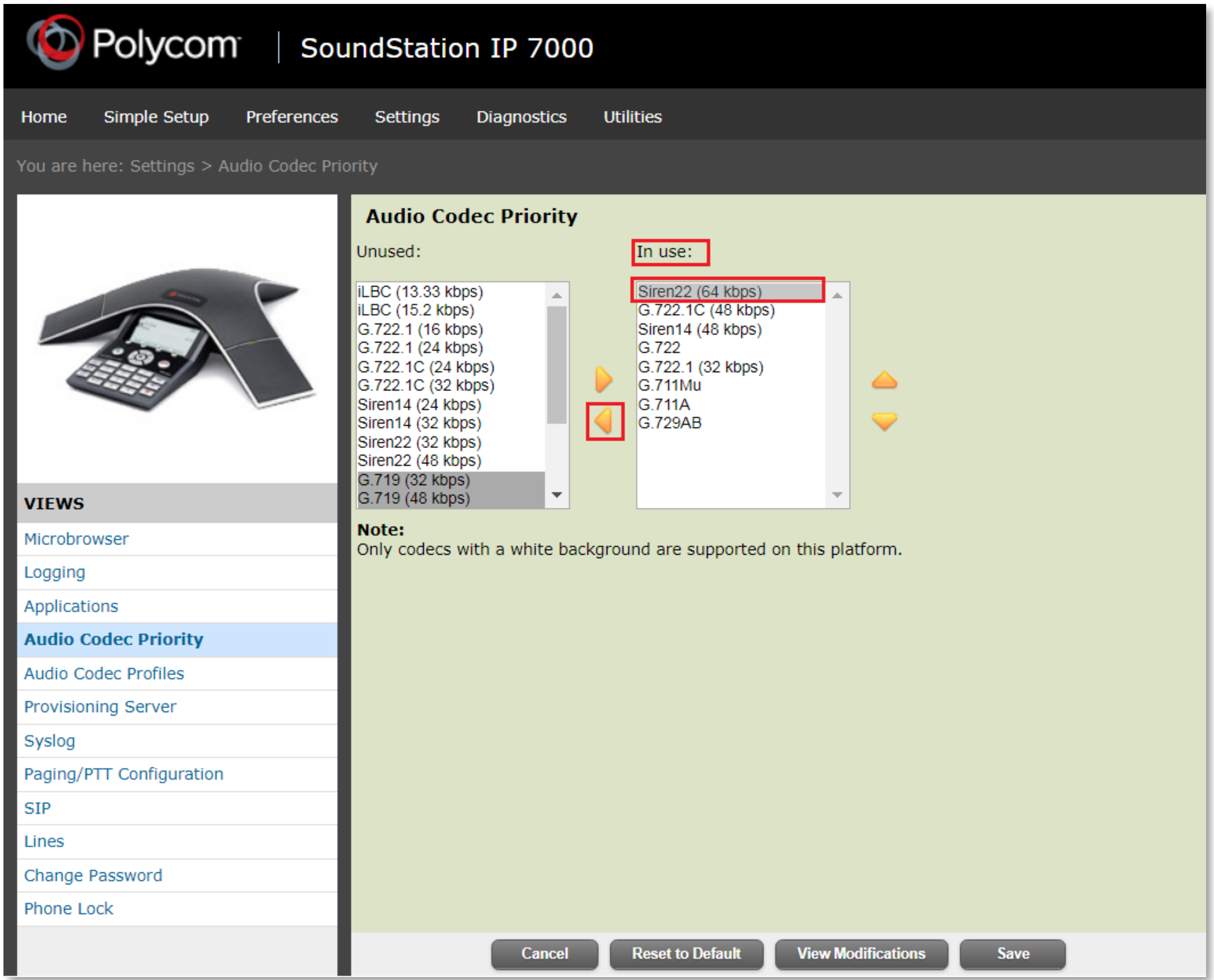
4. Define the line **Address**, in our example we entered the ShoreTel users extension number (example: 333) provisioned earlier with the ShoreTel Connect Director, under Administration > Users.
5. Define the **Authentication User ID**, in our example we entered the ShoreTel users extension number (example: 333).
6. Define the **Authentication Password**, enter the “SIP phone password” provisioned earlier with the ShoreTel Connect Director, under Administration > Users.
7. Enter the desired **Label,** in our example we entered the ShoreTel users extension number (example: x333). The label is the text that will be displayed on the phone.
8. Under the heading **Outbound Proxy**, for **Address,** enter the IP Address of the ShoreTel Proxy Server switch. In our example the ShoreTel Proxy switch IP Address is “10.140.5.113”. The ShoreTel Proxy switch was provisioned earlier with the ShoreTel Connect Director, under Administration > System > Sites.
9. Define the **Port**, enter port number “5060”.
10. Under the heading **Server 1**, for **Address,** enter the IP Address of the ShoreTel Proxy Server switch (same ShoreTel Proxy switch IP Address defined in Step 8).
11. Define the **Port**, enter port number “5060”.
12. Under the heading **Message Center**, define the **Subscription Address**, in our example we entered “102”. This is the Voice Mail Login extension provisioned in the ShoreTel Connect Director, and can be found under Administration > System > Dialing Plan > System Extensions.
13. Set the parameter **Callback Mode** to “Contact”.
14. Define the **Callback Contact**, in our example we entered “102” (same extension defined in Step 12).

    Now that the Message Center parameters are provisioned, SoundStation IP Phone users can access their voicemail messages when alerted by the message waiting indication on their phones.
15. Click the **Save** button.

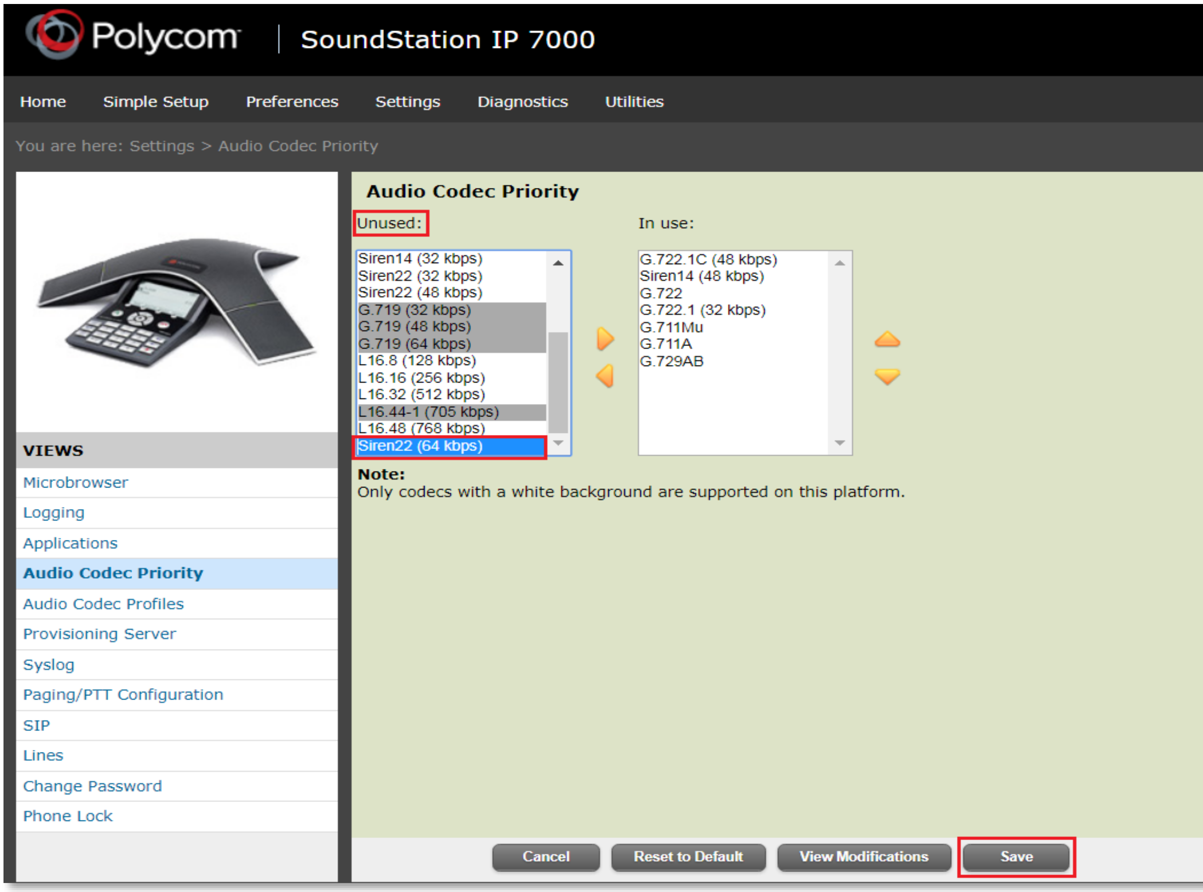
Configure the Audio Codec Priority (Required only on the Polycom SoundStation IP7000)

The following steps detail additional configuration parameters required on the Polycom SoundStation IP7000 Phone for interoperability with the ShoreTel Connect Onsite system.

1. From the Polycom Web Configuration Utility, set the Audio Codec Priority by selecting **Settings > Audio Codec Priority** on the menu bar.



2. Select the Codec “Siren22 (64kbps)” listed below the **In use** field.
3. Click on the left-pointing arrow, and the Codec will be displayed in the **Unused** field.



4. Click the **Save** button.

The SoundStation IP Phone will be ready for use and successfully registered with your ShoreTel Connect Onsite system.

Summary of Tests and Results

N/S = Not Supported N/T= Not Tested N/A= Not Applicable

ID	Result	Name	Description	Notes
1.1	PASS	Device initialization with static IP address	Verify successful startup and initialization of the device up to a READY/IDLE state using a static IP address	
1.2	PASS	Device reset – idle (for static configurations)	Verify successful re-initialization of device after power loss while device is idle	
1.3	PASS	Device initialization with DHCP	Verify successful startup and initialization of the device up to a READY/IDLE state using DHCP	
1.4	PASS	Device reset – idle (for dynamic configurations)	Verify successful re-initialization of device after power loss while device is idle	
1.5	N/T	Verify Diffserv Code Point support	Verify the ability to set Diffserv Code Point from SIP DUT (device under test)	
1.6	PASS	Verify Date and Time Update support	Verify setting of Date and Time Update on SIP DUT	
1.7	PASS	Place call	Verify successful call placement with normal dialing to a variety of terminating phones	
1.8	PASS	Receive call	Verify successful call placement with normal dialing to a variety of terminating phones	
1.9	PASS	Place call - redial	Verify successful call placement using re-dial to SIP Reference	
1.10	PASS	Place call – speed dial	Verify successful call placement using programmed speed dial	

ID	Result	Name	Description	Notes
1.11	PASS	CODEC support (DUT to ShoreTel Phone)	Verify successful call connection and audio path using all supported CODECs (G.711-Ulaw and G.729)	
1.12	PASS	CODEC support (DUT to SIP reference)	Verify successful call connection and audio path using all supported CODECs (G.711-Ulaw and G.729)	
1.13	PASS	CODEC negotiation	Verify successful negotiation between devices configured with different default CODECs (G.711-Ulaw and G.729)	
1.14	PASS	Hold DUT to SIP reference	Verify successful hold and resume of connected call	
1.15	PASS	Hold DUT to ShoreTel	Verify successful hold and resume of connected call	
1.16	PASS	Forward	Verify successful forwarding of incoming calls	
1.17	PASS	Forward from SIP DUT	Verify successful forwarding of incoming calls	
1.18	PASS	Mute	Verify device's mute function	
1.19	PASS	Out-of-band DTMF Transmission	Verify successful transmission of out-of- band digits (RFC2833) for calls placed to and from the DUT with a variety of other devices	
1.20	PASS	Missed call notification	Verify that device notifies the user about missed calls	
1.21	PASS	Volume	Verify the device's volume adjustment function	

ID	Result	Name	Description	Notes
2.1	PASS	Call waiting	Verify appropriate notification and successful connection of incoming call while busy with another party	
2.2	PASS	Park	Verify successful park and retrieval of connected call	
2.3	PASS	Extended forward	Verify extended call forwarding options – busy forwarding, ring no answer forwarding	
2.4	PASS	Extended forward from SIP DUT	Verify extended call forwarding options – busy forwarding, ring no answer forwarding	
2.5	PASS	Transfer – blind	Verify successful blind transfer of connected call	
2.6	PASS	Transfer – monitored	Verify successful monitored transfer of connected call	
2.7	PASS	Conference – ad hoc	Verify successful ad hoc conference of three parties	
2.8	PASS	Place call – secondary line	Verify successful call placement using secondary line	
2.9	PASS	Receive call – secondary line	Verify successful connection of incoming call on secondary line	
2.10	PASS	Callback	Verify successful connection of a call using the missed- call callback feature of the device	
2.11	PASS	Caller ID	Verify that Caller ID name and number is sent and received from SIP endpoint device	
2.12	PASS	SIP Device Generates Busy Tone	Verify that SIP DUT generates busy tone when calling a busy extension	

ID	Result	Name	Description	Notes
2.13	PASS	Initiate call to a Hunt Group	Initiate a call from DUT and verify that calls route to the proper Hunt Group and are answered by an available hunt group member with audio in both directions using G.729 and G.711 codecs.	
2.14	PASS	Initiate call to a Workgroup	Initiate a call from DUT and verify that calls route to the proper Workgroup and are answered successfully by an available workgroup agent with audio in both directions using G.729 and G.711 codecs.	
2.15	PASS	Hunt Group Member	Verify that incoming calls to a hunt group can be answered properly when DUT is a member of the hunt group.	
2.16	PASS	Workgroup Agent	Verify that incoming calls to a workgroup can be answered properly when DUT is an agent of the workgroup.	
2.17	PASS	Call Forward – “FindMe”	Verify that calls are forwarded to DUT’s “FindMe” destination. Verify that DUT works properly when it’s a “FindMe” destination	
2.18	PASS	ShoreTel Converged Conferencing Server	Verify that calls are properly forwarded to the ShoreTel Converged Conferencing Server and it properly accepts the access code and you’re able to participate in the conference.	
2.19	PASS	Bridged Call Appearance (BCA) extension	Verify that DUT can initiate calls properly to a BCA extension and the call is presented to all of the phones that have BCA configured. Verify that the call can be answered, placed on-hold and then transferred.	

ID	Result	Name	Description	Notes
2.20	PASS	Additional Phones (Simulring)	Verify that calls ring simultaneously on DUT and ShoreTel IP Phone	
2.21	PASS	Account Codes	Verify outbound calls when Account Codes is enabled on the system.	
2.22	PASS	Place call to an International Number	Verify an outbound call to the international number	
2.23	PASS	Place a private call using *67	Verify private call from DUT using *67	

Conclusion

Polycom SoundStation IP Phones were successfully validated and approved with ShoreTel Connect ONSITE.

Additional Resources

[ShoreTel Connect ONSITE System Administration Guide]

[ShoreTel Connect ONSITE Planning and Installation Guide]

[Polycom SoundStation IP Phones]

[Polycom Technical Notifications]

Version	Date	Contributor	Content
1.0	December 2017	J.Rodriguez	Original App Note

ShoreTel. Brilliantly simple business communications.

ShoreTel, Inc. (NASDAQ: SHOR) is a leading provider of brilliantly simple IP phone systems and unified communications solutions powering today's always-on workforce. Its flexible communications solutions for on-premises, cloud and hybrid environments eliminate complexity, reduce costs and improve productivity.

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