Application Note

ShoreTel™

ST-0217
December 14, 2006

Polycom Soundstation IP 4000 Full Duplex IP Conference Phone

Clear conference room conversations are critical for business communications. Full duplex conference phones running on the IP network allow successful conferencing while leveraging the power and cost effectiveness of IP communications. This application note provides the details on adding the Polycom Soundstation IP 4000 SIP Conference Phone to the ShoreTel IP Phone System.

Table of Contents

Overview

This document provides the details on the Polycom Soundstation IP 4000 Full Duplex IP Conference Phone and describes how to integrate this conference room phone into the ShoreTel IP Phone System. The document focuses on the configuration procedures needed to set-up the Polycom IP 4000 Conference Phone for the ShoreTel system and the configuration needed on the ShoreTel system to support the Polycom IP 4000 Conference Phone.

Features and Benefits

Quality conference room phones provide clear full duplex handsfree communications for all participants within a conference room as well as to the connected parties. Conference room phones on the ShoreTel IP phone system take advantage of this effective communications path while reaping the benefits of the power and cost effectiveness, through reduced costs of operation and maintenance, of ShoreTel's VoIP system.

Vendor Overview and Contact

Information regarding the Polycom IP 4000 Conference Room Phone can be found though the following contact information:

Polycom Headquarters

4750 Willow Road
Pleasanton, CA 94588-2708
USA

Phone:

1.800.POLYCOM (in North America)
or 1.925.924.6000
Fax: 1.925.924.6100

Sales:

1.800.POLYCOM
or 408.526.9000
http://www.polycom.com

Reseller Info:

Resellers who want to start selling this solution should contact us through contact info under:

http://www.polycom.com

Vendor Product Information

PolyCom Soundstation IP 4000 EX (Expandable)
SIP Conference Phone

Product Code: 2201-06642-001

Approximate List price: $1099 US MSR

Features

Polycom Soundstation IP 4000 SIP

Full-duplex conference phone - natural, two-way conversations

- Both directions of the audio path are active so you can talk and hear at the same time
- Uses Polycom's industry leading Acoustic Clarity Technology
- Self-adjusting operation - the phone automatically adjusts to the room conditions

SIP support

- Conference phone for your SIP enabled business
- Interoperates with Polycom VSX™-series and MGC™ equipment for a pure SIP solution

Expandable for greater room coverage

- Speak naturally from up to 10 feet away from a microphone
- Optional extension microphones to cover a larger room

Gated microphones

- Reduces echo and background noise

Backlit, graphical LCD display

- 248 x 68 pixel resolution LCD for easy menu navigation
- Informative interface that facilitates easy voice conferencing

Easily field upgradeable

- Upgrade SoundStation IP 4000 as new features become available
- Extends the life of the phone to protect your investment

Supports multiple calls

- See incoming caller ID when already on a call
- Handle up to 2 calls simultaneously - one active and one on hold

3 context-sensitive soft keys

- Simplifies use of features

Scroll keys

- Navigate easily within menus

State of the art technology

- Designed to comply to the latest and forward thinking industry standards--not just an old POTS conference phone stuffed with VoIP electronics

User selectable ringer tones

- Allows you to personalize and customize the way the phone will sound

Time and date synchronization

- Synchronizes with SNTP server for accuracy

Call lists

- Easy access to lists of Missed Calls, Received Calls or Placed Calls as well as to Corporate Directories

Do not disturb

- Temporarily turn ringer off if needed

IM and presence support

- Send short IM messages from the conference phone or Indicate your status to people on your buddy list

Multi-lingual user interface

- Set on-screen language to your preference. Choose from Chinese, Danish, Dutch.

Technical Specifications

Size

- 14.5 x 12.25 x 2.5 in
  (36.8 x 31.1 x 6.4 cm) (L x W x H)

Weight

- 1.75 lb. (0.8 kg)

Power

- Universal power supply (110/220 V, 50/60 cycles), (100 V for Japan)

Display

- High-resolution, backlit, graphical LCD
- Keypad
  - 12-key telephone keypad
  - On-hook/off-hook, redial, mute, volume up/down keys, menu navigation keys
  - Three context-dependent softkeys

Loudspeaker

- Frequency response: 300 - 3300 Hz
- Volume: Adjustable to 85 dB at 1/2 meter peak volume

Network interface

- Ethernet 10/100 Base-T
- IP address configuration
  - DHCP
  - Static IP
- Supported CODECs
  - G.711 (A-law and μ-law)
  - G.729a (Annex B)
- Time synchronization with SNTP server
- User selectable ringer tones
- Convenient volume adjustment keys
- Field upgradeable
  - Trouble-free software maintenance
  - Extends the life of the product to protect your investment

Regulatory compliance

Safety

- UL1950
- CSA C22.2, No. 950
- EN60950
- IEC60950
- AS/NZS3260

EMC

- FCC (47 CFR Part 15) Class B
- ICES-003 Class
- EN55022 Class B
- CISPR22 Class B
- AS/NZS 3548 Class B
- VCCI Class B
- EN55024

Protocol support

- IETF SIP (RFC 3261 and companion RFCs)
  - Local feature-rich GUI
  - Call transfer, hold, divert (forward)
  - Called, calling, connected party identification/ information
  - One-touch redial
  - Local 3 way conferencing
  - Up to 2 dedicated lines, 2 call appearances per line with call waiting indication
  - User configurable directories and call history (Missed, placed and received)
  - Presence: buddy list, my status
  - Instant messaging
  - Unicode character support
  - Distinctive incoming call treatment/call waiting

– Do not disturb function
– Local call timer
– Multilingual user interface
– Automatic off-hook call placement
– Security features such as digest authentication

Quality of service architecture

- Silence suppression
- Echo cancellation
- Jitter buffer

Management support

- Web-based configuration (HTTP)
- (T)FTP support for downloading of software updates and configuration files

SoundStation IP 4000 ships with:

- Quick reference guide
- Warranty registration card
- Tabletop console with integrated display and keypad
- Power Interface Module (PIM)
- Universal power supply brick
- 6 ft Cat 5 Ethernet cable
- 25 ft Power/Ethernet cable (Connects PIM to console)

Warranty

- One year

Environment conditions

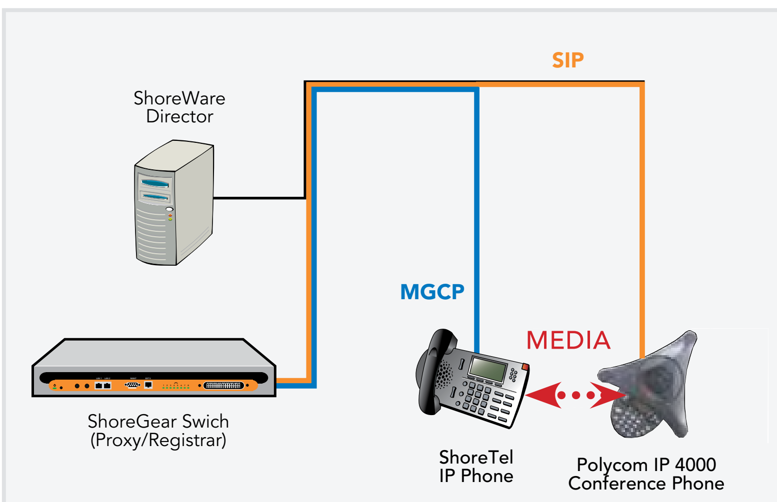
- Operating temperature: 32° - 104° F (0° - 40° C)
- Relative humidity: 20% - 85% (noncondensing)
- Storage temperature: -22° - 131° F (-30° - 55° C)

SoundStation IP 4000 features

- Full-duplex speakerphone
- Multiple call support
- Hold
- Call transfer (if supported by IP telephony system)
- Local 3 way conferencing
- Centralized conference (if supported by IP telephony system)
- Redial
- Mute
- Caller ID (if supported by IP telephony system)
- Phone book

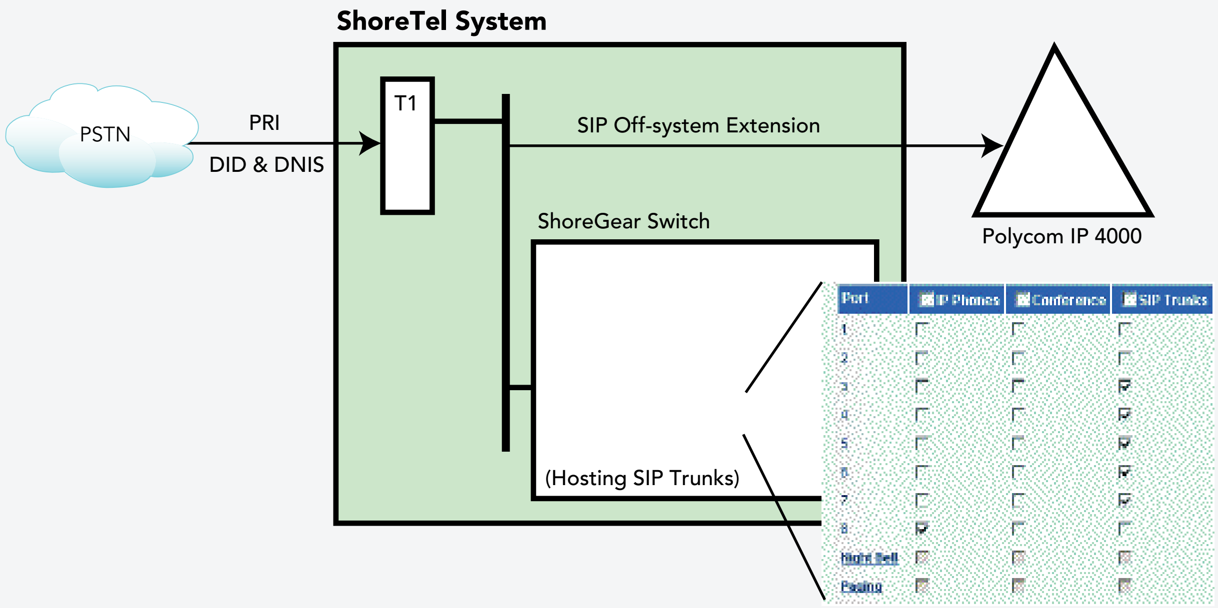
Architecture Overview

The following is a diagram of the solution architecture showing the integration between the Polycom Soundstation IP 4000 Conference Phone and the ShoreTel IP System:



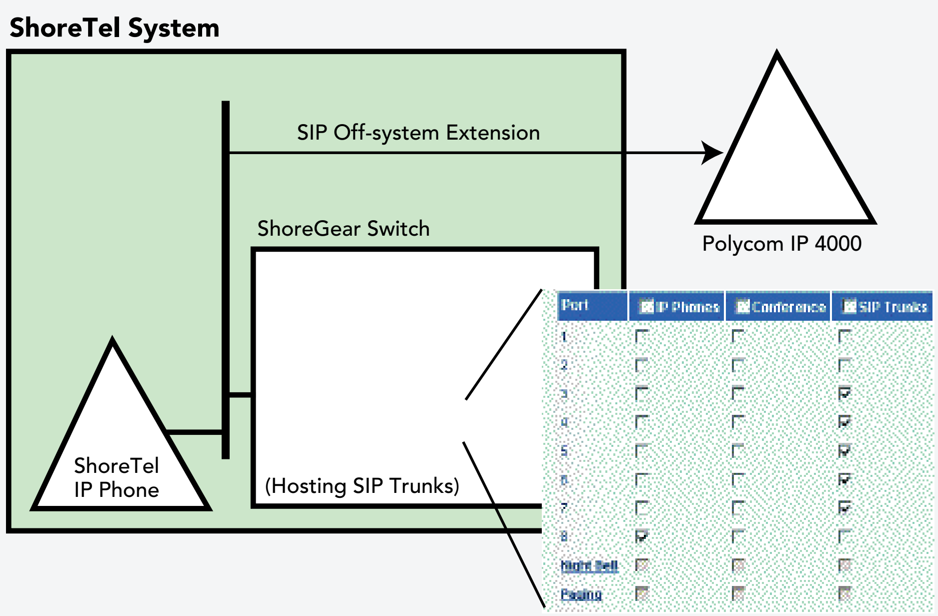
Sip devices are being implemented on the ShoreTel system with SIP trunks. The ShoreTel system will use outbound call routing for the SIP devices. There are various ways to handle the call routing for the SIP devices. The following drawings depict different call routing designs:

1. Route the call with DID (Direct Inward Dial) or DNIS (Dialed Number Identification Service) directly to the Polycom IP 4000 extension. In this configuration tandem trunking is enabled and off system extensions are defined in the trunk group. Shown in the drawing is a snapshot of the "Edit ShoreGear Switch" screen indicating the selection of SIP trunks.



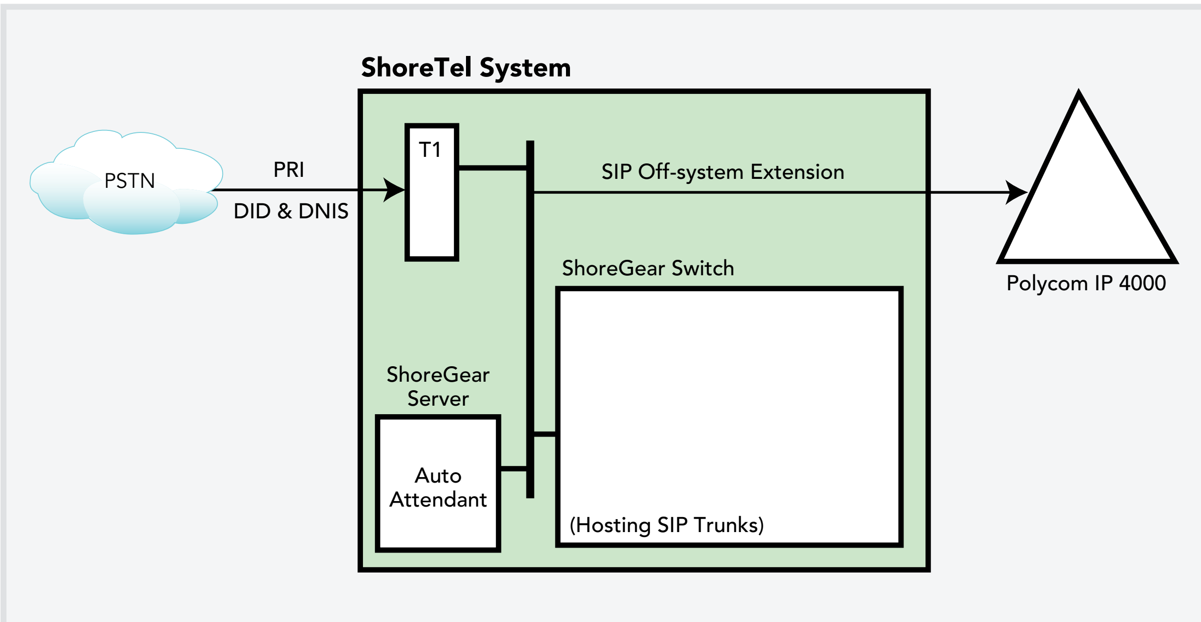
Edit ShoreGear Switch Screen Snapshot

2. Routing an internally dialed extension to the Polycom IP 4000 extension (off system extensions are defined in the trunk group). The selection of SIP trunks is shown again in the snapshot of the "Edit ShoreGear Switch" screen.



Edit ShoreGear Switch Screen Snapshot

3. Routing through the Auto Attendant can be accomplished in two ways.
   a. Program "Multiple digits" field in menu to "Go to extension" operation for routing to Polycom IP 4000.
   b. Program digit "0" to "Go to extension" operation for routing to Operator extension. Operator will have to transfer the call to the Polycom IP 4000.



Requirements, Certification and Limitations

The following requirements are necessary to integrate a Polycom IP 4000 Conference Phone to the ShoreTel IP Phone System as described in this Application Note.

ShoreTel Requirements

- ShoreWare Server Software, ShoreTel 6 or higher. Versions prior to this release will not support the Polycom product.
- ShoreTel SIP Trunk port licenses are required.

Polycom IP 4000 Phone Requirements

- Polycom IP 4000 Conference Phone – this device should be running the latest firmware (see version support table below).

Certification

CT LABS TESTING SERVICES

ShoreTel ENTERPRISE IP PHONE SYSTEMS

CT LABS VoIP Endpoints INTEROP TEST 2006

Interoperability Test Program for VoIP Endpoints

Report template revision v1.5. Last change Dec 11, 2006

VoIP Endpoint product information:

Date:	Dec 11, 2006
Vendor Name:	Polycom
Product Name:	SoundStation
Product Model No.:	IP 4000
Product Release:	2201-06642-001 Rev:A, software release 2.02.0076
Date Range of tests Performed:	Nov 21 through Dec 7 2006

Test Results Overview

This section presents an overview of the results of all the VoIP Endpoint test cases available in this test plan template.

Table 1-1: Basic Feature Test Cases

Passed?	ID	Optional?	Name	Description/Notes
Pass	1.1	Mandatory	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
Pass	1.2	Mandatory	Device reset – idle (for static configurations)	Verify successful re-initialization of device after power loss while device is idle
Pass	1.3	Mandatory	Device initialization with DHCP	Verify successful startup and initialization of the device up to a READY/IDLE state using DHCP
Pass	1.4	Mandatory	Device reset – idle (for dynamic configurations)	Verify successful re-initialization of device after power loss while device is idle
N/A	1.5	Mandatory	Verify Diffserv Code Point support	Verify the ability to set Diffserv Code Point from SIP DUT
Pass	1.6	Optional	Verify Date and Time Update support	Verify setting of Date and Time Update on SIP DUT
Pass	1.7	Mandatory	Place call	Verify successful call placement with normal dialing to a variety of terminating phones
Pass	1.8	Mandatory	Receive call	Verify successful reception of calls with normal dialing from a variety of calling phones
Pass	1.9	Optional	Place call – re-dial	Verify successful call placement using re-dial to SIP Reference
Pass	1.10	Optional	Place call – speed dial	Verify successful call placement using programmed speed dial
Pass	1.11	Mandatory for G.711, Optional for other CODECs	CODEC support – common (from DUT to ShoreTel Phone, REF-x)	Verify successful call connection and audio path using all supported CODECs (G.711-Ulaw and G.729) **Pass for G.711, G.729**

Passed?	ID	Optional?	Name	Description/Notes
Pass	1.12	Mandatory for G.711, Optional for other CODECs	CODEC support – common (from DUT to SIP Reference Phone, SIP-Ref)	Verify successful call connection and audio path using all supported CODECs (G.711-Ulaw and G.729) **Pass for G.711, G.729**
Pass	1.13	Mandatory (only if more than 1 CODEC is supported)	CODEC support – negotiated	Verify successful negotiation between devices configured with different default CODECs (G.711-Ulaw and G.729) **Pass for G.711, G.729**
Pass	1.14	Mandatory	Hold from DUT to SIP Reference	Verify successful hold and resume of connected call
Pass	1.15	Mandatory	Hold from DUT to ShoreTel Phone	Verify successful hold and resume of connected call
Fail	1.16	Mandatory	Forward	Verify successful forwarding of incoming calls **Issue to be fixed with a future revision of ShoreTel software.**
Fail	1.17	TBD	Forward from SIP DUT	Verify successful forwarding of incoming calls **See bug report Polycom12062006**
Pass	1.18	Optional	Mute	Verify device's mute function
Pass	1.19	Mandatory	Out-of-band / In-band DTMF Transmission	Verify successful transmission of in-band and out-of-band digits (RFC2833) for calls placed to and from the DUT with a variety of other devices **SIP INFO not supported but inband DTMF supported**
Pass	1.20	Optional	Missed call notification	Verify that device notifies the user about missed calls
Pass	1.21	Optional	Volume	Verify the device's volume adjustment function

Table 1-2: Performance Test Cases

Passed?	ID	Optional?	Name	Description/Notes
Pass	2.1	Mandatory	Speech quality – Minimal impairment	Verify acceptable voice quality between two parties with minimal network impairment condition
Pass	2.2	Mandatory	Speech quality – Moderate Impairment	Verify acceptable voice quality between two parties with low-to-moderate artificial network impairment condition
Pass	2.3	Mandatory	Speech quality – High Impairment	Verify acceptable voice quality between two parties with moderate-to-high artificial network impairment condition

Table 1-3: Extended Feature Test Cases

Passed?	ID	Optional?	Name	Description/Notes
Pass	3.1	Mandatory	Call Waiting	Verify appropriate notification and successful connection of incoming call while busy with another party
Pass	3.2	TBD	Park	Verify successful park and retrieval of connected call
Fail	3.3	Optional	Extended forward	Verify extended call forwarding options – busy forwarding, no-answer forwarding **Issue to be fixed with a future revision of ShoreTel software.**
Fail	3.4	Optional	Extended forward from SIP DUT	Verify extended call forwarding options – busy forwarding, no-answer forwarding **See bug report Polycom12062006**
Pass	3.5	Mandatory	Transfer – blind	Verify successful blind transfer of connected call
Pass	3.6	Mandatory	Transfer – monitored	Verify successful monitored transfer of connected call
Pass	3.7	Mandatory	Conference – ad hoc	Verify successful ad hoc conference of three parties
Pass	3.8	Optional	Place call – secondary line	Verify successful call placement using secondary line
Pass	3.9	Optional	Receive call – secondary line	Verify successful connection of incoming call on secondary line
Pass	3.10	Optional	Callback	Verify successful connection of a call using the missed-call callback feature of the device
N/A	3.11	Optional	Headset	Verify the device's support for external headsets (using headsets supplied by the 3P phone vendor)
Pass	3.12	Optional	Ring selection	Verify the device's ability to change the ring type
Pass	3.13	Mandatory	Caller ID Name and Number	Verify that Caller ID name and number is sent and received from SIP endpoint device
Pass	3.14	Optional	SIP Device Generates Busy Tone	Verify that SIP DUT generates busy tone when calling a busy extension
N/A	3.15	TBD	POTS Analog Gateway supports the transfer operation by "flashing"	Verify that the POTS Analog Gateway can support the transfer operation by "flashing"
Pass	3.16	Mandatory	Verify handling of "911"	Verify dialing "911" on DUT could connect with "911" services
N/A	3.17	Mandatory	Verify Fax Handling	Verify that fax can be sent and received through DUT

Limitations

The Office Anywhere™ feature will not be supported with SIP Trunks until the release of ShoreTel 6 Release 1.

Interaction with "911"Support

With the release of ShoreTel 6 support for 911 calls is limited. If a 911 call is made whatever number is configured for that SIP device is what will be sent out the local trunk which services 911 calls based on the "Site" of the ShoreGear switch which is associated for the SIP Trunk.

Other items to consider:

- If the SIP device is ONLY configured for example with a four digit extension then that is number which will be sent! Depending how the trunk from the CO is configured it may default to the billing address or something else. Check with your phone company on how 911 calls are handled.
- If the SIP device is configured with a full 10 digit DID then once again that's what will be sent out the local trunk where the ShoreGear Switch is located supporting the SIP Trunk.
- Should the SIP device be in NY and the ShoreGear switch which supports the SIP Trunk for the SIP device in San Jose then the 911 call will go out through whatever 911 trunk is configured for the San Jose site!
- It is recommended 911 is fully tested for based on the design!
- Should it be desired to use 10 digit DID, using "Digit Translation" can be used when on the ShoreTel system an "Off System" extension can be used. Example: Dialing 3510 can translate to 408-331-3510. This is only needed when configuring the "Out Bound" part of the call in the Trunk Group. See the Planning and Installation Guide for information on configuring Digit Translation.

Version Support

Product certified via the Technology Partner Certification Process for the ShoreTel system. Table below contains the matrix of Polycom SoundStation IP 4000 Conference Phone firmware releases certified on the identified ShoreTel software releases.

		Polycom SoundStation IP 4000 Conference Phone		
		Version 2.02.0076	N/A	N/A
ShoreTel Release	ShoreTel 6.1 11.14.5501.0 or newer	✓		
	N/A			
	N/A			
	N/A			

Configuration Overview

The following steps are required to configure the Polycom SoundStation IP 4000 Conference Phone to work with the ShoreTel system:

1. Configure General ShoreTel system settings
   a. Call Control Options, Site and Switch settings
2. Setup Trunk Groups
3. Setup Individual Trunks and User Groups
4. Configure Polycom Soundstation IP 4000 Conference Phone to function with the ShoreTel system
5. Configure Appropriate Call Routing options.

ShoreTel Configuration

This section describes the ShoreTel system configuration to support the Polycom SoundStation IP 4000 Conference Phone. The section is divided into general system settings and trunk configurations (both group and individual) needed to support the Polycom Conference Phone.

ShoreTel System Settings – General:

The first settings to address within the ShoreTel system are the general system settings. These configurations include the call control, the site and the switch settings. If these items have already been configured on the system, skip this section and go on to the "ShoreTel System Settings – Trunk Groups" section below.

Call Control Settings

The first settings to configure within ShoreWare Director are the Call Control Options. To configure these settings for the ShoreTel system, log into ShoreWare Director and select "Administration" then "Call Control" followed by "Options" (Figure 1).

Figure 1 – Administration Call Control Options

The “Call Control Options” screen will then appear (Figure 2).

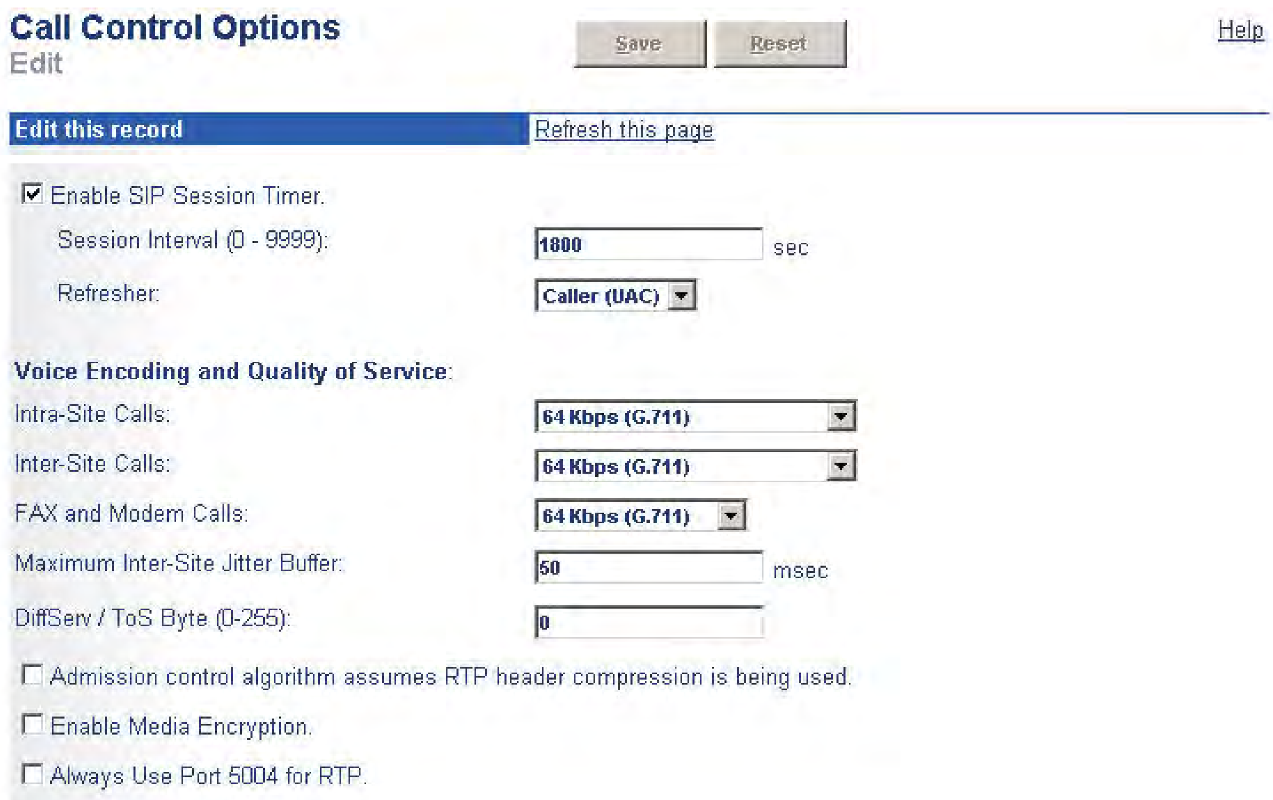


Figure 2 – Call Control Options

Within the “Call Control Options” screen confirm the appropriate settings are made for the “Enable SIP Session Timer”, “Intra-Site Calls”, “Inter-Site Calls” and “Always Use Port 5004 for RTP” fields.

The first step is to ensure the box is checked for the “Enable SIP Session Timer”. Next the Session Interval Timer needs to be set. The recommended setting for “Session Interval” is 1800 seconds. The last item to select is the appropriate refresher (from the pull down menu) for the SIP Session Timer. The “Refresher” field will be set either to “Caller (UAC)” [User Agent Client] or to “Callee (UAS)” [User Agent Server]. If the “Refresher” field is set to “Caller (UAC)” the Caller’s device will be in control of the session timer refresh. If “Refresher” is set to “Callee (UAS)” the device of the person called will control the session timer refresh.

The next settings to verify are the “Intra-Site Calls” and the “Inter-Site Calls” settings under the” Voice Encoding and Quality of Service” prompt. For the Intra-Site Calls verify the desired audio bandwidth is selected for the CODEC for calls within the system. The settings should then be confirmed for the desired audio bandwidth for the CODEC for Inter-Site calls (calls between sites).

Note: SIP uses both G.711 and G.729 CODECs. The CODEC setting will be negotiated to the highest CODEC supported.

Un-checking the box for “Always Use Port 5004 for RTP” is required for implementing SIP on the ShoreTel system. For SIP configurations, Dynamic UDP must be used for RTP Traffic. Note: If the box is unchecked MGCP will no longer use UDP port 5004; MGCP and SIP traffic will use dynamic UDP ports.

Sites Settings:

The next settings to address are the administration of sites. These settings are modified under the ShoreWare Director by selecting "Administration" then "Sites" (Figure 3).

Figure 3 –Administration Site

This selection brings up the "Sites" screen. Within the "Sites" screen select the name of the site to configure. The "Edit Site" screen will then appear. The only change required to the "Edit Site" screen is to the "Admission Control Bandwidth" field (Figure 4).

Figure 4 – Admission Control Bandwidth

The Admission Control Bandwidth defines the bandwidth available to and from the site. This is important as SIP devices will be counted against the site bandwidth. See the ShoreTel Planning and Installation Guide for more information.

Switch Settings - Allocating Ports for SIP Trunks:

The final general settings to make are to the ShoreGear switch. These changes are modified by selecting "Administration" then "Switches" in ShoreWare Director (Figure 5).

Figure 5 – Administration Switches

This action brings up the "Switches" screen. From the "Switches" screen simply select the name of the switch to configure. The "Edit ShoreGear …Switch" screen will be displayed. Within the "Edit ShoreGear …Switch" screen select the desired number of SIP trunks from the ports available (Figure 6).

Figure 6 – ShoreGear Switch Settings

ShoreTel System Settings – Trunk Groups: ShoreTel Trunk

Groups support both Dynamic and Static SIP end point Individual Trunks.

Note: *A ShoreGear switch can only support one Trunk Group with Dynamic IP addressing.*

In trunk planning a couple of things need to be considered.

1. Are the SIP devices using DHCP or Static IP?
2. Are the SIP devices endpoints (like ATAs, Conference Phone or WiFi handset) or non-endpoint devices like an ISP?

If the SIP Trunk Groups have already been configured on the system, skip down to the "ShoreTel System Settings - Individual Trunks" section. The settings for Trunk Groups are changed by selecting "Administration" then "Trunks" followed by "Trunk Groups" within ShoreWare Director (Figure 7).

Figure 7 – Administration Trunk Groups

This selection brings up the "Trunk Groups" screen (Figure 8).

Figure 8 – Trunk Groups Settings

From the pull down menus on the "Trunk Groups" screen select the site desired and select the "SIP" trunk type to configure and click on the "Go" link. The "Edit SIP Trunk Group" screen" will appear (Figure 9)

Figure 9 – SIP Trunk Groups Settings

Determine whether the trunks need to be configured as inter-site trunks (trunks between sites) or intra-site trunks (trunks within a site). In the "Edit SIP Trunks Group" screen, input a name for the trunk group. When naming the trunk group it is recommended to include the trunk type and the extension or extension range of the trunk group being established. Having this detail in the name is a convenient way to recall the trunk type and extension range. In the example in Figure 9 the name "SIP x3000-3019 Intra-site" has been created. This name indicates that a SIP Intra-site trunk with twenty off system extensions in the range of 3000-3019 has been created.

The next step is to choose the setting of the "Teleworker" check box.

- By NOT checking Teleworker no bandwidth usage will count against the site and the Intra-Site codec will be used when calls are made Intra-Site.
- If the Teleworker checkbox "is" checked then the call will be counted against the allocation of bandwidth and the Inter-Site codec will be used.
- Should Teleworker not be checked and the call goes from San Jose (location of the SIP Trunk) to another site in NY then Inter-Site will apply.
- Note: Location of the SIP Trunk NOT the SIP device is what's used to determine the location of the call.

The next item on this screen is the "Enable Digest Authentication" field. This is a SIP feature that allows a user ID and password to be established for authentication. If this feature is configured for the Trunk Group the same settings must be configured on the SIP endpoint devices.

The "Enable SIP Info for G.711 DTMF Signaling" box should not be checked. Enabling SIP info is currently only used with tie trunks between ShoreTel systems.

The next item to change for the SIP trunk groups is the handling of the inbound trunk configuration (Figure 10).

Figure 10 – Inbound Trunk Configuration

For the Inbound Trunk Settings ensure the "Number of Digits from CO" is correct. Select the "DNIS" (Dialed Number Identification Service) box to create a DNIS to extension mapping. Select the "DID" (Direct Inward Dial) box to input a DID range. The next step is to make sure the "Extension" box is checked along with the "Translation Table" button along with the translation table from the pull down menu. The last step for the Inbound Trunk configuration is to select the "Tandem Trunking" box if needed. Selecting this box will allow trunking between switches. The "Destination:" contains the number for the Auto Attendant. Trunk calls not routed by previous entries under the Inbound Trunk routing will be routed to the Auto Attendant.

The next item to change for the SIP trunk groups is the handling outbound trunk configuration (Figure 11).

For the Outbound Trunk Configuration set the "Access Code" and "Local Area Code" fields for the system. For the "Trunk Services" boxes, typically none of these will be selected.

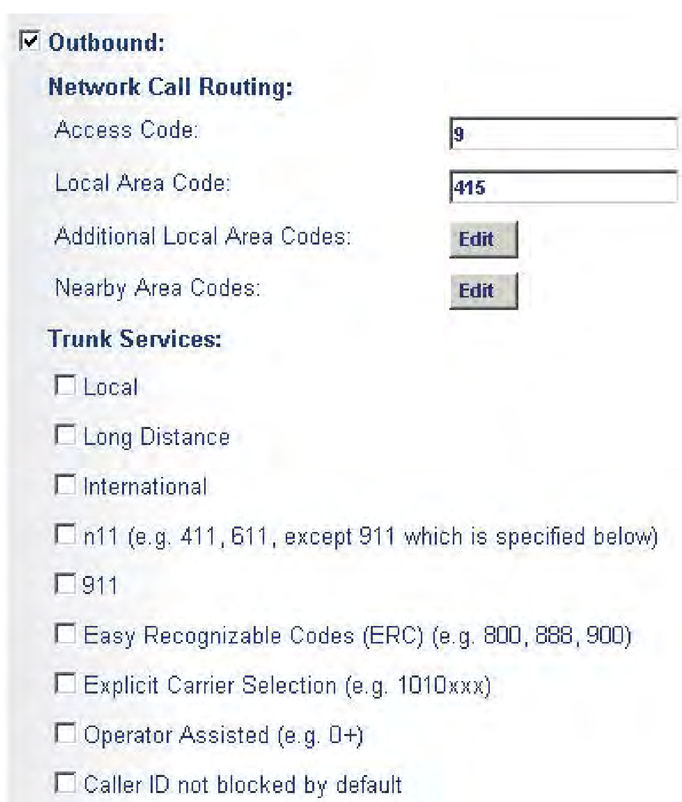


Figure 11 – Outbound Trunk Configuration

The next item to change for the SIP Trunk Groups is the Off System Extensions (Figure 12).

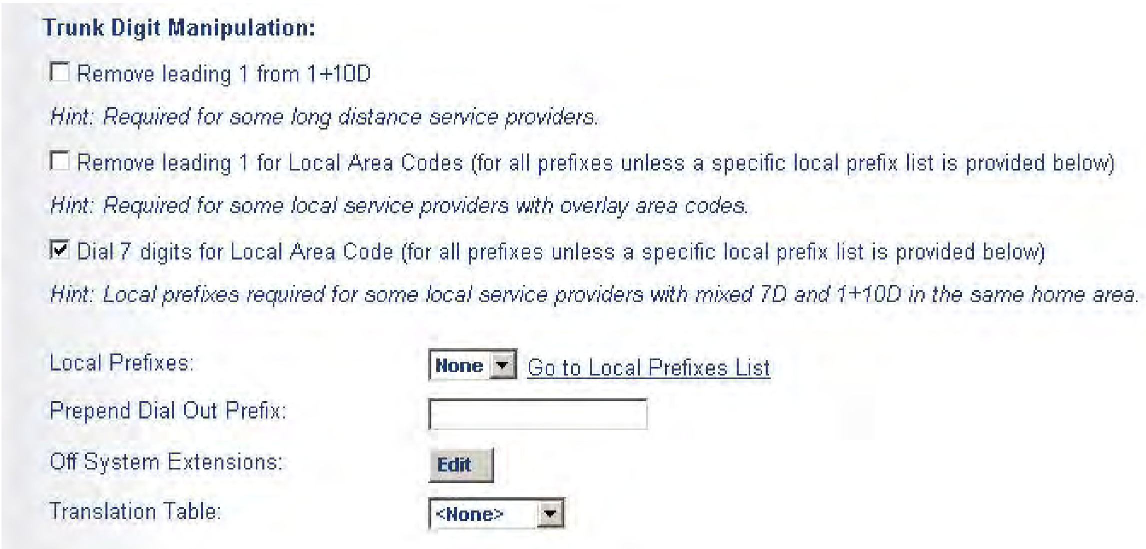


Figure 12 – Off System Extensions

Selecting the "Edit" button for the Off System Extensions will bring up the "Off System Extension Ranges" dialog box (Figure 13).

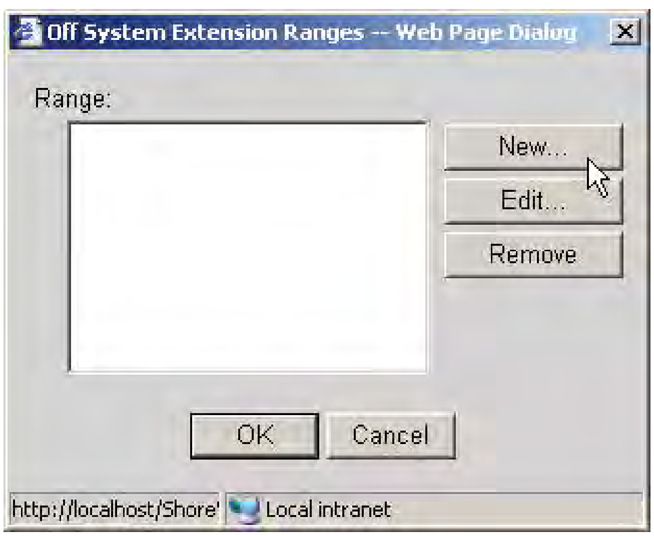


Figure 13 – Off System Extension Ranges

Selecting the "New" button will bring up the "New Range" dialog box (Figure 14).

Figure 14 – New Range Settings

Input a new range of off system extensions for the SIP devices and click "OK". For this example the extension range 3000-3019 has been input, which corresponds to the naming convention used in the previous example. This completes the settings needed to set up the SIP Trunk Groups on the ShoreTel system.

Note: *It is important to note that only one dynamic trunk group can be set up per ShoreGear switch. If another dynamic trunk group is desired another ShoreGear switch will be needed.*

ShoreTel System Settings – Individual Trunks:

This section covers the configuration of the individual trunks. Select "Administration" then "Trunks" followed by "Individual Trunks" to configure the individual trunks (Figure 15).

Figure 15 – Individual Trunks

The "Trunks by Group" screen, used to change the individual trunks settings, then appears (Figure 16).

Figure 16 – Trunks by Group

Select the site for the new individual trunk(s) to be added and select the appropriate trunk group from the pull down menu. In this example the site is "Headquarters" and the trunk group is SIP x3020 Intra-site". Click on the "Go" button to bring up the "Edit Trunk" screen (Figure 17).

Figure 17 – Edit Trunks Screen for Individual Trunks

From the individual trunks "Edit Trunk" screen, input a name for the individual trunks, select the appropriate switch, select the SIP trunk type and input the number of trunks. When selecting a name, the recommendation is to name the individual trunks the same as the name of the trunk group so that the trunk type and extension range can easily be tracked. Select the switch upon which the individual trunk will be created. For the SIP Trunk Type decide whether the trunks are to be configured as dynamic or static. Dynamic trunks are typically configured for endpoint devices like wireless handsets, conference phone or Integrated Access Devices (IADs). For the "SIP Trunk Type" field select either "Dynamic" or for a static configuration select "Use IP Address" button and input an IP address. In this example a dynamic SIP trunk type has been chosen. The last step is to select the number of individual trunks desired. In the example 20 trunks were chosen, which matches the naming convention used. Once these changes are complete, select the "Save" button to create the list of individual trunks (Figure 18).

Trunks by Group Help

Add new trunk at site: Headquarters in trunk group: SIPx3000-3019 Intra-site Go

Name/Number	Group	Site	Switch	Port/Channel	IsDigital	TrunkTypeID	DeviceIPAddress
SIPx3000-3019 Intra-site	SIPx3000-3019 Intra-site	Headquarters	SG-40/8	0	1	6	0.0.0.0
SIPx3000-3019 Intra-site (1)	SIPx3000-3019 Intra-site	Headquarters	SG-40/8	0	1	6	0.0.0.0
SIPx3000-3019 Intra-site (10)	SIPx3000-3019 Intra-site	Headquarters	SG-40/8	0	1	6	0.0.0.0
SIPx3000-3019 Intra-site (11)	SIPx3000-3019 Intra-site	Headquarters	SG-40/8	0	1	6	0.0.0.0
SIPx3000-3019 Intra-site (12)	SIPx3000-3019 Intra-site	Headquarters	SG-40/8	0	1	6	0.0.0.0
SIPx3000-3019 Intra-site (13)	SIPx3000-3019 Intra-site	Headquarters	SG-40/8	0	1	6	0.0.0.0
SIPx3000-3019 Intra-site (14)	SIPx3000-3019 Intra-site	Headquarters	SG-40/8	0	1	6	0.0.0.0
SIPx3000-3019 Intra-site (15)	SIPx3000-3019 Intra-site	Headquarters	SG-40/8	0	1	6	0.0.0.0
SIPx3000-3019 Intra-site (16)	SIPx3000-3019 Intra-site	Headquarters	SG-40/8	0	1	6	0.0.0.0
SIPx3000-3019 Intra-site (17)	SIPx3000-3019 Intra-site	Headquarters	SG-40/8	0	1	6	0.0.0.0
SIPx3000-3019 Intra-site (18)	SIPx3000-3019 Intra-site	Headquarters	SG-40/8	0	1	6	0.0.0.0
SIPx3000-3019 Intra-site (19)	SIPx3000-3019 Intra-site	Headquarters	SG-40/8	0	1	6	0.0.0.0
SIPx3000-3019 Intra-site (2)	SIPx3000-3019 Intra-site	Headquarters	SG-40/8	0	1	6	0.0.0.0
SIPx3000-3019 Intra-site (3)	SIPx3000-3019 Intra-site	Headquarters	SG-40/8	0	1	6	0.0.0.0
SIPx3000-3019 Intra-site (4)	SIPx3000-3019 Intra-site	Headquarters	SG-40/8	0	1	6	0.0.0.0
SIPx3000-3019 Intra-site (5)	SIPx3000-3019 Intra-site	Headquarters	SG-40/8	0	1	6	0.0.0.0
SIPx3000-3019 Intra-site (6)	SIPx3000-3019 Intra-site	Headquarters	SG-40/8	0	1	6	0.0.0.0
SIPx3000-3019 Intra-site (7)	SIPx3000-3019 Intra-site	Headquarters	SG-40/8	0	1	6	0.0.0.0
SIPx3000-3019 Intra-site (8)	SIPx3000-3019 Intra-site	Headquarters	SG-40/8	0	1	6	0.0.0.0
SIPx3000-3019 Intra-site (9)	SIPx3000-3019 Intra-site	Headquarters	SG-40/8	0	1	6	0.0.0.0

Figure 18 – Trunks by Group

Note: *Individual SIP trunks cannot span networks. SIP trunks can only terminate on the switch selected. There is no failover to another switch.*

After setting up the trunk groups and individual trunks refer to the ShoreTel Product Installation Guide to make the appropriate changes for the User Group settings. This completes the settings for the ShoreTel system side.

Polycom Soundstation IP 4000 Configuration

To setup the Polycom Soundstation IP 4000 conference phone to operate on the ShoreTel system the following configuration steps need to be made from the menus within the Polycom conference phone:

1. Press the "Menu" button on the conference phone to enter the "Main Menu" screen.
2. Scroll down using the down arrow key to item "3. Settings…" and press the "Select" key to enter the "Settings" screen.
3. Scroll down using the down arrow key to item "6. SIP Configuration…" and press the "Select" key.
4. On the "Authentication(1/Ascii)" screen for the "Admin Password:" prompt enter the password (the default is 456) and press the softkey under the "Enter" text that appears on the LCD display. This will bring up the "SIP Configuration" screen.
5. On the "SIP Configuration" screen for "Server Address:" press "Select" and press the "1/A/a" softkey until the screen title "SIP Configuration(1/Ascii) appears.

6. Enter the IP address for the ShoreGear Switch using the keypad (use "*" key for the periods in the IP address) and press "Select" to enter.
7. Scroll down using the down arrow key to item "Line 1" and press the "Select" key to enter the "Line 1" screen.
8. On the "Line 1" screen for the "Display Name:" prompt press "Select" to enter the display name.
9. Enter the desired display name (example: 3000) and press "Select" to input the changes. Use the "1/A/a" softkey to toggle between text and numbers.
10. Scroll down using the down arrow key to item "Address:" and press the "Select" key.
11. Enter the desired address (example: 3000) and press "Select" to input the changes. Use the "1/A/a" softkey to toggle between text and numbers.
12. Scroll down using the down arrow key to item "Label:" and press the "Select" key to enter the Label.
13. Enter the desired label (example: 3000) and press "Select" to input the changes. Use the "1/A/a" softkey to toggle between text and numbers.
14. If using an Authorization Id and Password (Optional), scroll down to the prompts "Auth User ID:" and "Auth Password" and enter a User ID and Password (using same text input procedure as in previous steps).
15. Press "Exit" key twice to enter "Network Configuration" screen
16. Scroll to item "2-Save Config" and press "Select". The changes made will then be saved.

This completes the configuration needed for the Polycom IP 4000 Conference Phone.

Note: *For forwarding to function, the full SIP address must be entered (e.g. sip:123@1.2.3.4). Extended forwarding features (Forward on busy, no answer, or unconditional) can be found in the HTTP admin (http://<phone ip address> . Login=admin, Password=456).*

Record of Change

This application note is subject to change. Updates and corrections are always welcome. Please submit any updates or corrections to info@shoretel.com.

Issue	Author	Reason for Change	Date
1.0	J. Casselman	Initial Release	December 14, 2006

ShoreTel™

960 Stewart Drive Sunnyvale, CA 94085 USA Phone +1.408.331.3300 +1.877.80SHORE Fax +1.408.331.3333 www.shoretel.com