ShoreTel Innovation Network

bandwidth

SHORETEL APPLICATION NOTE

for

Bandwidth SIP Trunking

Date:	January 19, 2016
App Note Number:	16005
For use with:	Bandwidth SIP Trunking
Product:	ShoreTel Connect Onsite
System:	ShoreTel Connect Onsite Build 21.73.9904.0

This document and its contents was prepared by:

tekVizion LABS

Contents

Audience

This document is intended for the SIP Trunk Customer's technical staff and Value Added Reseller (VAR) having installation and operational responsibilities

Introduction

This Configuration Guide describes configuration steps for Bandwidth Business SIP Trunking to ShoreTel Connect Onsite System

Bandwidth

Bandwidth SIP trunking is a scalable and efficient IP trunking telecommunication solution for your business that provides all the traditional services such as Direct Inward Dialing, Hunting, Calling Name, Calling Number, Local/Long Distance and Business Continuity options, including:

- Burstable Trunk Capacity – Dynamically increases call capacity during peak busy periods so your customers never receive a busy signal
- Call Forward Always – On the trunk group pilot number for all calls in case of an outage (i.e., flood, fire, loss of power, etc.)
- Call Forward Not Reachable – On the trunk group pilot number that operates on a per-call contingency basis to forward the call to any PSTN number (i.e., call center or alternate office location) during temporary call completion impairments
- Route Exhaustion – Automatic reroute of trunk group calls to any PSTN phone number (i.e., a call center) if calls can't be completed to the PBX
- Support for geo-redundant PBX deployments and automatic reroute of SIP trunks to the backup customer data center

For Bandwidth sales or support, please visit
http://www.bandwidth.com/contact

SIP Trunking Network Components

The network for the SIP Trunk reference configuration is illustrated below and is representative of a ShoreTel Connect Onsite System configuration.

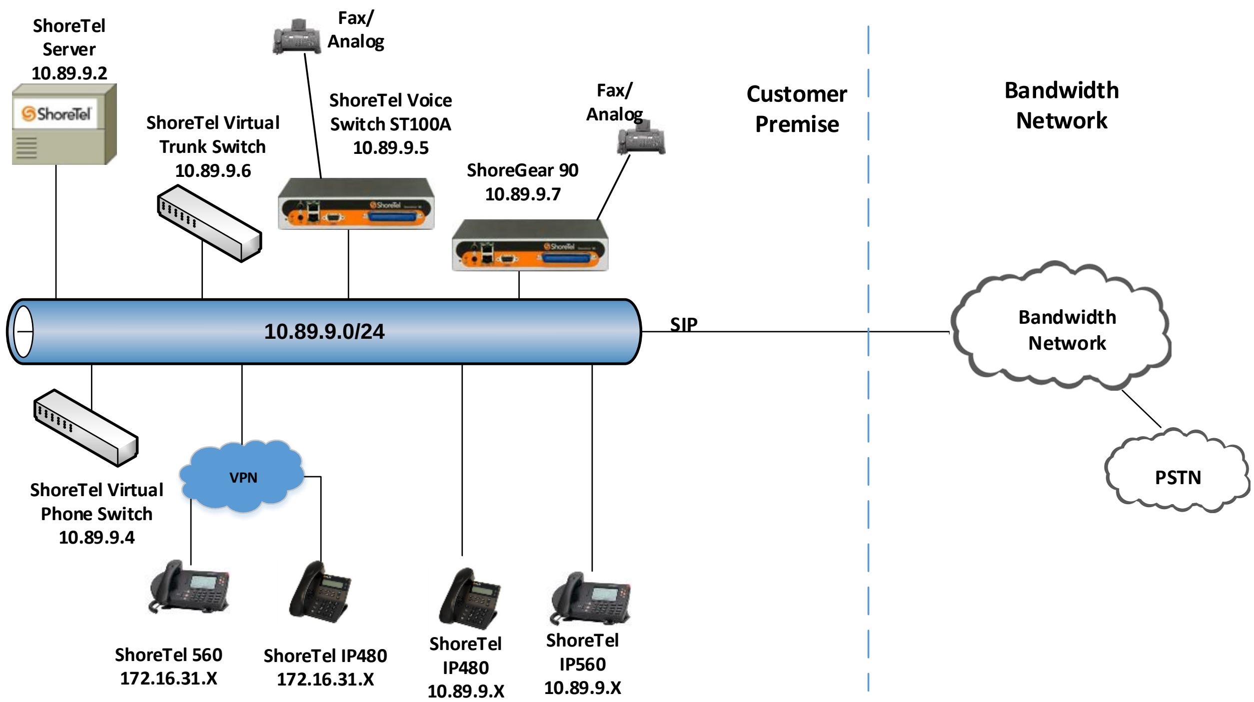

Figure 2: SIP Trunk Lab Reference Network

Hardware Components

- ShoreTel Server
- ShoreTel Voice Switch ST100A
- ShoreGear 90
- Analog Fax Machine
- ShoreTel 560/565 IP Phones
- ShoreTel 485G IP Phone (SIP Ref)
- ShoreTel Virtual Phone Switch
- ShoreTel Virtual Trunk Switch

Software Requirements

- ShoreTel Connect Onsite Build 21.73.9904.0
- Windows Server 2012 R2

Features

SIP Registration Method

This test used a Static Registration Method between the ShoreTel Connect Onsite PBX and Bandwidth. There is no registration requirement for the ShoreTel Connect Onsite PBX.

Features Supported

- Basic calls with G711Ulaw and G729
- Call Hold and Resume
- MOH-Music On Hold
- Call Transfer
- Call Forwarding
- Three-Way Calling
- DTMF RFC 2833
- Calling Party Number Presentation
- Calling Party Number Restricted
- Hunt Group
- Group Pickup
- Call Park/Unpark
- Simul Ring
- Call Forward – “FindMe”
- Call Recording
- Auto-Attendant
- Bridge Call Appearances(BCA)
- Work Group
- Office Anywhere External
- Silent Monitor / Barge-In / Whisper Page
- Fax (T.38 & G711)

Features Not Supported

- Operator Assistance services 0+ are not supported by Bandwidth

Caveats and Limitations

- Bandwidth uses G711 as the preferred codec for both Inbound and Outbound Calls.
- ShoreTel Service Appliance was not tested in the current testbed due to the requirement of additional hardware (This feature is not covered in this configuration guide)
- Contact Center was not tested in the current test-bed due to the requirement of additional hardware (This feature is not covered in this configuration guide)
- "SIP Media Proxy" is required to provide feature parity of PRI Trunks with SIP Trunks. This includes features like Office Anywhere, Simultaneous Ringing, 3-way Mesh Conferencing, Call Recording, Silent monitoring, Barge-In, Whisper Page etc. It is enabled by default on ShoreTel Virtual Trunk switches, but needs to be assigned manually on new ShoreTel Voice Switches as well as legacy half-width ShoreGear Switches. For further information on "SIP Media Proxy" please refer to Chapter 19 of the ShoreTel Connect Administration Guide.

NOTE*: There may be other feature limitations when using SIP Trunks. Please refer to Chapter 19 of the* ***ShoreTel Connect Administration Guide*** *for more information.*

Configuration

Configuration Steps

Overview of the steps that are required to configure a ShoreTel PBX for SIP Trunking to Bandwidth

Table 1 – PBX Configuration Steps

Step	Description

Step 1	Codec Lists and Sites
Step 2	SIP Trunk Configuration

IP Address Worksheet

The specific values listed in the table below and in subsequent sections are used in the lab configuration described in this document, and are for **illustrative purposes only**. The customer must obtain and use the values for your deployment.

Table 2 – IP Address Worksheet

Component	MSO Lab Value	Customer Value
ShoreTel Connect Onsite IP-PBX		
ShoreTel Director	10.89.9.2	Unique to every deployment
ShoreTel Voice Switch ST100A	10.89.9.5	Unique to every deployment
ShoreTel Virtual Trunk Switch	10.89.9.6	Unique to every deployment
ShoreTel Virtual Phone Switch	10.89.9.4	Unique to every deployment

Create Custom Codec Lists and Sites

Create Codec Lists

1. Navigate to **Features > Call Control > Codec Lists**
2. Click **NEW**
3. Set **Description**: Bandwidth is used for this example
4. **Codec List Members**: PCMU/8000 & G729/8000 was chosen and added for this test
5. Click **SAVE**

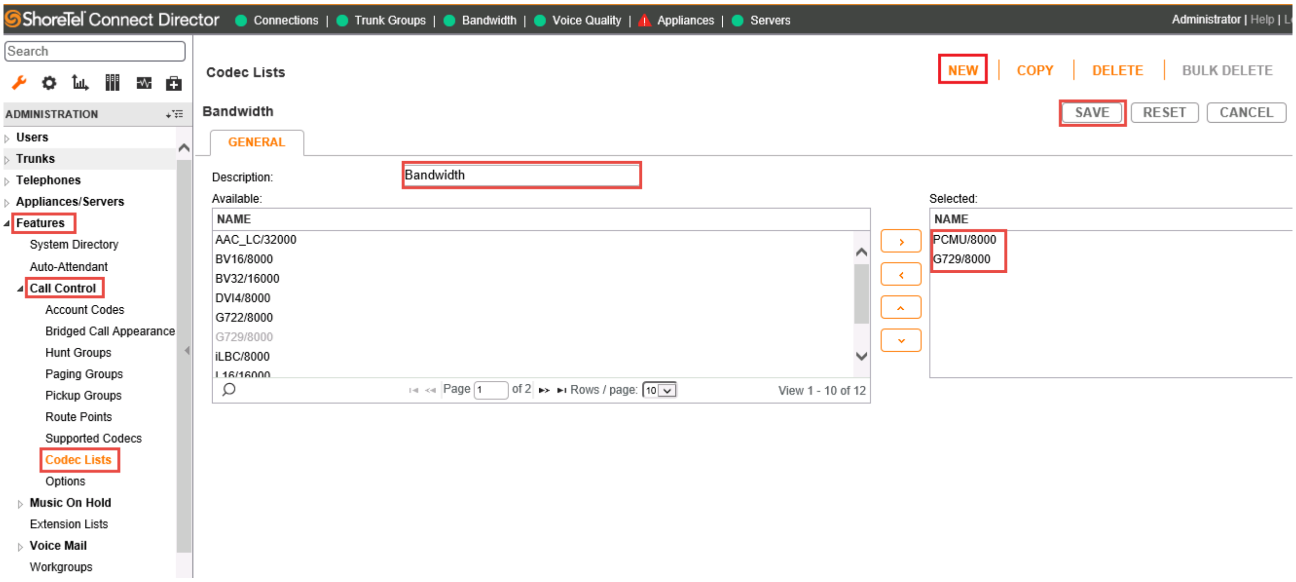

Figure 3: Codec Lists

Create Sites

1. Navigate to **System** > **Sites**
2. Set **Name**: Headquarters
3. Set **Local Area Code**: 972 is used in this test
4. **Intra-Site Calls**: The newly created Codec List, **Bandwidth,** which contains only G711 & G729 codecs is selected in this example
5. **Inter-Site Calls**: Codec List **Bandwidth** is selected from the drop down menu
6. **Fax and Modem Calls**: The default Codec List, **Fax Codecs – Low Bandwidth Passthrough,** is selected from the drop down menu
7. Leave all other fields as default
8. Click **SAVE**

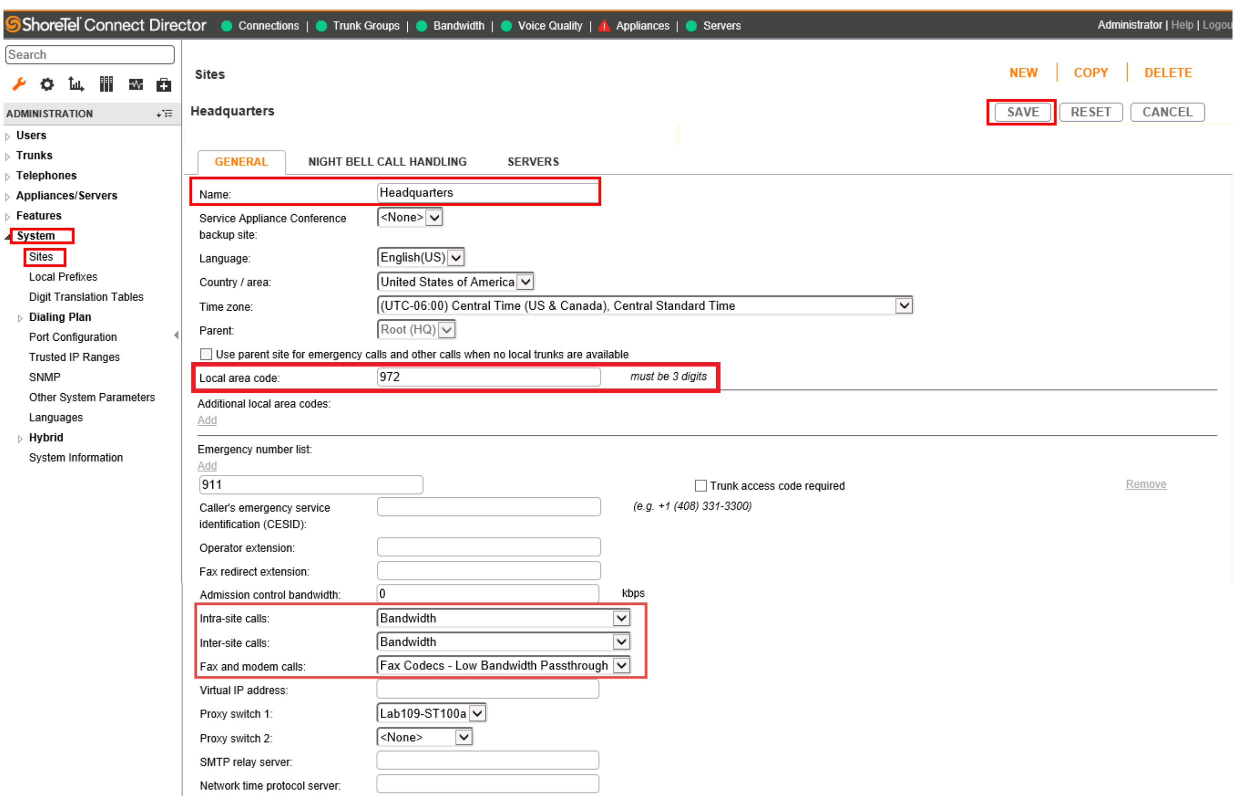

Figure 4: Create Sites

SIP Trunk Configuration

This section describes the ShoreTel configuration necessary to support connectivity to the Bandwidth SIP Trunking service.

Create SIP Trunk Profiles

SIP Profile is critical for SIP Trunking deployment. The **Default ITSP** SIP Profile is copied to create the SIP Profile for this test.

1. Navigate to **Trunks > SIP Profiles**
2. Check **Default ITSP** under **NAME**
3. Click **COPY**

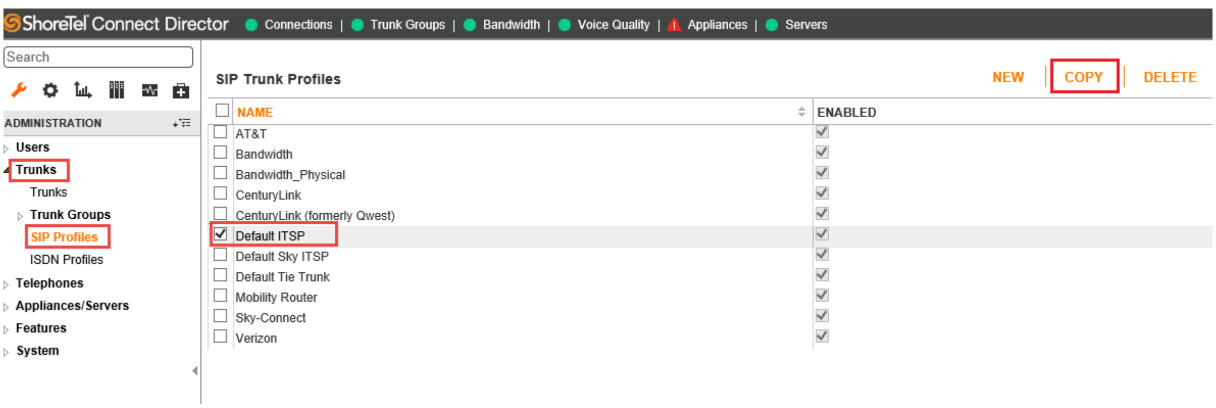

Figure 5: SIP Trunk Profiles

4. Set **Name**: Change from *Default ITSP* to *Bandwidth*
5. Leave all other fields as default
6. Click **SAVE**

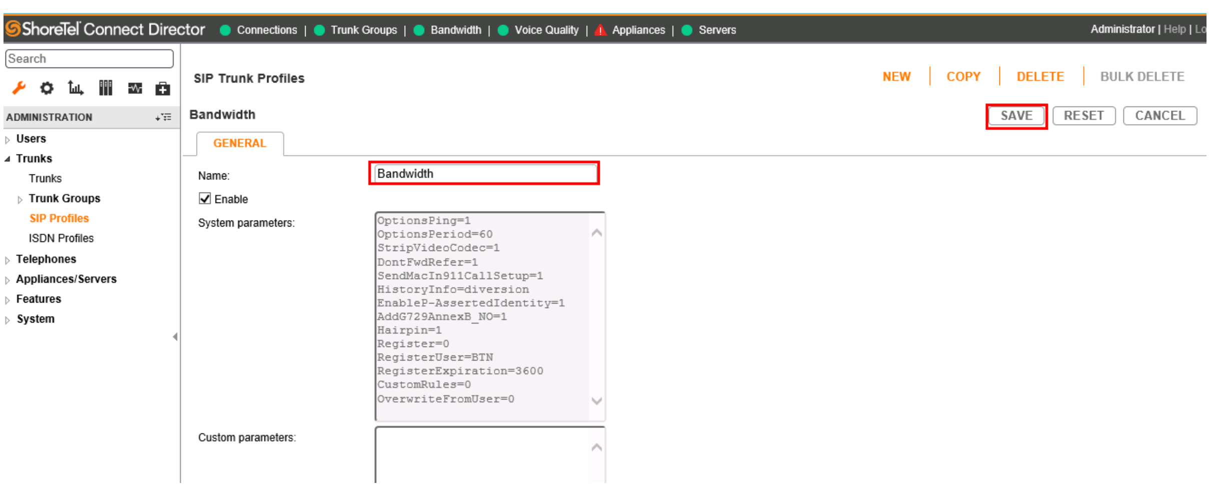

Figure 6: SIP Profile - Cont.

Add Trunk Group

1. Navigate to **Trunks > Trunk Groups > Trunk Groups**
2. Set **Name**: Bandwidth
3. Set **Trunk Type**: SIP
4. Set **Profile**: SIP Profile *Bandwidth,* created in previous step, is selected from drop down
5. Set **Digest Authentication**: -None- is selected from drop down
6. Set **Username**: The Trunk Group Pilot Number (username) will be provided by your Bandwidth Account Representative
7. Set **Password**: Password will be provided by your Bandwidth Account Representative and must be kept confidential
8. Click **SAVE**

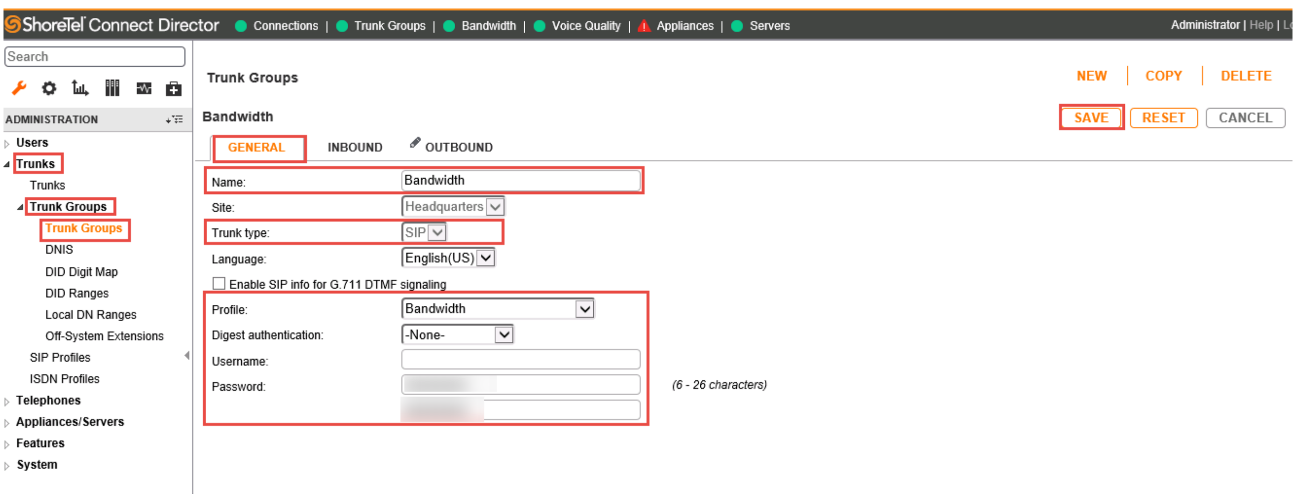

Figure 7: Trunk Groups

9. Go to the **INBOUND** tab
10. Set **Number of Digits from CO**: 12 is used in this setup
11. **DNIS**: *Unchecked*
12. **DID**: Checked
13. Click **SAVE**

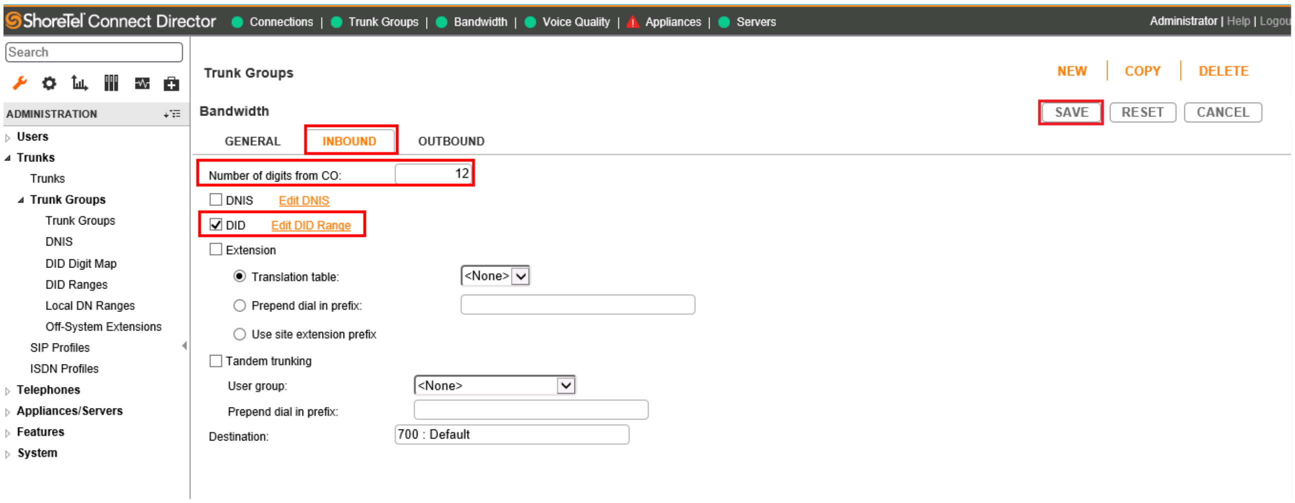

Figure 8: Trunk Groups - Cont.

14. Select the **OUTBOUND** tab
15. **Outgoing**: Checked
16. Set **Access Code**: 9 is used in this example
17. Set **Local Area Code**: 972 is used in this example
18. Set **Billing Telephone Number**: Pilot number will be provided by your Bandwidth Account Representative and must be kept confidential
19. Scroll down to the bottom of the screen

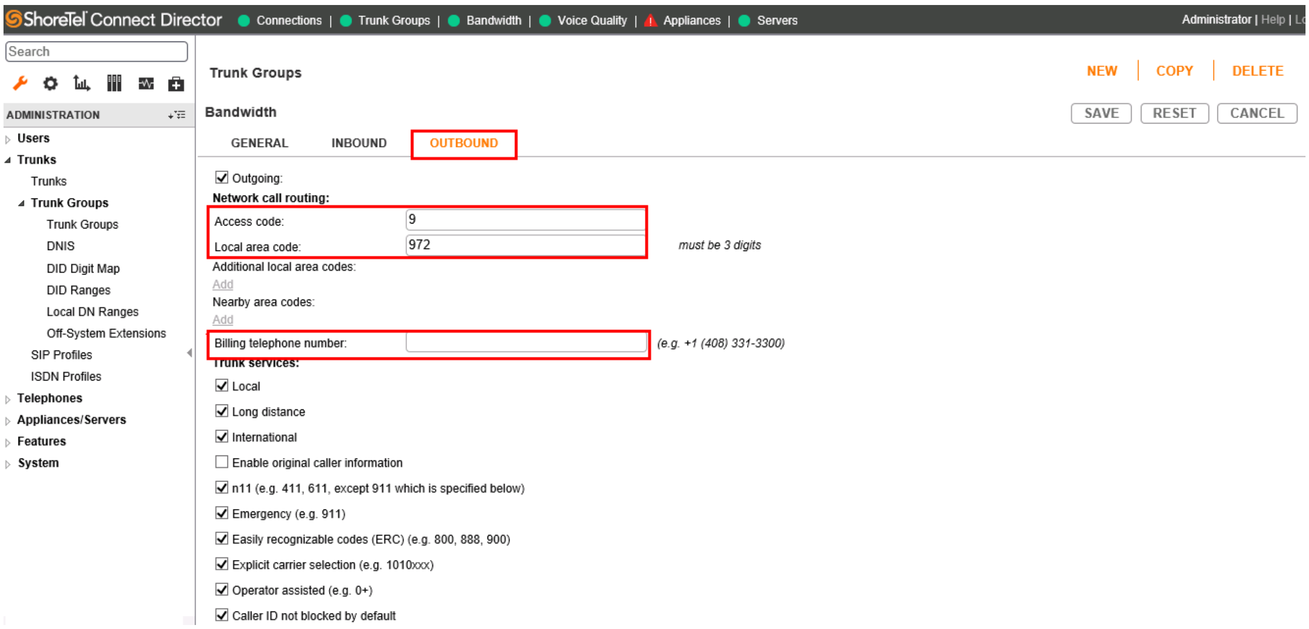

Figure 9: Trunk Groups - Cont.

20. **Dial 7 digits for Local Area Code**: Checked
21. Leave all other fields as default
22. Click **SAVE**

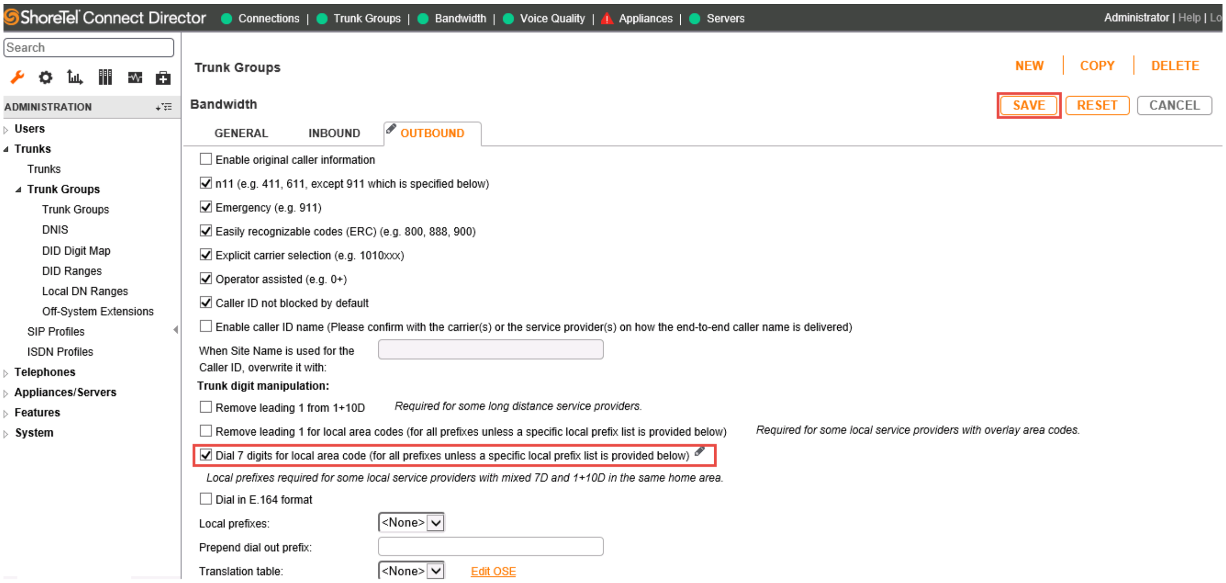

Figure 10: Trunk Groups - Cont.

Create Individual Trunks

1. Navigate to **Trunks > Trunks**
2. Set **Trunk Group**: Bandwidth (SIP)
3. Set **Name**: Bandwidth is used in this setup
4. Set **Switch**: Lab109- vTS1 is selected
5. Set **IP Address or FQDN**: Enter the IP Address or FQDN of Bandwidth
6. Click **SAVE**

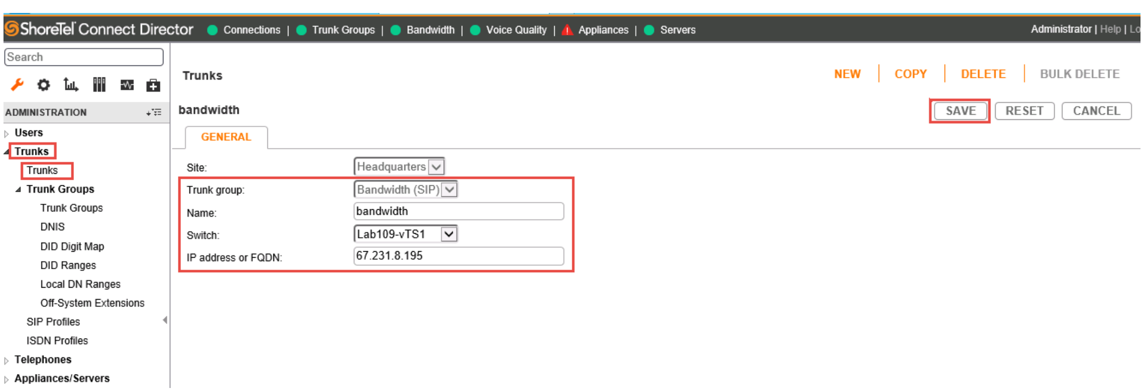

Figure 11: Individual Trunks

Trusted IP Range

1. Navigate to **System > Trusted IP Range**
2. Click **NEW**
3. Set **Name**: Bandwidth is used in this setup
4. Set **Low IP Address**: Enter ITSP SIP/Media Gateway
5. Set **High IP Address**: Enter last IP in Range here.
6. Click **SAVE**

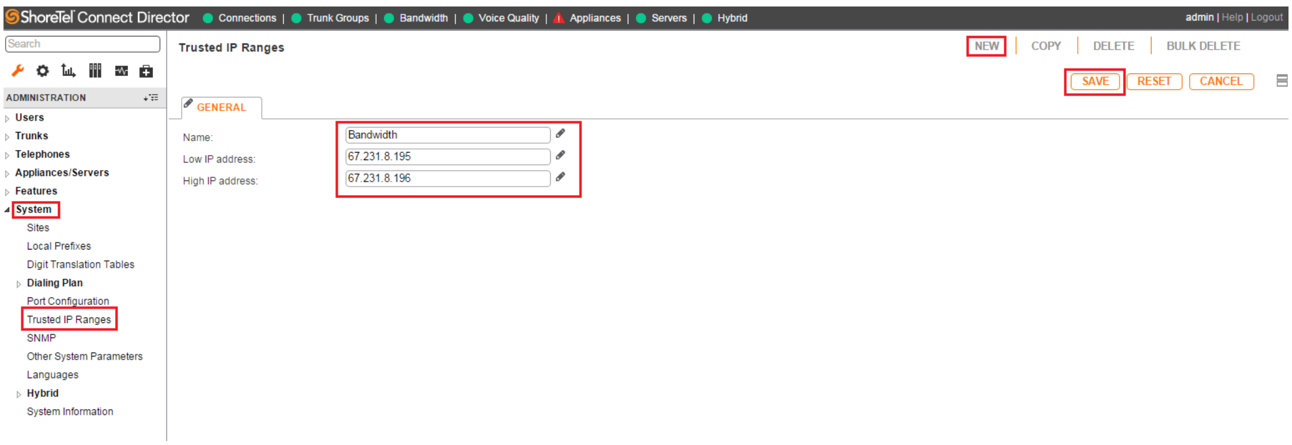

Figure 12: Trusted IP Range

Summary of Tests and Results

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

Primary Switch Test Plan (ShoreTel Virtual Trunk Switch)

ID	Result	Name	Description	Notes
1.1	PASS	Setup and Initialization	Verify successful setup and initialization of the SUT	
1.2	PASS	Outbound Call (Domestic)	Verify calls outbound placed through the SUT reach the external destination	
1.3	PASS	Inbound Call (Domestic)	Verify calls received by the SUT are routed to the default trunk group destination	
1.4	PASS	All Trunks Busy – Inbound Callers	Verify an inbound callers hears busy tone when all channels/trunks are in use	
1.5	PASS	All Trunks Busy – Outbound Callers	Verify an outbound callers hears busy tone when all channels/trunks are in use	
1.6	PASS	Incomplete Inbound Calls	Verify proper call progress tones are provided and proper call teardown for incomplete inbound calls	

ID	Result	Name	Description	Notes
2.1	G711 PASS – Virtual Switch FAIL – Physical Switch G729 N/S	Codec Negotiation	Verify codec negotiation between the SUT and the calling device with each side configured for a different codec	G711 - Physical Switch always sends out Codec list with order 0, 8, 18 even when the codec is configured with G711 only or G729 is setup as preferred codec for the site. This is a known issue at ShoreTel. The ShoreTel BUG ID is ENG-349616. The known issue will be fixed in a future release) G729 - Bandwidth supports both G711 and G729 codecs. G711 is the preferred codec

ID	Result	Name	Description	Notes
2.2	G711 PASS G729 N/S	DTMF Transmission – Out of Band/ In Band	Verify transmission of in-band and out-of-band digits per RFC 2833 for various devices connected to the SUT	Bandwidth supports both G711 and G729 codecs. G711 is the preferred codec
2.3	PASS	Auto Attendant Menu	Verify that inbound calls are properly terminated on the ShoreTel Auto Attendant menu and that you can transfer to the desired extension	
2.4	PASS	Auto Attendant Menu checking Voicemail mailbox	Verify that inbound calls are properly terminated on the ShoreTel Auto Attendant menu and that you can transfer to the Voicemail Login Extension	

ID	Result	Name	Description	Notes
3.1	PASS	Post Dial Delay	Verify that post dial delay is within acceptable limits	

ID	Result	Name	Description	Notes
4.1	PASS	Caller ID Name and Number - Inbound	Verify that Caller ID name and number is received from SIP endpoint device	
4.2	PASS	Caller ID Name and Number - Outbound	Verify that Caller ID name and number is sent from SIP endpoint device	
4.3	PASS	Hold from SUT to SIP Reference	Verify successful hold and resume of connected call	
4.4	PASS	Call Forward - SUT	Verify outbound calls that are being forwarded by the SUT are redirected and connected to the appropriate destination	
4.5	PASS	Call Forward – External PSTN Number	Verify outbound calls that are being forwarded by the SUT are redirected and connected to the appropriate destination	
4.6	PASS	Call Transfer – Blind	Verify a call connected from the SUT to the ShoreTel phone can be transferred to an alternate destination	
4.7	PASS	Call Transfer – Consultative	Verify a call connected from the SUT to the ShoreTel phone can be transferred to an alternate destination	
4.8	PASS	Outbound 911	Verify that outbound calls to 911 are routed to the correct PSAP for the calling location and that caller ID information is delivered	
4.9	N/S	Operator Assisted	Verify that 0+ calls are routed to an operator for calling assistance	Operator Assistance services 0+ are not supported by Bandwidth
4.10	PASS	Inbound / Outbound call with Blocked Caller ID	Verify that calls with Blocked Caller ID route properly and the answering phone does not display any Caller ID information	
4.11	G711 PASS G729 N/S	Inbound call to a Hunt Group	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	Bandwidth supports both G711 and G729 codecs. G711 is the preferred codec for Bandwidth.

ID	Result	Name	Description	Notes
4.12	G711 PASS G729 N/S	Inbound call to a Workgroup	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	Bandwidth supports both G711 and G729 codecs. G711 is the preferred codec for Bandwidth.
4.13	PASS	Inbound call to DNIS/DID and leave a voice mail message	Verify that inbound calls to a user, via DID/DNIS, routes to the proper user mailbox and a message can be left with proper audio	
4.14	PASS	Call Forward – “FindMe”	Verify that inbound calls are forwarded to a user’s “FindMe” destination	
4.15	G711 PASS T38 N/S	Inbound / Outbound Fax Calls	Verify that inbound / outbound fax calls complete successfully	Bandwidth supports T38 fax, but success rate is low for outbound fax to re-negotiate with T38 fax
4.17	PASS	Inbound call to Bridged Call Appearance (BCA) Extension	Verify that inbound calls properly presented to all of the phones that have BCA configured and that the call can be answered, placed on-hold and then transferred	
4.18	PASS	Inbound call to a Group Pickup Extension	Verify that inbound calls properly presented to all of the phones that have Group Pickup configured and that the call can be answered, placed on-hold and then transferred	
4.19	PASS	Office Anywhere External	Verify that inbound calls are properly presented to the Office Anywhere External PSTN destination	
4.20	PASS	Simul Ring	Verify that inbound calls are properly presented to the desired extension and the “Additional Phones” destinations	
4.21	PASS	MakeMe Conference	Verify that an inbound call can be conferenced with three (or more) additional parties	

ID	Result	Name	Description	Notes
4.22	PASS	Park / Unpark	Verify that an inbound call can be parked and unparked	
4.23	G711 PASS G729 N/S	Call Recording	Verify that external calls can be recorded via the SIP Trunk using ShoreTel Communicator	Bandwidth supports both G711 and G729 codecs. G711 is the preferred codec for Bandwidth.
4.24	G711 PASS G729 N/S	Silent Monitor / Barge-In / Whisper Page	Verify that external calls can be silently monitored, barged-in and whisper paged via the SUT	Bandwidth supports both G711 and G729 codecs. G711 is the preferred codec for Bandwidth.
4.25	PASS	Long Duration – Inbound	Verify that an inbound call is established for a minimum of 30 minutes	
4.26	PASS	Long Duration – Outbound	Verify that an outbound call is established for a minimum of 30 minutes	

ID	Result	Name	Description	Notes
5.1	N/A	Registration or Digest Authentication	Verify the SUT supports the use of registration or digest authentication for service access for inbound and outbound calls	Bandwidth does not require SIP registration and inbound/outbound call authentication

Secondary Switch Sanity Test Results (ShoreTel Voice Switch)

ID	Result	Name	Description	Notes
1.2	PASS	Outbound Call (Domestic)	Verify calls outbound placed through the SUT reach the external destination	
1.3	PASS	Inbound Call (Domestic)	Verify calls received by the SUT are routed to the default trunk group destination	
2.2	G711 PASS G729 N/S	DTMF Transmission – Out of Band / In Band	Verify transmission of in-band and out-of-band digits per RFC 2833 for various devices connected to the SUT	Bandwidth supports both G711 and G729 codecs. G711 is the preferred codec for Bandwidth.

4.5	PASS	Call Forward – External PSTN Number	Verify outbound calls that are being forwarded by the SUT are redirected and connected to the appropriate destination	
4.6	PASS	Call Transfer – Blind	Verify a call connected from the SUT to the ShoreTel phone can be transferred to an alternate destination	
4.7	PASS	Call Transfer – Consultative	Verify a call connected from the SUT to the ShoreTel phone can be transferred to an alternate destination	
4.12	G711 PASS G729 N/S	Inbound call to a Workgroup	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	Bandwidth supports both G711 and G729 codecs. G711 is the preferred codec for Bandwidth.
4.15	G711 PASS T38 PASS	Inbound / Outbound Fax calls	Verify that inbound / outbound fax calls complete successfully	Bandwidth supports T38 fax, but success rate is low for outbound fax to re-negotiate with T38 fax
4.16	N/T	ShoreTel Service Appliance Unified Communication System	Verify that inbound calls are properly forwarded to the ShoreTel Service Appliance and it properly accepts the access code and you're able to participate in the conference bridge	
4.21	PASS	MakeMe Conference	Verify that an inbound call can be conferenced with three (or more) additional parties	
4.23	G711 PASS G729 N/S	Call Recording	Verify that external calls can be recorded via the SIP Trunk using ShoreTel Communicator	Bandwidth supports both G711 and G729 codecs. G711 is the preferred codec for Bandwidth.
4.24	G711 PASS G729 N/S	Silent Monitor / Barge-In / Whisper Page	Verify that external calls can be silently monitored, barged-in and whisper paged via the SUT	Bandwidth supports both G711 and G729 codecs. G711 is the preferred codec for Bandwidth.

4.27	N/T	Contact Center	Verify that an inbound call can be established directly to the ShoreTel Contact Center, that all prompts are heard and the agent can answer the call	

Tertiary Switch Sanity Test Results (ShoreTel ShoreGear Switch)

ID	Result	Name	Description	Notes
1.2	PASS	Outbound Call (Domestic)	Verify calls outbound placed through the SUT reach the external destination	
1.3	PASS	Inbound Call (Domestic)	Verify calls received by the SUT are routed to the default trunk group destination	
2.2	G711 PASS G729 N/S	DTMF Transmission – Out of Band / In Band	Verify transmission of in-band and out-of-band digits per RFC 2833 for various devices connected to the SUT	Bandwidth supports both G711 and G729 codecs. G711 is the preferred codec for Bandwidth.
4.5	PASS	Call Forward – External PSTN Number	Verify outbound calls that are being forwarded by the SUT are redirected and connected to the appropriate destination	
4.6	PASS	Call Transfer – Blind	Verify a call connected from the SUT to the ShoreTel phone can be transferred to an alternate destination	
4.7	PASS	Call Transfer – Consultative	Verify a call connected from the SUT to the ShoreTel phone can be transferred to an alternate destination	
4.12	G711 PASS G729 N/S	Inbound call to a Workgroup	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	Bandwidth supports both G711 and G729 codecs. G711 is the preferred codec for Bandwidth.

4.15	G711 PASS T38 PASS	Inbound / Outbound Fax Calls	Verify that inbound / outbound fax calls complete successfully	Bandwidth supports T38 fax, but success rate is low for outbound fax to re-negotiate with T38 fax
4.21	PASS	MakeMe Conference	Verify that an inbound call can be conferenced with three (or more) additional parties	
4.23	G711 PASS G729 N/S	Call Recording	Verify that external calls can be recorded via the SIP Trunk using ShoreTel Communicator	Bandwidth supports both G711 and G729 codecs. G711 is the preferred codec for Bandwidth.
4.24	G711 PASS G729 N/S	Silent Monitor / Barge-In / Whisper Page	Verify that external calls can be silently monitored, barged-in and whisper paged via the SUT	Bandwidth supports both G711 and G729 codecs. G711 is the preferred codec for Bandwidth.
4.27	N/T	Contact Center	Verify that an inbound call can be established directly to the ShoreTel Contact Center, that all prompts are heard and the agent can answer the call	

Conclusion

Bandwidth SIP Trunking has been successfully tested with ShoreTel Connect Onsite Build 21.73.9904.0

Additional Resources

- ShoreTel Administration Guide

Version	Date	Contributor	Content
1.0	Nov 23, 2015	R. Jerome	Converted 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.

World Headquarters
960 Stewart Drive
Sunnyvale, CA 94085
USA
shoretel.com

+1 (800) 425-9385 Toll Free
+1 (408) 331-3300 Tel
+1 (408) 331-3333 Fax

EMEA
Inspired
Easthampstead Road
Bracknell, RG12 1YQ
+44 (0) 1344 208800 Tel

APAC
8 Temasek Boulevard#41-03
Suntec Tower 3
Singapore 038988
+65 6517 0800 Tel