SHORETEL APPLICATION NOTE

for

First Communications

Date: September 18, 2017

App Note Number: TC - 17049

For use with: First Communications Native SIP Trunking

Product: ShoreTel Connect ONSITE

System: ShoreTel Connect ONSITE
Build [21.82.2142.0]

This document and its contents was prepared by:

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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 Application Note describes the configuration steps required to configure a ShoreTel Connect ONSITE system with First Communications SIP Trunks.

First Communications

First Communications is a leading technology solutions provider offering data networking, voice, and managed services throughout the Midwest. Founded in 1998, First Communications network has grown to include more than 600 on-net wire centers and supports over 35,000 customers.

Headquartered in Akron, Ohio and a 24x7x365 Network Management Center in Chicago, First Communications is dedicated to pairing effective customer communications with next generation technology.

We create solutions to align with your business objectives, while our built-in scalability accommodates for the future. Combined with a strong focus on the customer experience and operational expertise, First Communications bridges technology with world class customer service.

To contact First Communications sales or support, please visit
http://www.firstcomm.com/about/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.

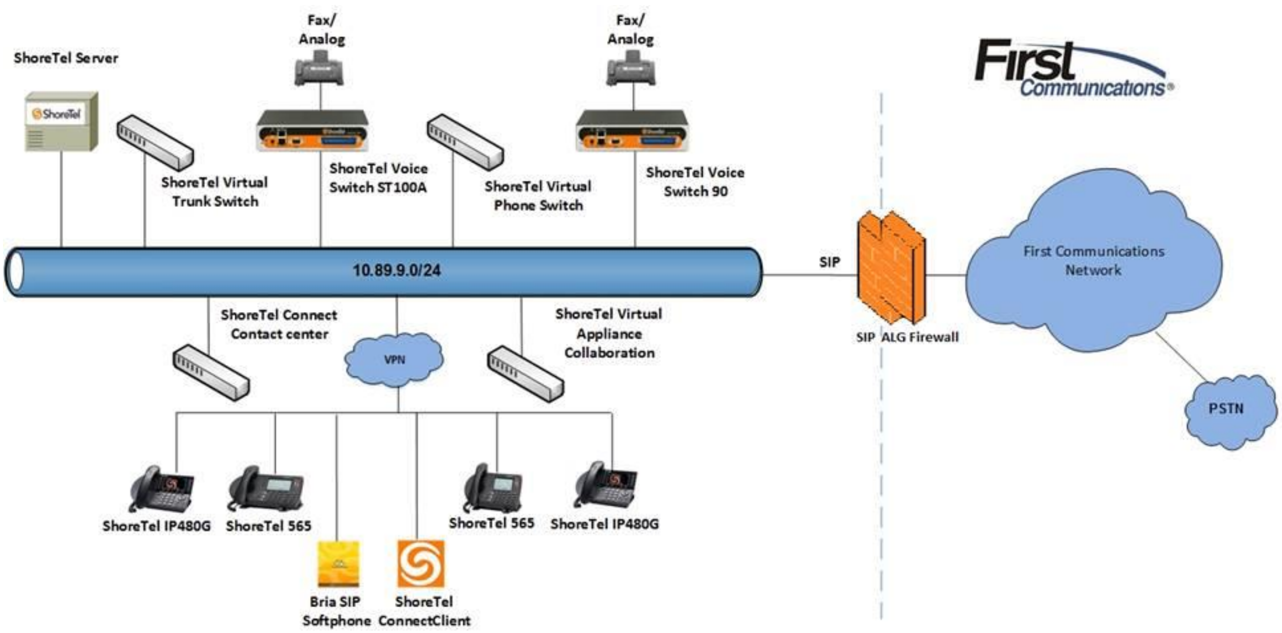

Figure 1: SIP Trunk Lab Reference Network

Test Environment

- ShoreTel Connect ONSITE HQ Server
- ShoreTel Voice Switch
- ShoreTel Virtual Trunk Switch
- ShoreTel Connect Client
- Analog Fax Machine
- ShoreTel 565 IP Phones
- ShoreTel 480G IP Phones
- ShoreTel Collaboration Service Appliance
- ShoreTel Connect Contact Center

Special Notes

The following are the caveats and limitations of First Communications SIP Trunking with ShoreTel Connect Onsite system. At this time, we are unable to provide additional information on a resolution of these limitations, but suggest to periodically refer to the ShoreTel Connect Software Build Notice for updates.

http://www.support.shoretel.com

SIP Registration

This test used a Static IP Authentication method between the ShoreTel Connect Onsite PBX and First Communications SIP Trunks. SIP Registration is not required for First Communications SIP Trunks.

Fax Support

Fax support is only limited to G711 Passthrough with ShoreTel Voice switches and Virtual Trunk switches. The support for T38 will be added in a later release.

Switch Support

The following switch types are supported with First Communications SIP Trunking

- Virtual Trunk Switch
- ShoreTel Voice Switch

Note: Although ST14.2 was not directly tested with First Communications SIP Trunks, ST14.2 is supported and retrogressively compatible with the Virtual Trunk Switches as per the recommended config in this Application Note. Additionally, 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

SIP Trunk Media Proxy

“SIP Media Proxy” is required to provide the feature parity of PRI Trunks with SIP Trunks. This includes the features like Office Anywhere, Simultaneous Ringing, 3-way Mesh Conferencing, Call Recording, Silent monitoring, Barge-In, Whisper Page etc. “SIP Media Proxy” is enabled by default on ShoreTel Virtual Trunk switches, but needs to be assigned manually on the new ShoreTel Voice Switches as well as for legacy half-width ShoreGear Switches. For further information on the “SIP Media Proxy”, please refer to Chapter 19 of the ShoreTel Connect Onsite System Administration Guide.

Unsupported Features and Limitations

The following section contains some of the features and limitations with ShoreTel SIP Trunks

- The maximum number of music on hold (MOH) streams that a SIP-enabled switch can support varies with the switch model and the switch's configuration. Also, the allotment of resources for jack-based MOH includes streams for Backup Auto Attendant and transmission of ringback tones. The range of such streams across all the voice switch models is 14–60.
- 4 to 6 party conferences, when a SIP trunk is involved, utilize Make Me conference ports
- Silent Monitor, Barge-In, Silent Coach and Call recording features are supported by SIP Trunk only if the trunk has a SIP trunk profile with hairpinning and the trunk is on a half-width switch or a virtual switch
- The ShoreTel system does not initiate calls with a 30ms payload; all calls are initiated with a 20ms payload
- For inbound calls, ShoreTel does remote codec honoring and will negotiate media per Incoming Request preference; for outbound call, First Communications will negotiate the codec based on the preference of its configuration.

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

Configuration

Configuration Steps

This section contains an overview of the steps required to configure a ShoreTel Connect ONSITE IP-PBX with First Communications SIP Trunks.

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 per the topology deployed.

Table 2 – IP Address Worksheet

Component	MSO Lab Value	Customer Value
ShoreTel Connect ONSITE IP-PBX		
ShoreTel Server	10.89.9.2	Unique to every deployment
ShoreTel Voice Switch ST100DA	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
ShoreTel Virtual Collaboration SA	10.89.9.8	Unique to every deployment
ShoreTel Connect Contact Center	10.64.4.48	Unique to every deployment

Create Custom Codec Lists and Sites

Create Codec Lists

1. Navigate to **Administration > Features > Call Control > Codec Lists**
2. Click **NEW**
3. Set **Description**: First Com was used for this example
4. **Codec List Members**: PCMU/8000, PCMA/8000, and G729/8000 were moved from the **Available** column to the **Selected** column for this test
5. Click **SAVE**

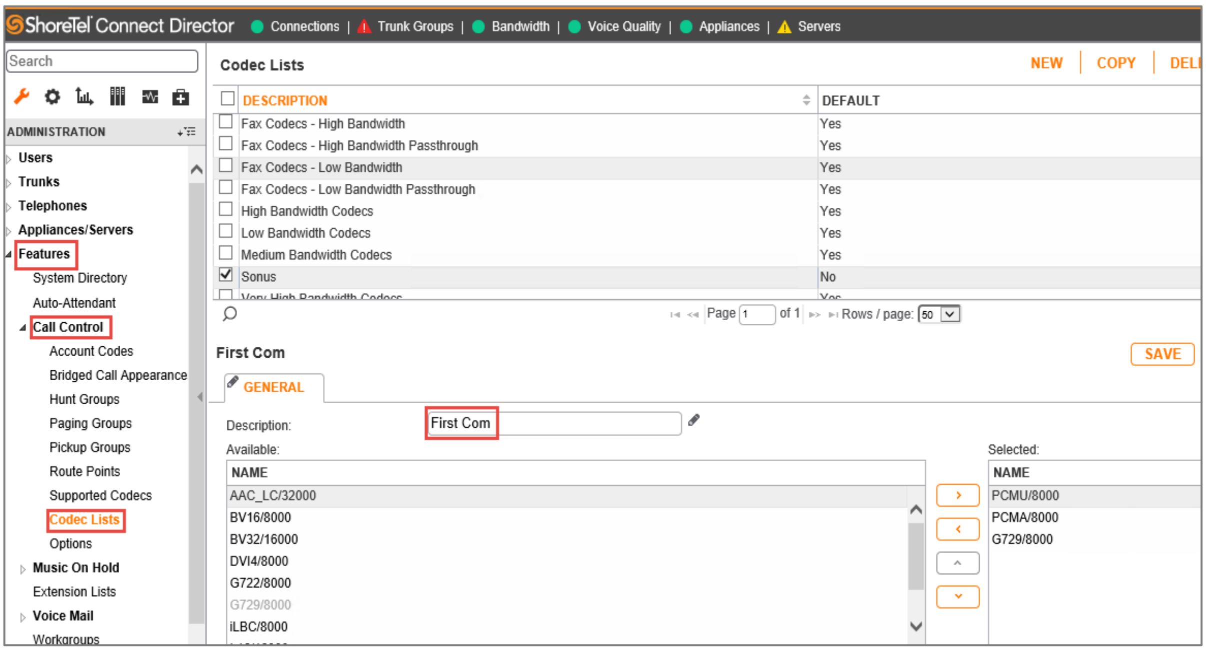

Figure 2: Codec Lists

Create Sites

1. Navigate to **Administration** > **System** > **Sites**
2. Set **Name**: Headquarters is used as an example
3. Set **Local Area Code**: 224 is used in this test
4. Set **Admission control bandwidth:** 1024 kbps is used in this test
5. Set **Intra-Site Calls**: Codec List *FirstCom* was selected from the drop-down menu as an example
6. Set **Inter-Site Calls**: Codec List *FirstCom* is selected from the drop-down menu
7. Set **FAX and Modem Calls**: Default Codec List *Fax Codecs – Low Bandwidth Passthrough* is selected from the drop-down menu
8. Set **PROXY Switch 1**: Select Lab109-VPs1
9. Leave all other fields as default
10. Click **SAVE**

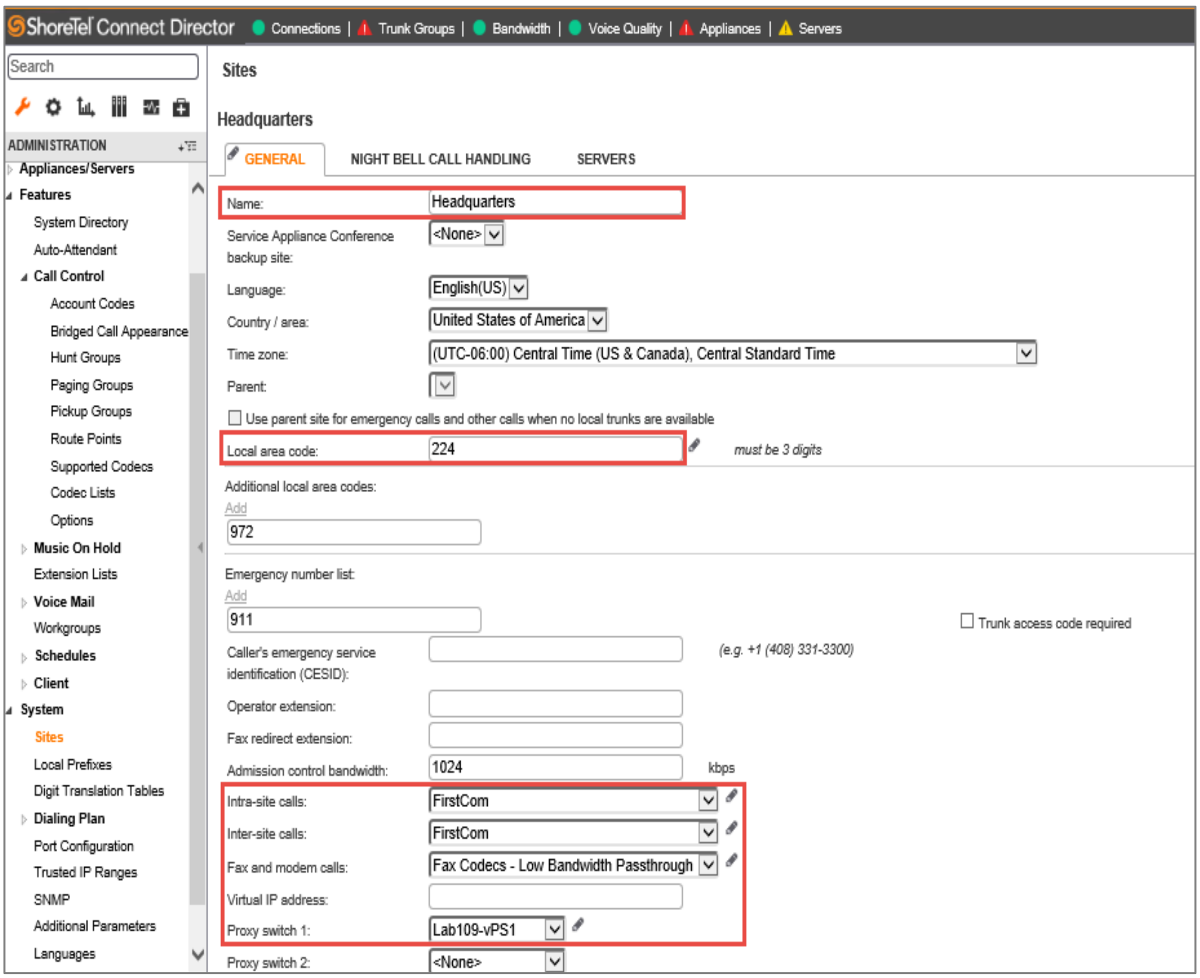

Figure 3: Sites

SIP Trunk Configuration

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

SIP Trunk Profile

1. Navigate to **Administration > Trunks > SIP Profiles**
2. The **Default ITSP** SIP Profile was selected for this test

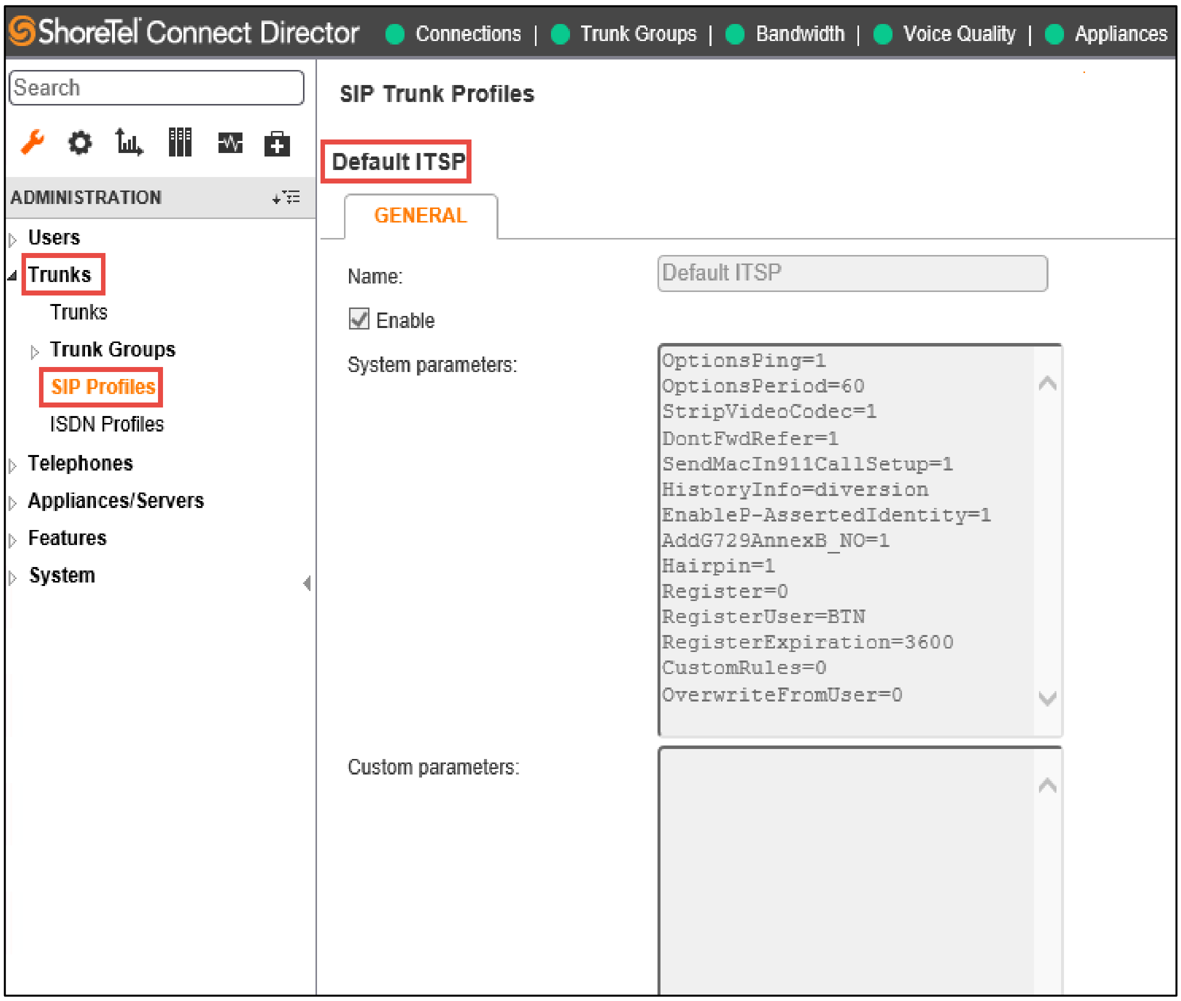

Figure 4: SIP Profile

Add Trunk Group

1. Navigate to **Administration > Trunks > Trunk Groups > Trunk Groups**
2. Select the **GENERAL** tab
3. Set **Name**: *FirstCom*
4. Set **Trunk Type**: *SIP* is selected from the drop-down menu
5. Set **Profile**: SIP Profile *Default ITSP* is selected from drop-down menu
6. Set **Digest Authentication**: *None* is selected
7. Click **SAVE**

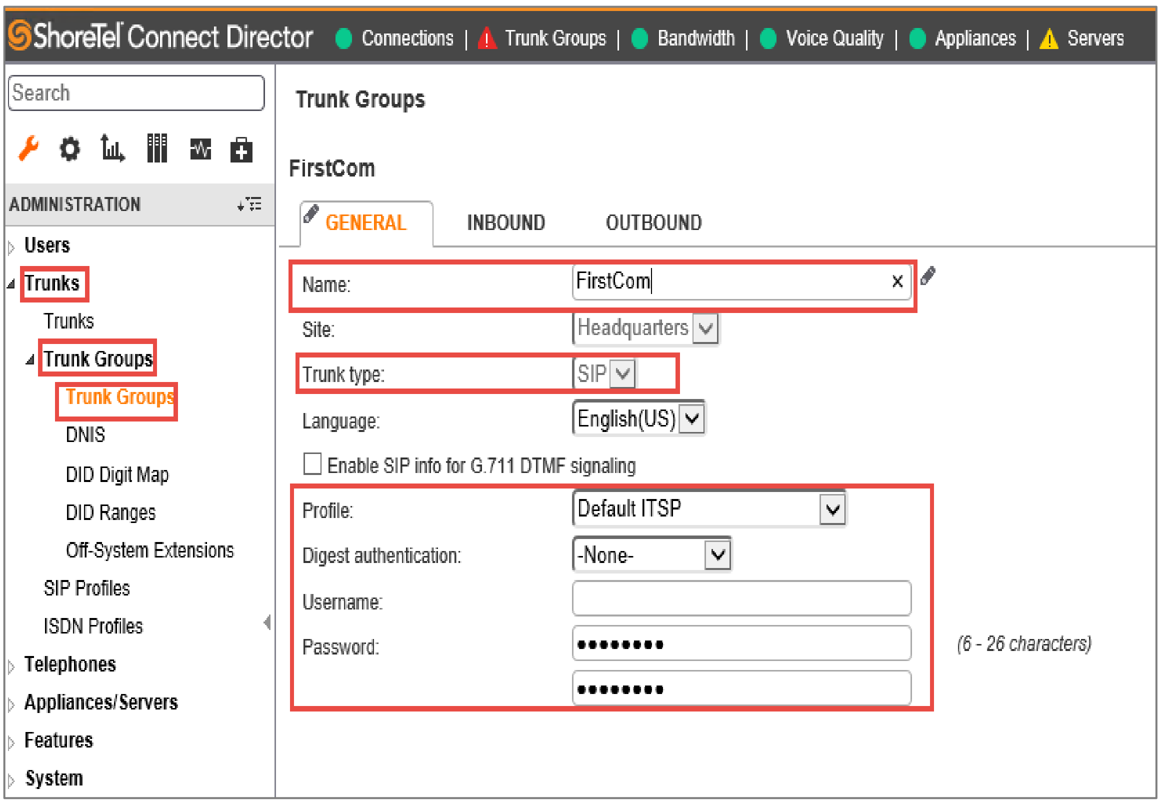

Figure 5: Trunk Groups

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

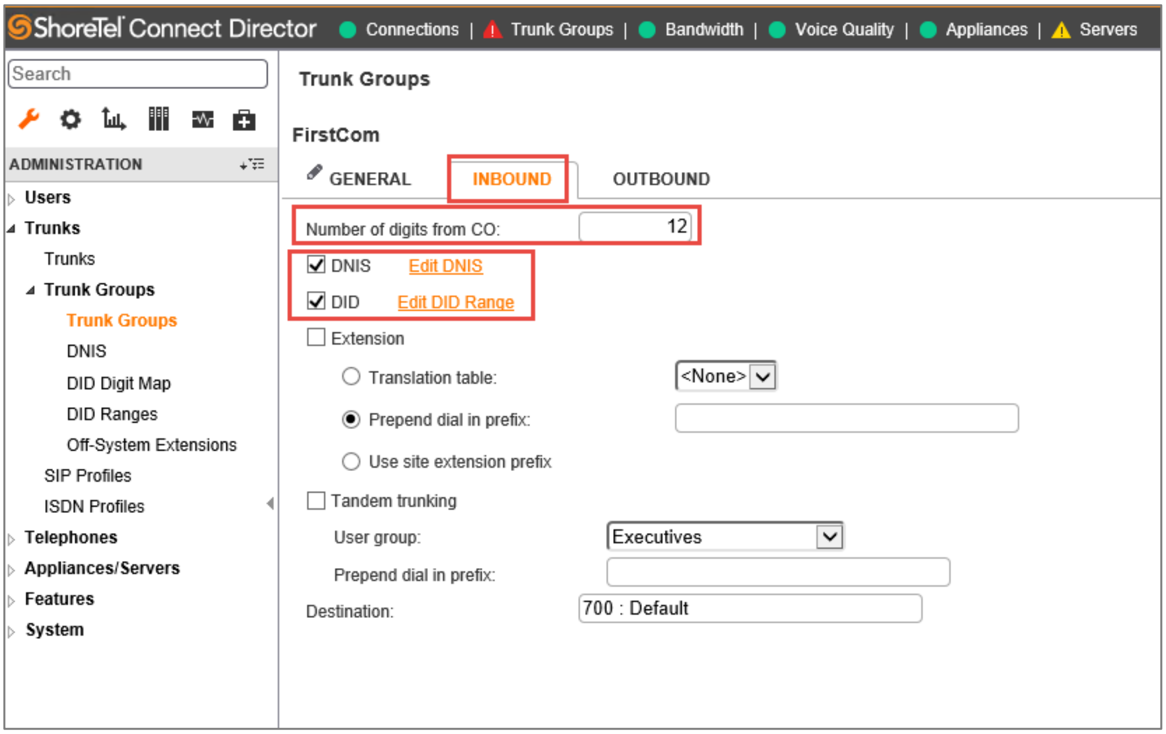

Figure 6: Trunk Groups – Cont.

13. Select the **OUTBOUND** tab
14. **Outgoing**: Checked
15. Set **Access Code**: 9 is used in this example
16. Set **Local Area Code**: 224 is used in this example
17. Set **Billing Telephone Number**: The Pilot number will be provided by your First Communications Account Representative and must be kept confidential
18. Leave all other fields as default
19. Click **SAVE**

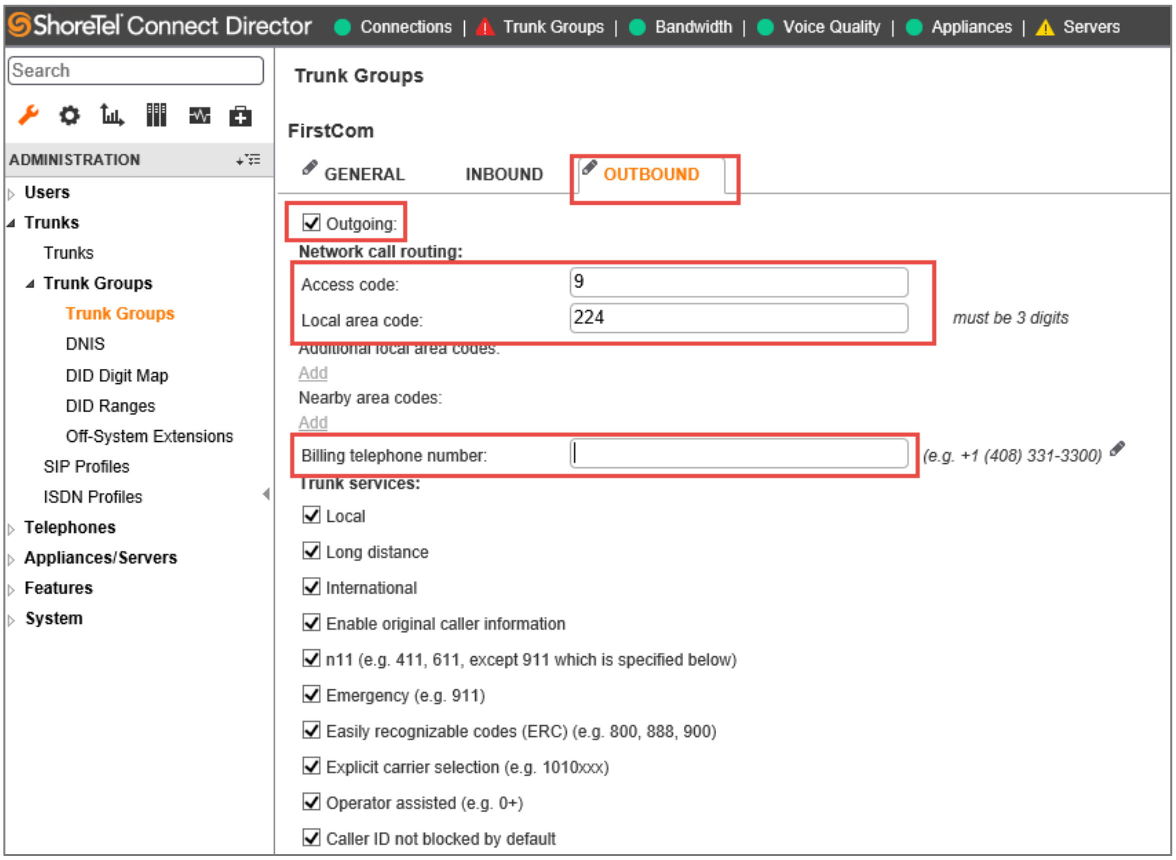

Figure 7: Trunk Groups – Cont.

Trusted IP Ranges

In order to transmit the SIP signaling and RTP packets properly, the service provider Signaling and Media IP address needs to be added into Trusted IP Ranges

1. Navigate to **System** > **Trusted IP Ranges**
2. Click **NEW**
3. Set **Name**: **FirstCom** is used for this setup
4. Set **Low IP Address**: Enter the service provider lowest Signaling/Media IP address
5. Set **High IP Address**: Enter the service provider highest Signal/Media IP address
6. Click **SAVE**

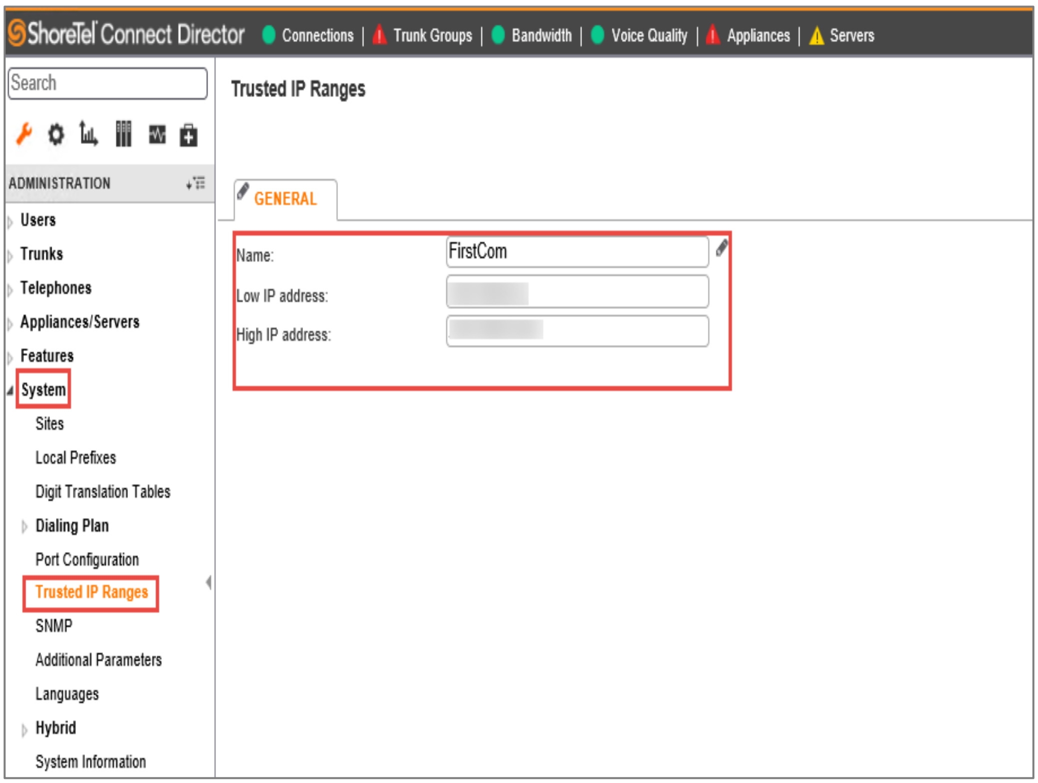

Figure 8: Trusted IP Ranges

Create Individual Trunks

1. Navigate to **Administration > Trunks > Trunks**
2. Set **Trunk Group**: FirstCom *(SIP)* is selected from the drop-down menu
3. Set **Name**: *FirstCom* is used in this setup
4. Set **Switch**: *Lab109-vTS1* is selected from the drop-down menu
5. Set **IP Address or FQDN**: Enter the IP Address of the First Communications SIP Trunks. Please contact your First Communications sales representative for additional information.
6. Click **SAVE**

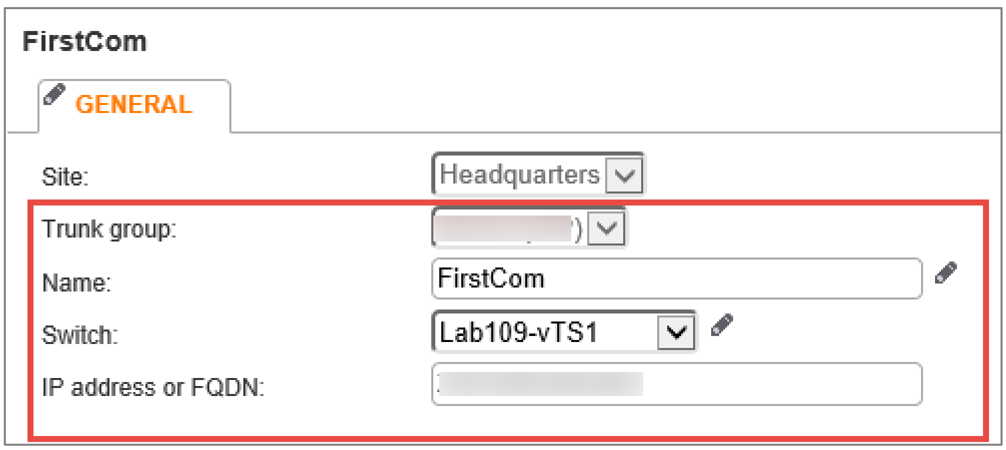

Figure 9: Individual Trunks

Summary of Tests and Results

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

Primary Switch Test Plan (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 Caller	Verify an inbound caller hears busy tone when all channels/Trunks are in use	
1.5	PASS	All Trunks Busy–Outbound Caller	Verify an outbound caller 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	PASS	Codec Negotiation	Verify codec negotiation between the SUT and the calling device with each side configured for a different codec	
2.2	PASS	DTMF Transmission – Out of Band / Inband	Verify transmission of inband and out-of-band digits per RFC 2833 for various devices connected to the SUT	
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	PASS	Operator Assisted	Verify that 0+ calls are routed to an operator for calling assistance	
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	

ID	Result	Name	Description	Notes
4.11	PASS	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	
4.12	PASS	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	
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	PASS	Inbound / Outbound Fax Calls	Verify that inbound / outbound fax calls complete successfully	Only G.711 Passthrough fax is supported
4.16	PASS	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.	
4.17	PASS	Inbound call to Bridged Call Appearance (BCA) 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	

ID	Result	Name	Description	Notes
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	Make Me Conference	Verify that an inbound call can be conferenced with three (or more) additional parties	
4.22	PASS	Park / Unpark	Verify that an inbound call can be parked and unparked	
4.23	PASS	Call Recording	Verify that external calls can be recorded via the SIP Trunk using ShoreTel Communicator	
4.24	PASS	Silent Monitor / Barge-In / Whisper Page	Verify that external calls can be silently monitored, barged-in and whisper paged via the SUT	
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	
4.27	PASS	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.	

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	

Secondary Switch Sanity Test Results (ST 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	PASS	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	
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	PASS	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	

4.15	PASS	Inbound / Outbound Fax Calls	Verify that inbound / outbound fax calls complete successfully	Only G.711 Passthrough fax is supported
4.16	PASS	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	Make Me Conference	Verify that an inbound call can be conferenced with three (or more) additional parties	
4.23	PASS	Call Recording	Verify that external calls can be recorded via the SIP Trunk using ShoreTel Communicator	
4.24	PASS	Silent Monitor / Barge-In / Whisper Page	Verify that external calls can be silently monitored, barged-in and whisper paged via the SUT	
4.27	PASS	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

First Communications SIP Trunking has been successfully tested with ShoreTel Connect ONSITE.

Additional Resources

ShoreTel Connect ONSITE System Administration Guide

ShoreTel Connect ONSITE Planning and Installation Guide

ShoreTel Connect ONSITE Client User Guide

Version	Date	Contributor	Content
1.0	July 19, 2017	Pradeep Nagubandi	Original Release
1.1	August 18, 2017	Pradeep Nagubandi	ShoreTel requested revisions

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