SHORETEL APPLICATION NOTE

for

BT Wholesale SIP Trunking (WSIPT) &
BT Global Services One Voice SIP Trunk UK

Date:	July 6, 2016
App Note Number:	TC - 16040
For use with:	BT Wholesale SIP Trunking (WSIPT) &
	BT Global Services One Voice SIP Trunk UK
Product:	ShoreTel Connect ONSITE
System:	ShoreTel Connect ONSITE Build [21.76.3106.0]

This document and its contents was prepared by:

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 Application Note describes configuration steps for configuring the BT SIP Trunking with ShoreTel Connect ONSITE System.

British Telecom

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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.

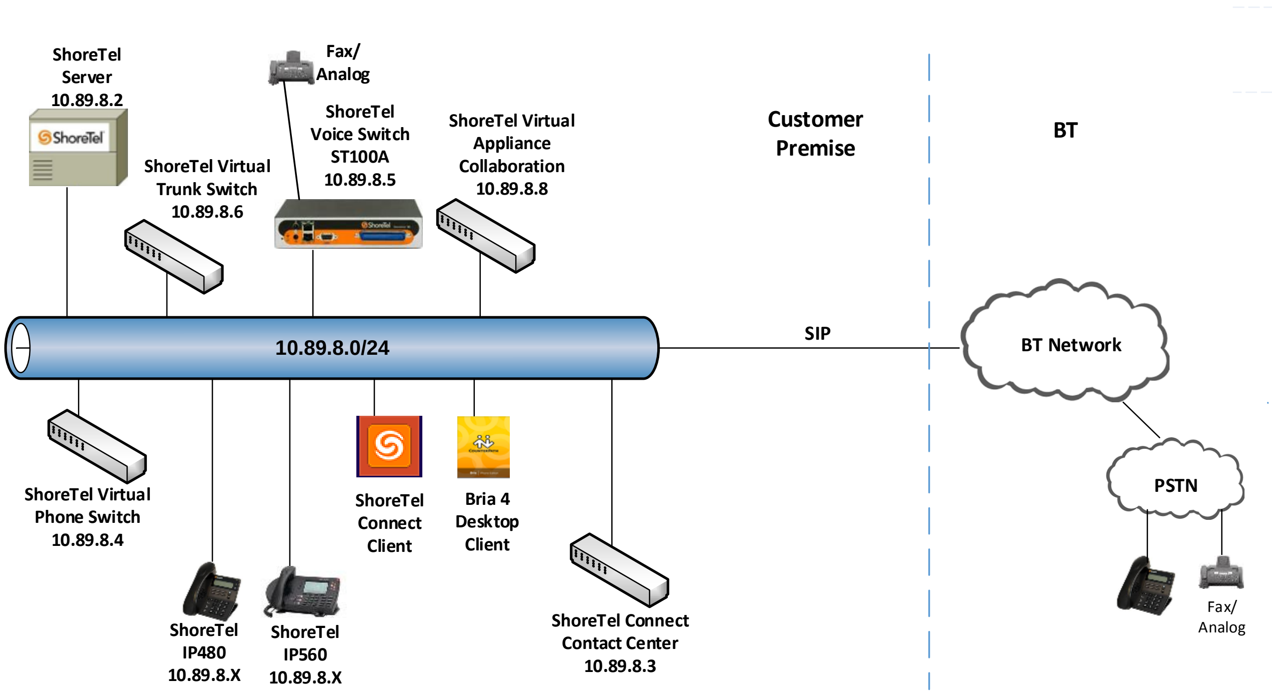

Figure 1: SIP Trunk Lab Reference Network

Hardware Components

- VMware ESXi 5.5
- ShoreTel Voice Switch (ST100A)
- Analog Fax Machine
- ShoreTel 560 IP Phones
- ShoreTel 230 IP Phones
- ShoreTel 480 IP Phones
- ShoreTel 115 IP Phones
- ShoreTel Collaboration Service Appliance
- ShoreTel Connect Contact Center

Software Requirements

- ShoreTel Connect ONSITE, Build 21.76.3106.0
- ShoreTel Connect Client, Build 212.4000.3169.0
- ShoreTel Collaboration Service Appliance
- ShoreTel Connect Contact Center, Build 507.80.6109
- Windows Server 2012 R2 Standard

Features

SIP Registration Method

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

Features Supported

- Basic calls with G711Alaw
- Call Hold and Resume
- Call Transfer
- Call Forwarding
- DTMF RFC 2833
- Calling Party Number Presentation
- Calling Party Number Restricted
- Hunt Group
- Group Pickup
- Call Park/un-park
- Simul Ring
- Call Forward – “FindMe”
- Call Recording
- Auto-Attendant
- Meetme Conference
- Reservation-less conference
- Contact Center
- Work Group
- Office Anywhere External
- Silent Monitor / Barge-In / Whisper Page
- Voicemail Message Deposit

Features Not Supported

- Operator Assistance Calls
- G711 Pass-Through
- t.38 fax

Caveats and Limitations

- Fax is not supported with BT SIP Trunks
- **NOTE**: There may be other feature limitations when using SIP Trunks. Please refer to Chapter 19 of the **ShoreTel Connect ONSITE Administration Guide** for more information**.**

Configuration

Configuration Steps

In this section an overview is presented of the steps that are required to configure the ShoreTel Connect ONSITE IP-PBX to connect to the BT site via a SIP Trunk.

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	10.89.8.x	Unique to every deployment
ShoreTel Voice Switch ST100A	10.89.8.x	Unique to every deployment
ShoreTel Virtual Trunk Switch	10.89.8.x	Unique to every deployment
ShoreTel Virtual Phone Switch	10.89.8.x	Unique to every deployment
ShoreTel Virtual Collaboration SA	10.89.8.x	Unique to every deployment
ShoreTel Connect Contact Center	10.89.8.x	Unique to every deployment
BT SBC		
WAN IP Address	192.65.x.x	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 **Name**: BT was used for this example
4. **Codec List Members**: PCMA/8000, PCMU/8000 were added for this test
5. Click **SAVE**

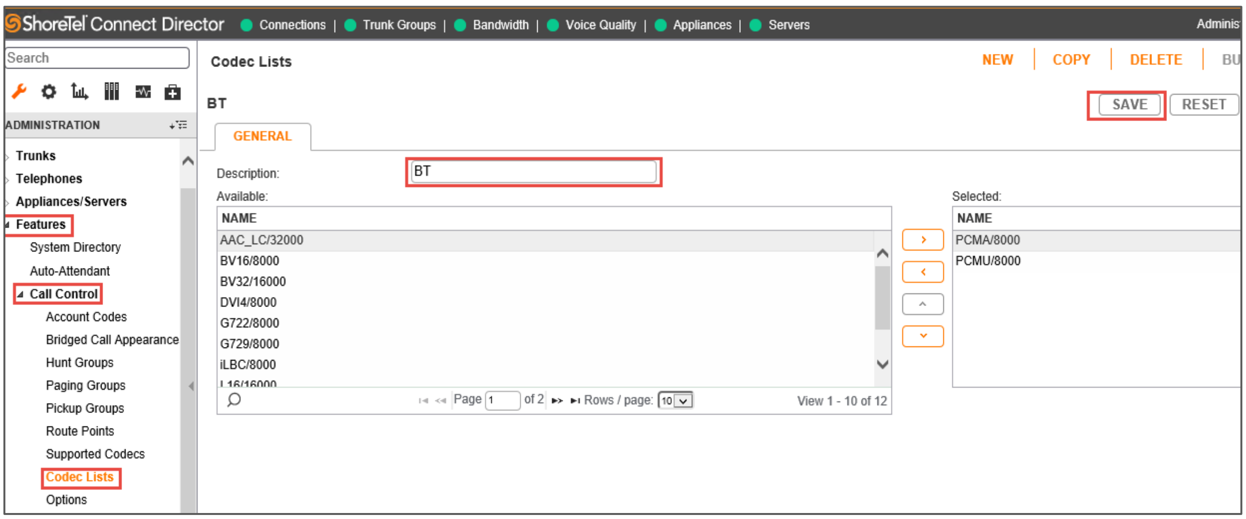

Figure 2: Codec Lists

Create Sites

1. Navigate to **System** > **Sites**
2. Set **Name**: BT
3. Set **Local Area Code**: 122 is used in this test
4. Set **Intra-Site Calls**: Codec List *BT* was selected from the drop down menu as an example. This selection contains only G711 codecs.
5. Set **Inter-Site Calls**: Codec List *BT* is selected from the drop down menu
6. Set **FAX and Modem Calls**: Default Codec List *Fax Codecs – Low Bandwidth Passthrough* is selected from the drop down menu
7. Set PROXY Switch 1: Select Lab108-vps1
8. Leave all other fields as default
9. Click **SAVE**

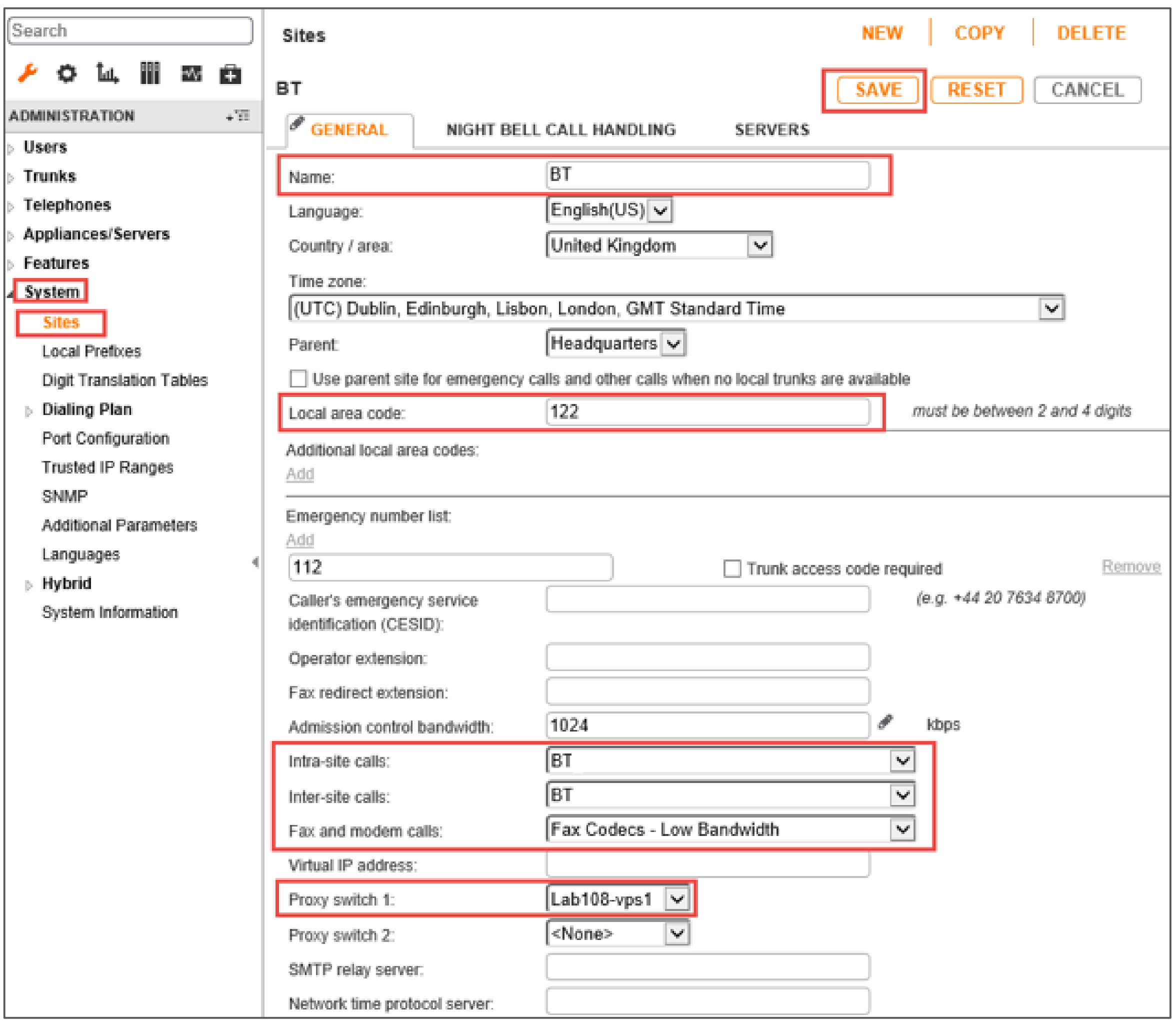

Figure 3: Sites

SIP Trunk Configuration

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

SIP Trunk Profile

The **Default ITSP** SIP Profile was selected for this test.

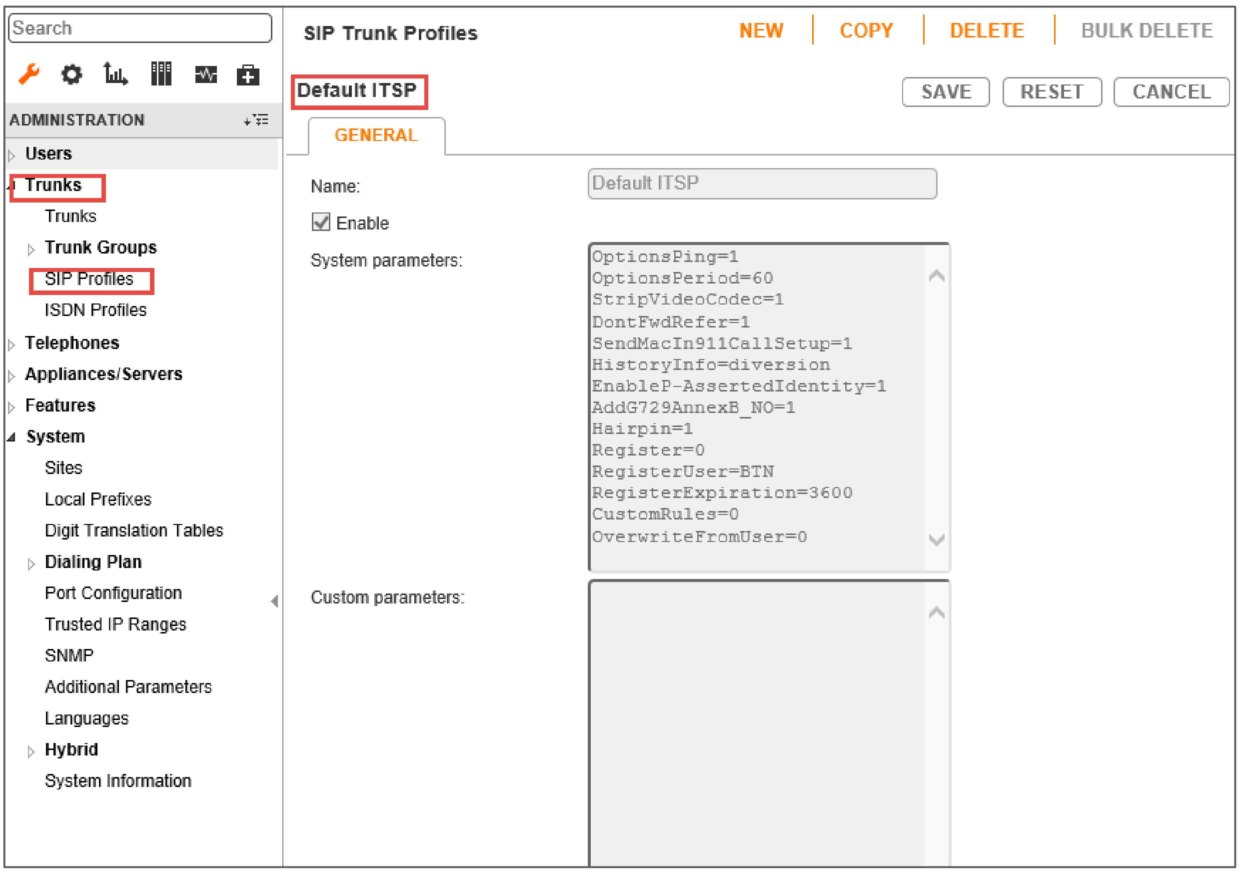

Figure 4: SIP Profile

Add Trunk Group

1. Navigate to **Trunks > Trunk Groups > Trunk Groups**
2. Select the **GENERAL** tab
3. Set **Name**: *BT*
4. Set **Trunk Type**: *SIP*
5. Set **Profile**: SIP Profile *Default ITSP* is selected from drop down
6. Set **Digest Authentication**: *None* is selected
7. Click **SAVE** (Not Shown Here)

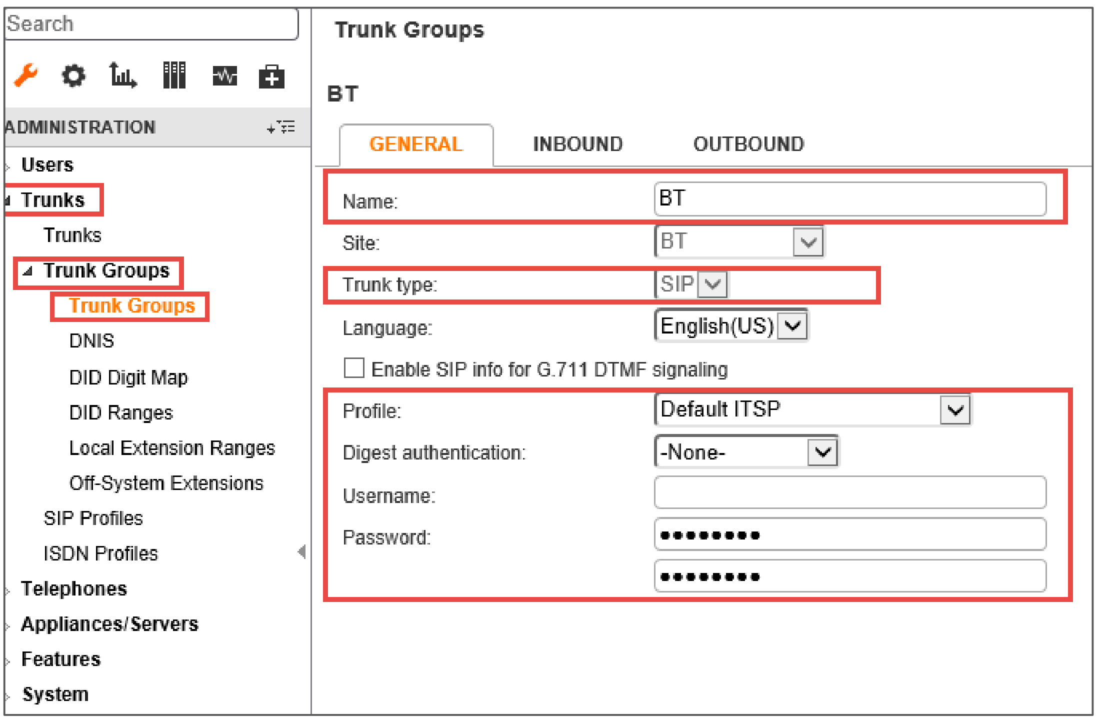

Figure 5: Trunk Groups

8. Select the **INBOUND** tab
9. Set **Number of Digits from CO**: 13 is used in this setup
10. **DNIS**: Checked
11. **DID**: Checked
12. Click **SAVE** (Not Shown Here)

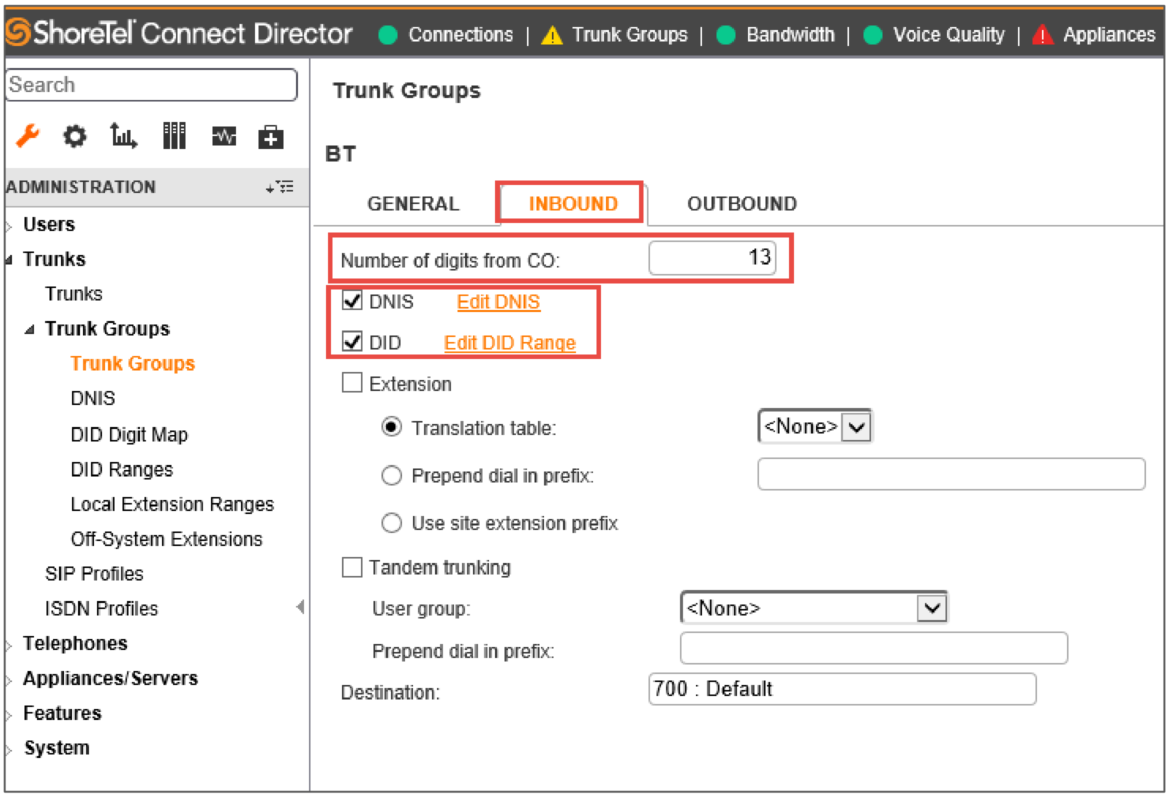

Figure 6: Trunk Groups - Cont.

13. Select the **OUTBOUND** tab
14. **Outgoing**: Checked
15. Set **Access Code**: 9 is used in this example
16. Set **Local Area Code**: 122 is used in this example
17. Set **Billing Telephone Number**: Pilot number will be provided by your BT Business Account Representative and must be kept confidential
18. Leave all other fields as default
19. Click **SAVE** (Not Shown Here)

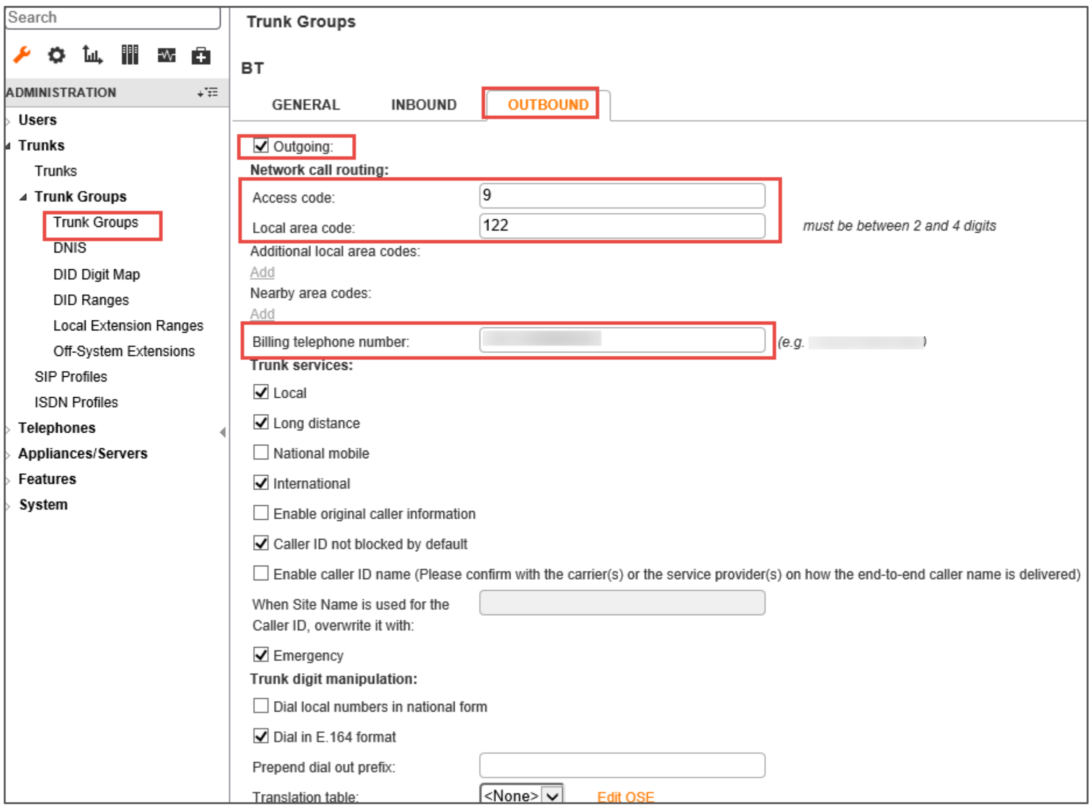

Figure 7: Trunk Groups - Cont.

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**: **BT** is given 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**

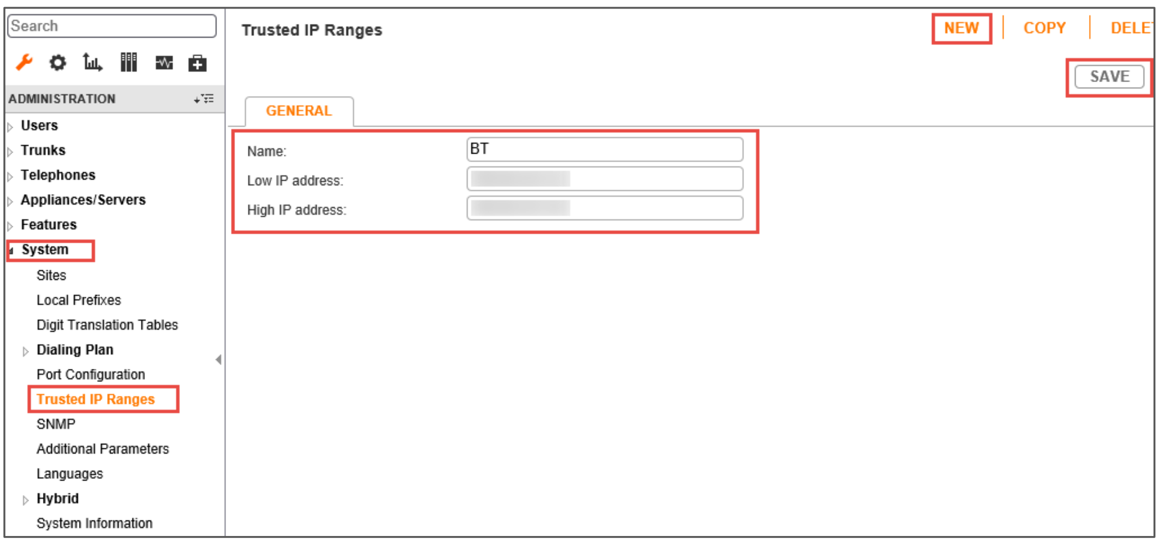

Figure 8: Trusted IP Ranges

Create Individual Trunks

1. Navigate to **Trunks > Trunks**
2. Set **Trunk Group**: BT *(SIP)* is selected from the drop down menu
3. Set **Name**: *siptrunk* is used in this setup
4. Set **Switch**: *Lab108-Vts* is selected from the drop down menu
5. Set **IP Address or FQDN**: Enter the LAN IP Address of the remote BT SBC. Please contact BT for more information regarding your deployment.
6. Click **SAVE** (Not Shown Here)

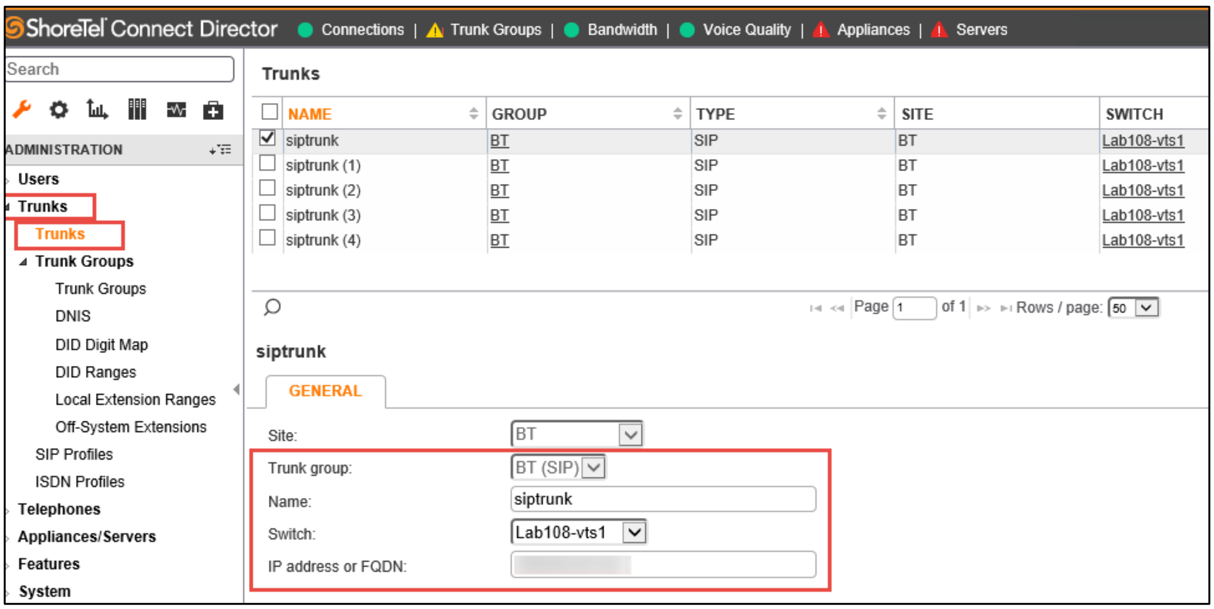

Figure 9: Individual Trunks

BT SIP Trunk Configuration

When ordering the service there will be multiple options available to tailor the trunks deployment, the following choices need to be made to ensure correct interworking with the ShoreTel Connect ONSITE (Build 21.76.3106.0)

Option	Value
Transport Protocol	UDP
Inbound Calling Party Format	Global
Inbound Called Party Format	Global
Network CLI	PAID
Presentation CLI	From

Summary of Tests and Results

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

Primary Switch Test Plan

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	C-PASS	Incomplete Inbound Calls	Verify proper call progress tones are provided and proper call teardown for incomplete inbound calls	Incoming call to PBX- no answer: An unanswered call is routed to the auto-attendant, but the call to the called party is never canceled and keeps ringing

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	UK emergency number is 999
4.9	N/T	Operator Assisted	Verify that 0+ calls are routed to an operator for calling assistance	Operator Assistance service is not available on BT lab trunk
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	N/S	Inbound / Outbound Fax Calls	Verify that inbound / outbound fax calls complete successfully	Fax interworking is not supported with BT SIP Trunks
4.17	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	
4.18	PASS	Office Anywhere External	Verify that inbound calls are properly presented to the Office Anywhere External PSTN destination	
4.19	PASS	Simul Ring	Verify that inbound calls are properly presented to the desired extension and the “Additional Phones” destinations	
4.20	PASS	MakeMe Conference	Verify that an inbound call can be conferenced with three (or more) additional parties	
4.21	PASS	Park / Unpark	Verify that an inbound call can be parked and unparked	

ID	Result	Name	Description	Notes
4.22	PASS	Call Recording	Verify that external calls can be recorded via the SIP Trunk using ShoreTel Communicator	
4.23	PASS	Silent Monitor / Barge-In / Whisper Page	Verify that external calls can be silently monitored, barged-in and whisper paged via the SUT	
4.24	PASS	Long Duration – Inbound	Verify that an inbound call is established for a minimum of 30 minutes	
4.25	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	BT does not require Digest Authentication and SIP trunk registration for this lab test setup.

Secondary Switch Sanity Test Results

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	N/S	Inbound / Outbound Fax Calls	Verify that inbound / outbound fax calls complete successfully	Fax interworking is not supported with BT SIP Trunks
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	MakeMe 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	

BT Test Case Results

ID	Result	Name	Description	Notes
BT1	PASS	Inbound Calls using ptime= 10ms, 30ms	Verify that ShoreTel can establish call with incoming BT call using ptime=10ms or ptime=30ms Verify call establishes and speech path between call parties	Verified Shoretel was able to establish call with ptime=10ms and 30ms
BT2	PASS	ShoreTel PBX can failover to secondary trunk when primary trunk is down	Verify that ShoreTel can failover to a secondary trunk when the primary trunk path goes down.	2 SBCs and trunks will be configured on the BT side

Conclusion

BT SIP Trunking has been successfully tested with ShoreTel Connect ONSITE Build 21.76.3106.0.

Additional Resources

ShoreTel Administration Guide

Version	Date	Contributor	Content
1.0	April 19, 2016	Richard Moreno	Original Release
1.1	April 21, 2016	Richard Moreno	Revisions after internal review
1.2	May 9, 2016	Rachel Jerome	Added G711 pass-through as N/S features
1.3	May 24, 2016	Richard Moreno	ShoreTel requested revisions

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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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