SHORETEL APPLICATION NOTE

for

DIDforSale SIP Trunking

Date:	March 1, 2016
App Note Number:	TC-16019
For use with:	DIDforSale SIP Trunking
Product:	ShoreTel Connect ONSITE
System:	ST Connect 21.75.4111.0

Contents

Introduction

This Configuration Guide describes configuration steps for DIDforSale SIP Trunking to ShoreTel Connect Onsite System.

DIDforSale

DIDforSale is a subsidiary of Cebod Technologies LLC that has been in VOIP business since 2007. DIDforSale's SIP Trunking solution, offers business class telephone service that delivers local phone numbers, Tollfree and long distance calling services.

With over 13,000 rate centers, we offer the largest coverage in USA, UK and Canada. Our network is optimized to minimize call drops, lags, and echoes allowing us to handle your calls with amazing call clarity. We take pride in providing quality service to our customers

With our largest single tier SIP Trunking coverage, businesses can have virtual phone anywhere, and share pool of SIP trunks between multiple locations, resulting in reduction in the monthly billing.

For more information please visit www.didforsale.com

For Sales Inquiry:

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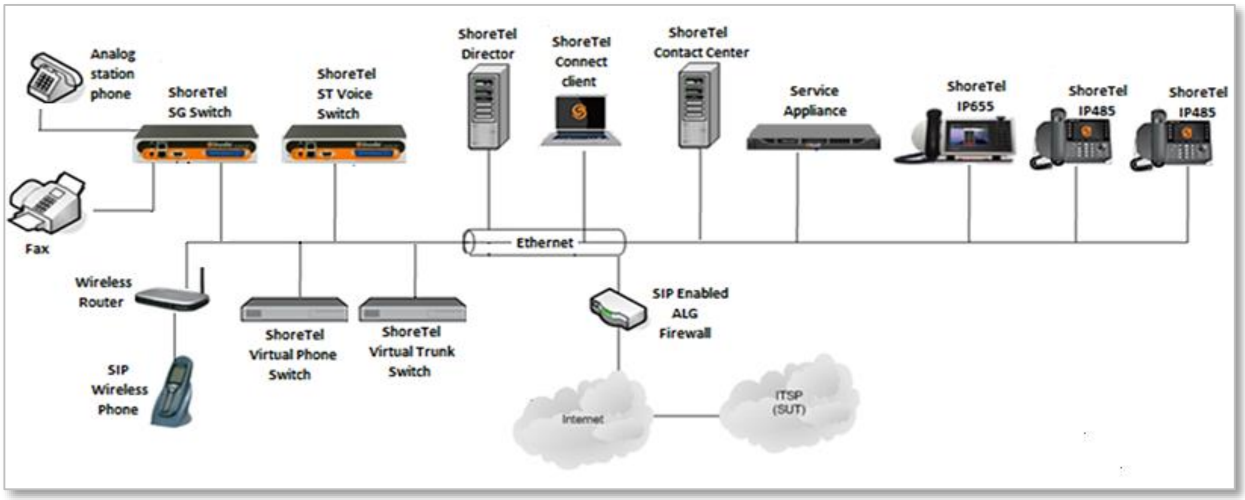
Email: contact-sales@didforsale.com
Website: http://www.didforsale.com/contact/sales

Technical Support

Phone: 800-579-7676 x 2
Email: contact-support@didforsale.com
Website: http://www.didforsale.com/contact/support

Network Topology



Test Environment

- ShoreTel Connect Onsite Server
- ShoreTel Voice Switch
- ShoreGear Switch
- Analog Fax Machine
- ShoreTel 655 IP Phones
- ShoreTel 485G IP Phone (SIP Ref)
- ShoreTel Virtual Phone Switch
- ShoreTel Virtual Trunk Switch
- Wireless SIP Phone
- ST Connect Contact Center
- ShoreTel Collaboration Service Appliance
- ShoreTel Connect Client

Special Notes

The following are the caveats and limitations of DIDforSale 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, which can be found at the following location:
http://www.support.shoretel.com

SIP Registration

SIP Registration is required between the ShoreTel Connect Onsite PBX and DIDforSale SIP Trunks.

Fax Support

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

Switch Support

The following switch types are supported with DIDforSale SIP Trunking

- Virtual Trunk Switch
- ShoreTel Voice Switch

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.

ShoreTel Unsupported Features and Limitations

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

- Fax redirect not supported via SIP Trunks using G.711 (though Direct Inward Dialing (DID) to fax endpoint is supported)
- ShoreTel supports Music On Hold (MOH) over SIP trunks. The maximum number of music on hold (MOH) streams that a SIP-enabled switch can support varies with the switch model. The range of such streams across all the voice switch models is 14–60. Limitation: MOH source needs be on SIP trunk switch.
- ShoreTel supports the Service Appliance (SA-100) conferencing / IM system from Release 12. SIP trunk calls from / to the SA-100 is supported. The SA-100 accepts access codes in DTMF RFC2833 only.
- 4 to 6 party conferences, when a SIP trunk is involved, utilize Make Me conference ports.
- Silent Monitoring, Barge-In, Silent Coach, Park/Unpark , Call recording features are supported on a SIP trunk call only if SIP trunk is configured with SIP profile supporting media hairpinning and the trunk is on a half-width switch.
- Silence detection on trunk-to-trunk transfers is not supported, it requires a physical trunk.
- The ShoreTel system does not initiate calls with a 30ms payload; all calls are initiated with a 20ms payload.

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

This section describes the detailed steps for ShoreTel system configuration to support DIDforSale SIP Trunking.

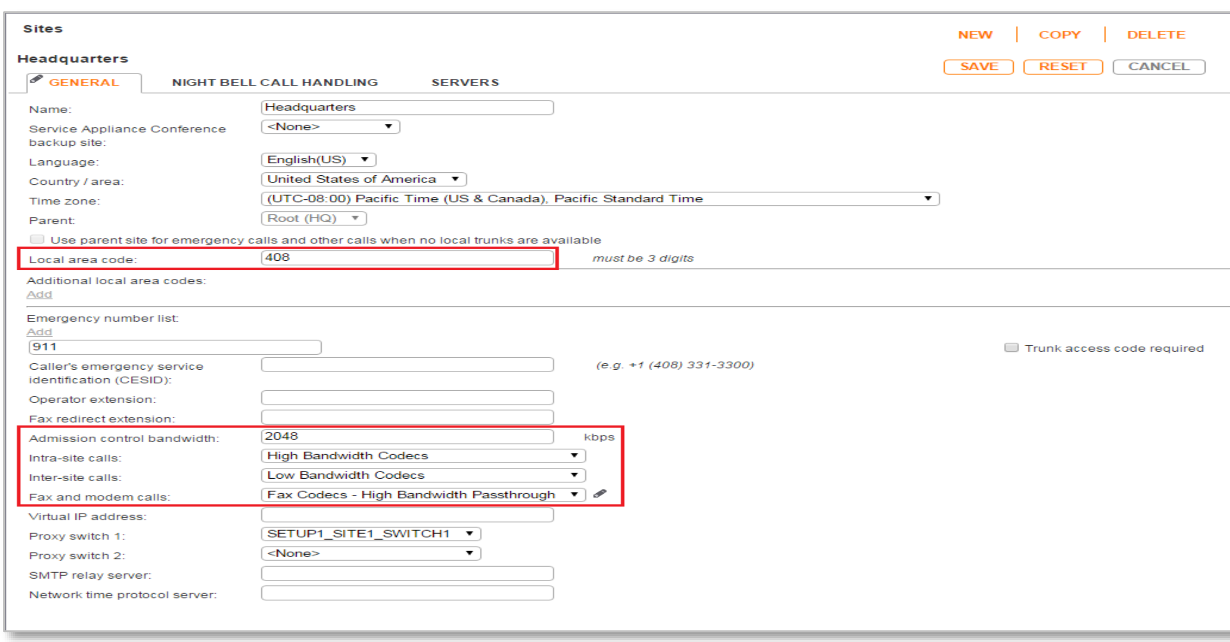
Codec Configuration

This section describes the codec configuration required on the ShoreTel system to work with DIDforSale SIP Trunking. DIDforSale SIP Trunking uses the default Codec list “High Bandwidth Codec” for Intra-Site and “Low Bandwidth Codec” for Inter-Site calls. No modifications are required in this section.

Site Configuration

This section describes the Site configuration required on the ShoreTel system to work with DIDforSale SIP Trunking.

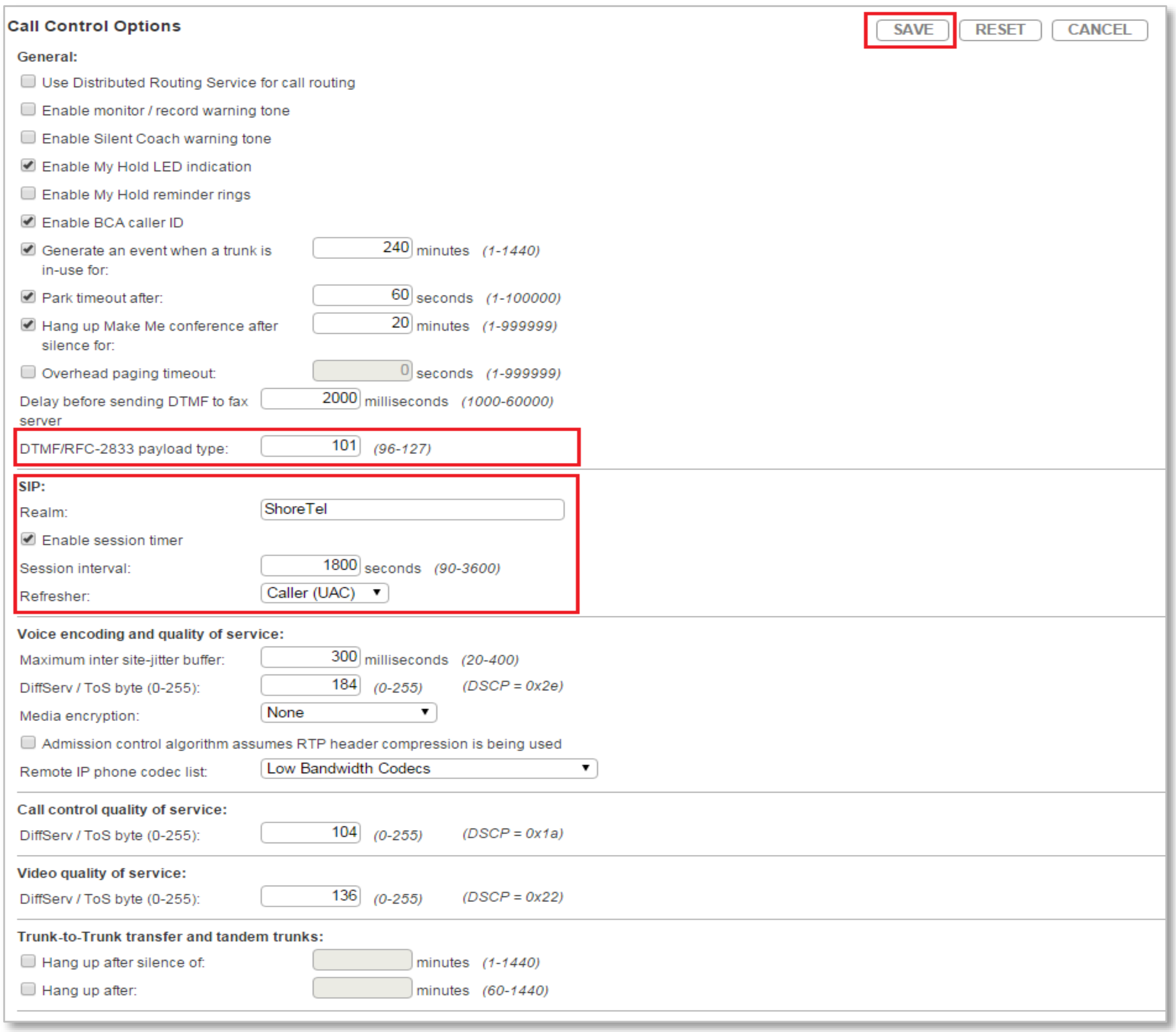
1. Navigate to System > Sites
2. Set **Name**: Headquarters
3. Set **Local Area Code**: 408 is used in this example
4. **Intra-Site Calls**: The default codec list “High Bandwidth Codecs” is selected from the drop down menu
5. **Inter-Site Calls**: The default codec list “Low Bandwidth Codecs” is selected from the drop down menu
6. **Fax and Modem Calls**: The default Codec List, Fax Codecs – High Bandwidth Passthrough, is selected from the drop down menu.
7. Leave all other fields as default
8. Click **SAVE**



Call Control Configuration

This section describes the Call Control configuration required on the ShoreTel system to work with DIDforSale SIP Trunking.

1. Navigate to Features > Call Control > Options
2. Set **DTMF/RFC-2833 payload type**: 101
3. Leave all other fields as default
4. Click **SAVE**



Call Control Options

SAVE RESET CANCEL

General:

- [ ] Use Distributed Routing Service for call routing
- [ ] Enable monitor / record warning tone
- [ ] Enable Silent Coach warning tone
- [x] Enable My Hold LED indication
- [ ] Enable My Hold reminder rings
- [x] Enable BCA caller ID
- [x] Generate an event when a trunk is in-use for: 240 minutes (1-1440)
- [x] Park timeout after: 60 seconds (1-100000)
- [x] Hang up Make Me conference after silence for: 20 minutes (1-999999)
- [ ] Overhead paging timeout: 0 seconds (1-999999)

Delay before sending DTMF to fax server: 2000 milliseconds (1000-60000)

DTMF/RFC-2833 payload type: 101 (96-127)

SIP:

Realm: ShoreTel

- [x] Enable session timer

Session interval: 1800 seconds (90-3600)

Refresher: Caller (UAC)

Voice encoding and quality of service:

Maximum inter site-jitter buffer: 300 milliseconds (20-400)

DiffServ / ToS byte (0-255): 184 (0-255) (DSCP = 0x2e)

Media encryption: None

- [ ] Admission control algorithm assumes RTP header compression is being used

Remote IP phone codec list: Low Bandwidth Codecs

Call control quality of service:

DiffServ / ToS byte (0-255): 104 (0-255) (DSCP = 0x1a)

Video quality of service:

DiffServ / ToS byte (0-255): 136 (0-255) (DSCP = 0x22)

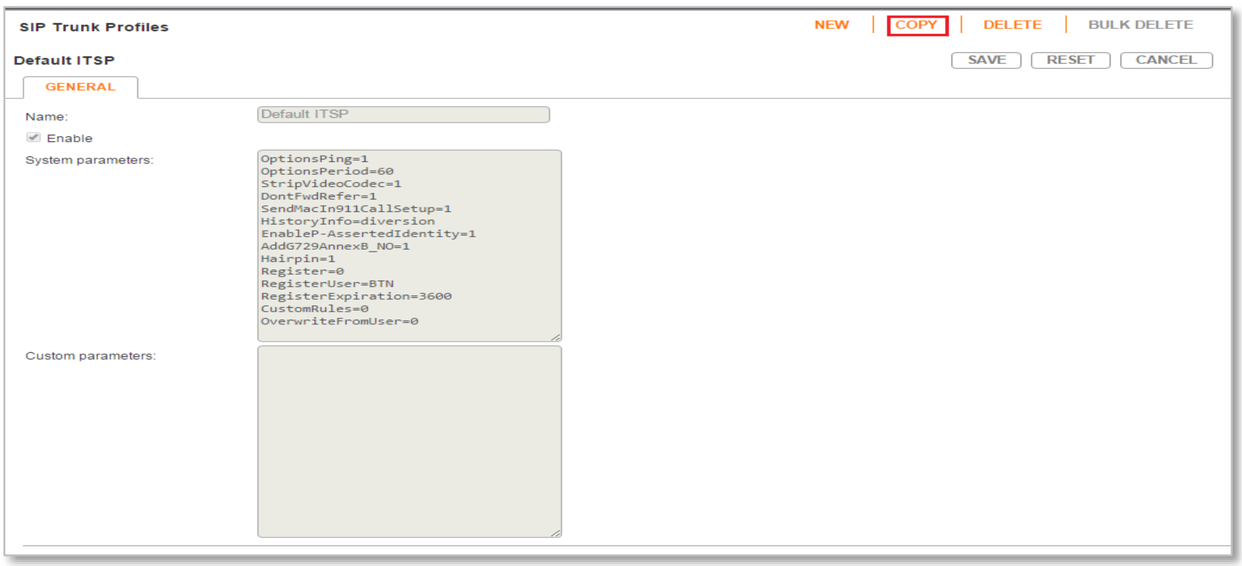
Trunk-to-Trunk transfer and tandem trunks:

- [ ] Hang up after silence of: minutes (1-1440)
- [ ] Hang up after: minutes (60-1440)

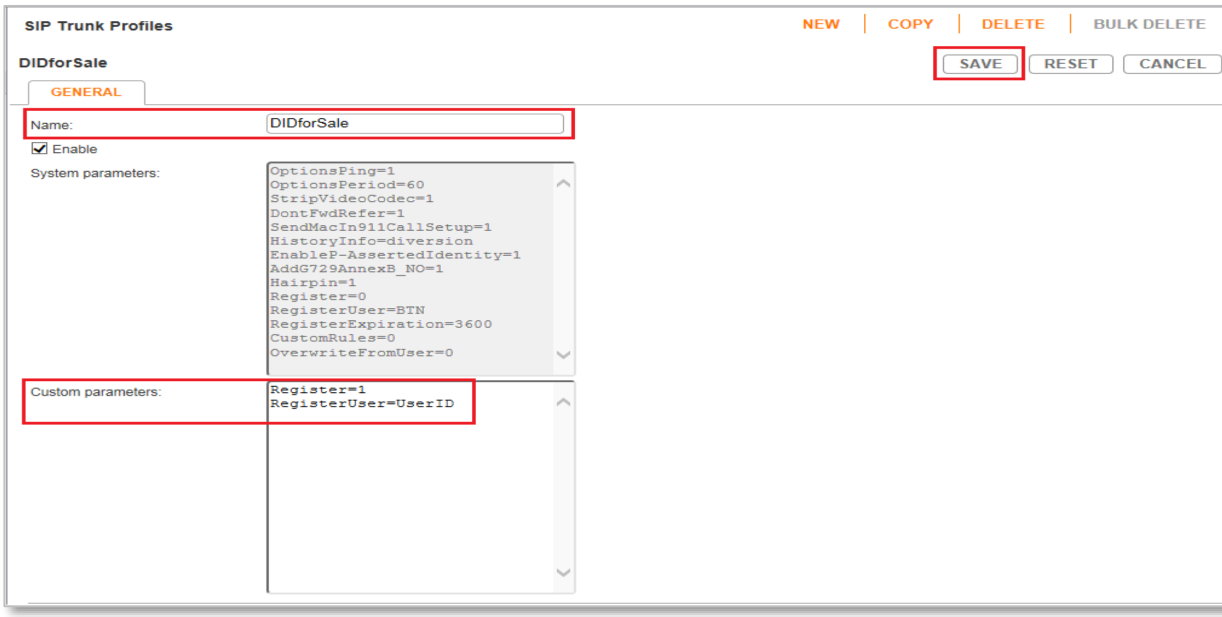
SIP Trunk Profile

This section describes the SIP Profile configuration required on the ShoreTel system to work with DIDforSale SIP Trunking. DIDforSale SIP Trunking requires custom SIP profile parameters to work properly with ShoreTel system. To create the custom SIP profile list, follow the steps listed below:

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



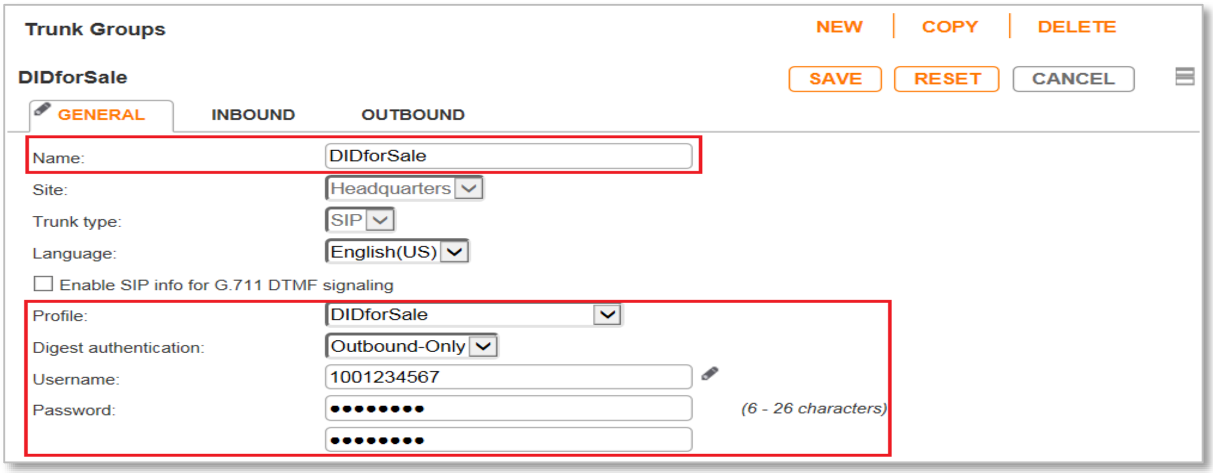
1. Set **Name**: Change from Default ITSP to DIDforSale
2. Set **Custom Parameters:** Enter the following custom parameters
    - Register=1
    - RegisterUser=UserID
3. Leave all other fields as default
4. Click **SAVE**



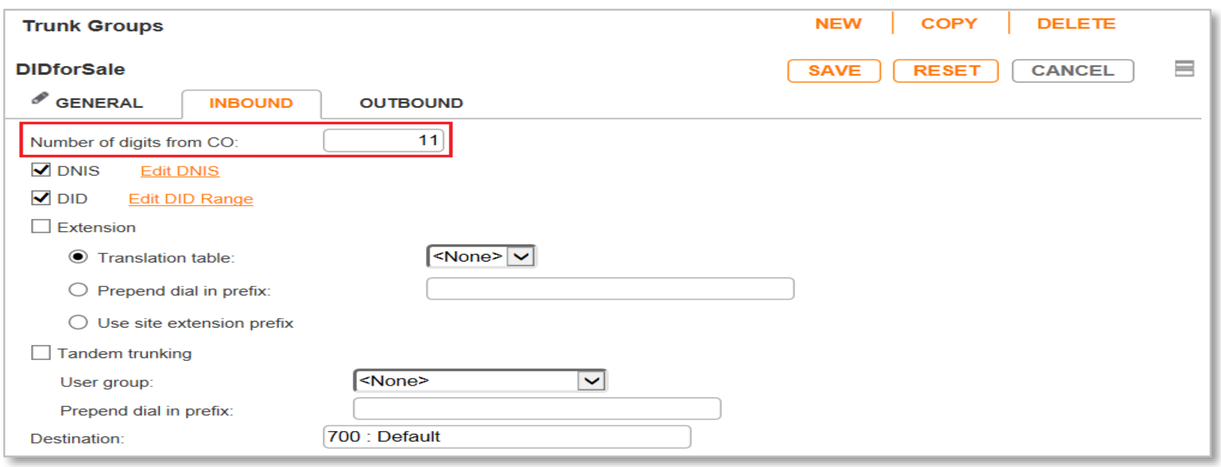
Trunk Group Configuration

This section describes the procedure to create SIP Trunk Group for DIDforSale SIP Trunking.

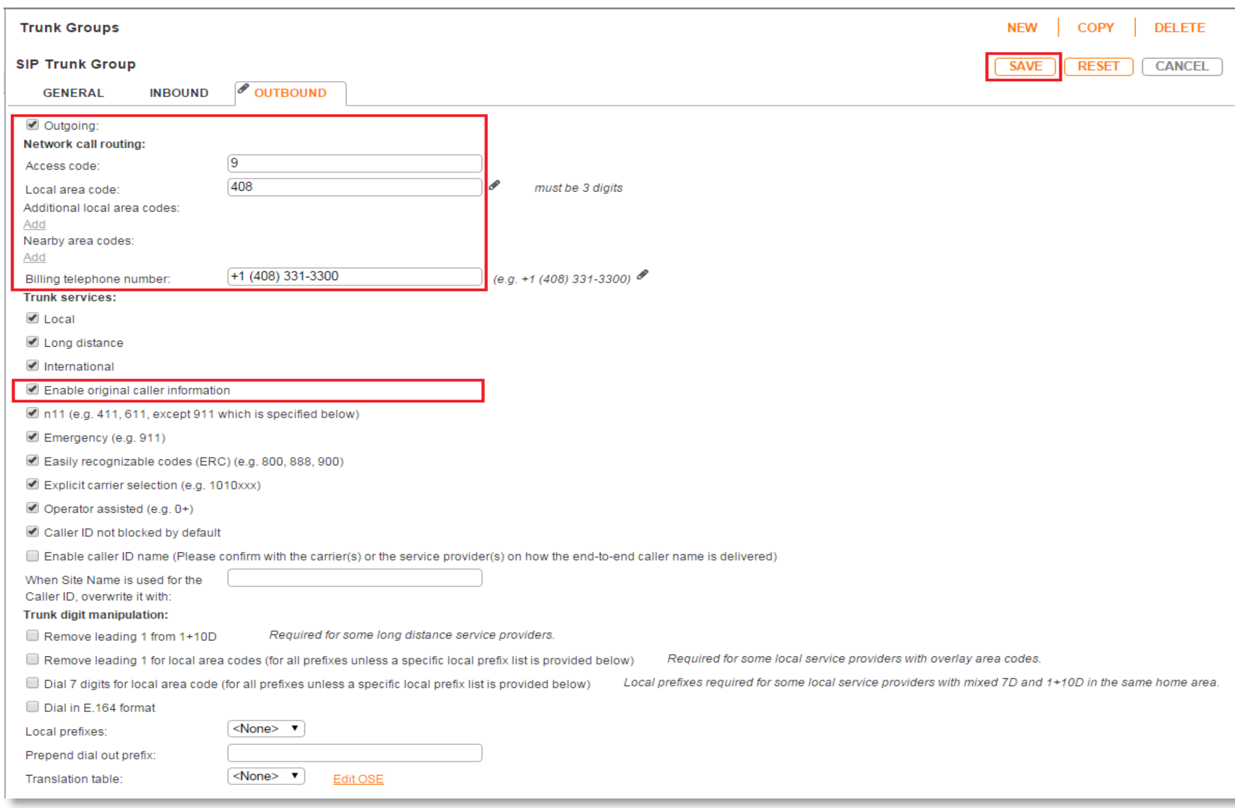
1. Navigate to Trunks > Trunk Groups > Trunk Groups
2. Click **NEW**
3. Set **Name**: SIP Trunk Group
4. Set **Trunk Type**: SIP
5. Set **Profile**: The SIP Profile *DIDforSale* is selected from drop down menu.
6. Set **Digest Authentication:** “Outbound-Only” is selected from the drop down menu.
7. Set **Username**: Enter the username provided by your DIDforSale Representative
8. Set **Password:** Enter the password provided by your DIDforSale Representative
9. Leave other fields as default.



10. Go to the INBOUND tab
11. Set **Number of Digits from CO:** 11 was used in this validation
12. Leave other fields as default.



13. Select the OUTBOUND tab
14. **Outgoing:** Checked
15. Set **Access Code:** 9 is used in this example
16. Set **Local Area Code:** 408 is used in this example
17. Set **Billing Telephone Number:** Pilot number will be provided by your DIDforSale Account Representative and must be kept confidential
18. Leave all other fields as default
19. Click **SAVE**

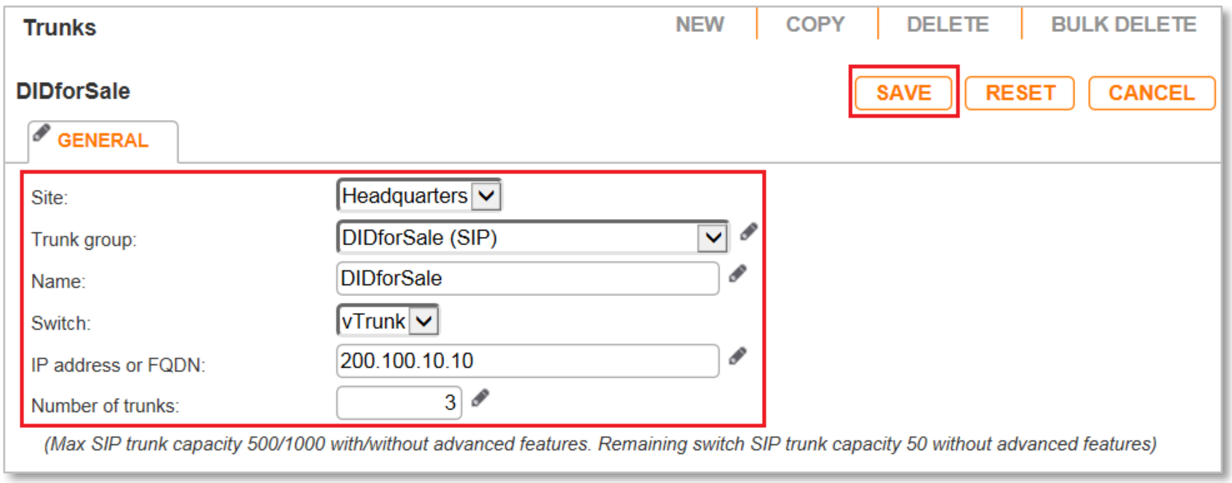


Individual Trunks

This following procedure outlines the steps required to create SIP Trunks for DIDforSale SIP Trunking.

1. Navigate to Trunks > Trunks
2. Click **NEW**

3. Set **Trunk Group**: *DIDforSale* (SIP)
4. Set **Name**: *DIDforSale* is used in this example
5. Set **Switch**: *vTrunk* is selected to host the SIP Trunks
6. Set **IP Address or FQDN**: Enter the IP Address of the DIDforSale SIP Trunks
7. Click **SAVE**

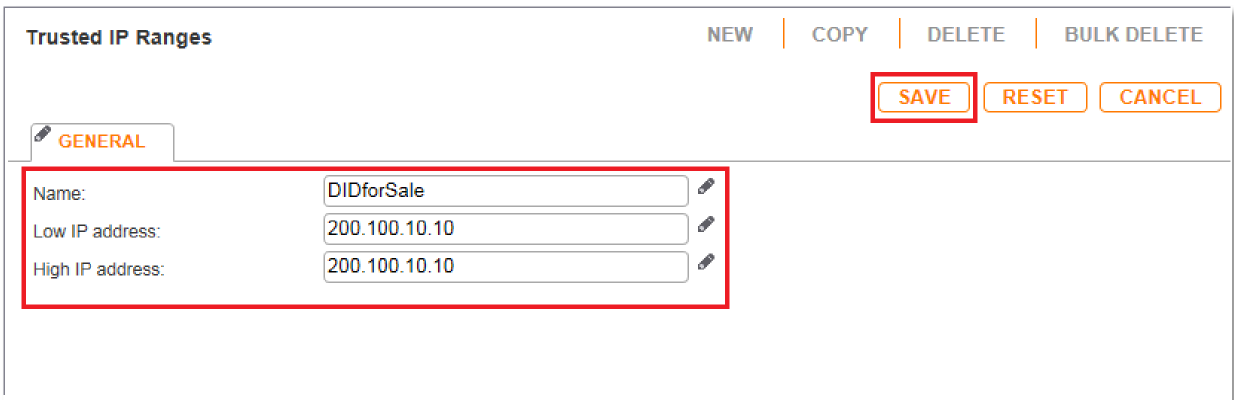


Trusted IP Ranges

The following procedure outlines the steps required to create Trusted IP Ranges for DIDforSale SIP Trunking.

1. Navigate to System > Trusted IP Ranges
2. Click **NEW** to create the new Trusted IP Range.

3. Set **Name**: *DIDforSale* is used in this example.
4. Set **Low IP address**: Enter the lower range of DIDforSale Signaling and Media IP Address.
5. Set **High IP address**: Enter the upper range of DIDforSale Signaling and Media IP Address.
6. Click **SAVE**



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	
2.1	PASS	Codec Negotiation	Verify codec negotiation between the SUT and the calling device with each side configured for a different codec	DIDforSale uses G711 as a preferred codec.
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	
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	

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

ID	Result	Name	Description	Notes
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 DIDforSale
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	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	CONDITIONAL PASS	Inbound / Outbound Fax Calls	Verify that inbound / outbound fax calls complete successfully	Only G711 Passthrough fax is supported.
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	

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	MakeMe 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	
5.1	PASS	SIP Registration		

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	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	CONDITIONAL PASS	Inbound / Outbound Fax calls	Verify that inbound / outbound fax calls complete successfully	Only G711 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	

ID	Result	Name	Description	Notes
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	

Conclusion

DIDforSale SIP Trunking was successfully validated and approved with ShoreTel Connect Onsite release.

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	February 2016	J.Rodriguez	Original App Note
1.1	March 2016	J.Rodriguez	Feedback incorporated

ShoreTel. Brilliantly simple business communications.

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

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