ShoreTel®

Innovation Network

Innovation Network Test Plan for Internet Telephony Service Providers (ITSPs)

Template Revision History:

Rev/ Date	Who	Description
V1.4 2012-5-30	Juan Rubio	Added several additional test cases and removed TPP references.
V1.3 2007 – 9-26	Juan Rubio	Added several additional test cases and removed some of the performance test cases.
V1.2a, 2007-8-20	Don Wallace	Updated to reflect ShoreTel® registered trade mark and other minor edits
V1.2, 2007-2-7	Don Wallace	Update to reflect TPP updated test methodology
V1.2, 2006-12-04	Chris Bajorek	Added Product Info, ShoreTel Platform Info, and Exec Summary sections to test plan since Test Report, Test Case Overview, and Results.Overview documents have been eliminated. Other changes include: (a) added reminder to review Application Note and add comments / suggestions. (b) added SIP trace capture line in each qualifying test case table.
V1.1, 2/23/06	Carla Reece	Edited to include review input from Jeff Ridley, ShoreTel.
V1.0, 12/14/05	Carla Reece	Initial release based on inputs from Jeff Ridley, ShoreTel.
1/11/2013	Neena Pemmaraju	Removed the Capacity test case, tekVizion.
1/15/2013	Suresh Kadiyala	Updated logo based on feedback from ShoreTel.
V1.1, 5/1/2013	Joshua Alphin / Eder Moncada C.	Test Report
V1.2, June 5, 2014	Craig Newman / Santhosh Immanuel	Completed test case 4.26 which was missed due to human error in the original test. This test cases also passes with release 13.1.
V1.3, June 5, 2014	Craig Newman / Santhosh Immanuel	Corrected tw telecom naming through the document
V1.4, July 24, 2014	Craig Newman	Clarified testing of recording scenarios with both release 13 and 14
V1.5, July 24, 2014	Craig Newman	Minor changes after review

ShoreTel Platform Information:

	[Component #1]	[Component #2]	[Component #3]
Machine Type:	Director	ShoreGear90	FortisVox/QFlex
OS Type and SP Level:			
Software & Version	ShoreWare 13.1	ShoreWare 13.1	
RAM			

Tw telecom Vendor Information:

Vendor company name:

Vendor rep present (if any):
Name:
Title:
Phone:
Email:

Primary testing contact:
Name: **Dick Richards**
Title: Sr. *Mgr. Voice & Signaling Engineering*
Phone: 303.566.1535
Email: dick.richards@twtelecom.com

Description of Service Tested

Features Supported

- Basic G.711ulaw calls
- Inbound and Outbound local, long distance and international calls
- Calling Party Number Presentation and Restriction
- Calling Name
- Blind Call Transfer
- Consultative Call Transfer
- Attended Call Transfer
- Intra- and Inter-site Conference
- Call Hold and Resume
- Call Forward All, Busy and No Answer
- Fax using G.711 Pass-through
- Outbound calls to IP and TDM networks
- Call Recording via Shoretel Communicator

ITSP service information:

Service name: tw telecom
Service type: SIP Trunk
Service release:
Options installed (if any):

Executive Summary

It's important to note that this testing was conducted without an Ingate SBC. This testing utilizes a FortisVox/GenBand Qflex SBC directly to the Shoretel system. Testing was revisited in June and July of 2014, in particular test case 4.26 regarding call recording was tested on multiple versions of Shoretel release 13 as well as Shoretel release 14.2. Specifically, releases 13.1, 13.3 (18.62.3403), and (18.63.1800) were tested. No problems were identified with recording from the Shoretel Communicator. Test Case 4.26 passed for all tested software releases.

Test cases Not Tested:
Test Case 4.19: ShoreTel Service Appliance Unified Communication System. Additional hardware required.
Test Case 4.30: Contact Center, (Optional). Agent tool bar not available for testing.
Test Case 4.31: ShoreTel Mobility Router (SMR) (Optional). Test case execution steps not defined yet.
Test Case 5.1: Registration or Digest Authentication (Optional). Not digest configured on test bed.

Test Case 2.4: DTMF Transmission – Out of Band / In Band. DTMF not supported via SIP INFO messages.

The following tests were confirmed as failed as indicated in the Application Note:
N/A

The following tests failed that were not indicated in the Application Note: N/A
N/A

Chronology of Testing Events

The table below describes the sequence of testing events, including any pauses or interruptions in the testing, and any new revisions that were received, and any test phases performed with new revisions.

Date Range	Description of Testing Events Performed During the Date Range
May 2013	Original Test
June-July 2014	Testing of Call Recording using Shoretel Communicator

Important Note to Tester: Application Note Guidelines

It is important that before the execution of this test plan you spend time thoroughly reviewing the latest revision of Application Note (AN) for the ITSP service being tested. This document can be obtained by contacting the primary ShoreTel contact for interoperability testing.

Throughout this test you are instructed to follow the AN as a guide to setup and configuration of the ShoreTel IP PBX platform as well as the ITSP service.

If you discover areas where the AN is not clear, or if the AN is lacking a procedure that you have discovered could save an installer time, then you should mark up a copy of the AN with Microsoft Word's revision marking enabled. This marked-up copy of the AN will then be delivered to ShoreTel along with this completed Test Plan when the test is concluded.

TESTER INITIAL AND DATE: [EMC, April 1st, 2013]

1. Test Cases Overview

This section presents an overview of all the SIP ITSP test cases available in this test plan template. The Notes field should be used to indicate if the test is required or optional, along with any other pertinent information.

Table 1-1: Initialization and Basic Calls

ID	Optional?	Name	Description	Notes
1.0	Mandatory	Configuration Application Note	The Innovation Network Lab will use the configuration application note provided by the vendor to configure the vendor's product to work with the ShoreTel system.	
1.1	Mandatory	Setup and initialization	Verify successful setup and initialization of the SUT	
1.2	Mandatory	Outbound Call (Domestic)	Verify calls outbound placed through the SUT reach the external destination.	
1.3	Mandatory	Inbound Call (Domestic)	Verify calls received by the SUT are routed to the default trunk group destination.	
1.4	Mandatory	Device restart – Power Loss	Verify that the SUT recovers after power loss to the SUT	
1.5	Mandatory	Device restart – Network Loss	Verify the SUT recovers after loss of network link to the SUT.	
1.6	Mandatory	All Trunks Busy – Inbound Callers	Verify an inbound callers hears busy tone when all channels/trunks are in use	
1.7	Mandatory	All Trunks Busy – Outbound Callers	Verify an outbound callers hears busy tone when all channels/trunks are in use	
1.8	Mandatory	Incomplete Inbound Calls	Verify proper call progress tones are provided and proper call teardown for incomplete inbound calls.	

Table 1-2: Media and DTMF Support

ID	Optional?	Name	Description	Notes
2.1	Mandatory	Media Support - ShoreTel Phone to SUT	Verify call connection and audio path from a ShoreTel phone to an external destination through the service provider using all supported codes with both sides set to a common codec.	Only G711 is supported

ID	Optional?	Name	Description	Notes
2.2	Mandatory	Media Support – SIP Reference to SUT	Verify call connection and audio path from a SIP Reference phones to an external destination through the service provider using all supported codes with both sides set to a common codec.	Only G711 is supported
2.3	Mandatory	Codec Negotiation	Verify codec negotiation between the SUT and the calling device with each side configured for a different codec.	(G729 not supported)
2.4	Mandatory	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.	G729 and SIP INFO not supported by TW
2.5	Mandatory	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.	
2.6	Mandatory	Auto Attendant Menu “Dial by Name”	Verify that inbound calls are properly terminated on the ShoreTel Auto Attendant menu and that you can transfer to the desired extension using the “Dial by Name” feature.	
2.7	Mandatory	Auto Attendant Menu checking Voice Mail mailbox	Verify that inbound calls are properly terminated on the ShoreTel Auto Attendant menu and that you can transfer to the Voice Mail Login Extension.	

Table 1-3: Performance & Quality of Service

ID		Name	Description	Notes
3.1	Optional	Voice Quality Service Levels	Verify the SUT can provide a voice quality SLA across the WAN from the customer premises to the SUT SIP gateway.	
3.2	Mandatory	Post Dial Delay	Verify that post dial delay is within acceptable limits.	
3.3	Mandatory	Billing Accuracy	Verify that all test calls made are accurately reflected in the SUT’s CDR and billing reports.	

Table 1-4: Enhanced Services and Features

ID		Name	Description	Notes
4.1	Mandatory	Caller ID Name and Number - Inbound	Verify that Caller ID name and number is received from SIP endpoint device	
4.2	Mandatory	Caller ID Name and Number - Outbound	Verify that Caller ID name and number is sent from SIP endpoint device	
4.3	Mandatory	Hold from SUT to SIP Reference	Verify successful hold and resume of connected call	
4.4	Mandatory	Call Forward - SUT	Verify outbound calls that are being forwarded by the SUT are redirected and connected to the appropriate destination.	
4.5	Mandatory	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	Mandatory	Call Transfer – blind	Verify a call connected from the SUT to the ShoreTel phone can be transferred to an alternate destination.	
4.7	Mandatory	Call Transfer – Consultative	Verify a call connected from the SUT to the ShoreTel phone can be transferred to an alternate destination.	
4.8	Mandatory	Conference – ad hoc	Verify successful ad hoc conference of three parties	
4.9	Mandatory	Inbound DID/DNIS	Verify the SUT provides inbound "dialed number information" and is correctly routed to the configured destination.	
4.10	Mandatory	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.	
4.11	Mandatory	Operator Assisted	Verify that 0+ calls are routed to an operator for calling assistance.	
4.12	Mandatory	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.	
4.13	Mandatory	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.	

ID		Name	Description	Notes
4.14	Mandatory	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	Mandatory	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.	
4.16	Mandatory	Call Forward – “FindMe”	Verify that inbound calls are forwarded to a user’s “FindMe” destination.	
4.17	Mandatory	Call Forward Always	Verify that inbound calls are immediately automatically forwarded to a user’s external destination.	
4.18	Mandatory	Inbound / Outbound Fax calls	Verify that inbound / outbound fax calls complete successfully.	
4.19	Optional	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.	
4.20	Mandatory	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.	
4.21	Mandatory	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	
4.22	Mandatory	Office Anywhere External	Verify that inbound calls are properly presented to the Office Anywhere External PSTN destination.	
4.23	Mandatory	Simul Ring	Verify that inbound calls are properly presented to the desired extension and the “Additional Phones” destinations.	
4.24	Mandatory	MakeMe Conference	Verify that an inbound call can be conferenced with three (or more) additional parties	
4.25	Mandatory	Park / Unpark	Verify that an inbound call can be parked and unparked	

ID		Name	Description	Notes
4.26	Mandatory	Call Recording	Verify that external calls can be recorded via the SIP Trunk using ShoreTel Communicator	
4.27	Mandatory	Silent Monitor / Barge-In / Whisper Page	Verify that external calls can be silently monitored, barged-in and whisper paged via the SUT.	
4.28	Mandatory	Long Duration – Inbound	Verify that an inbound call is established for a minimum of 30 minutes.	
4.29	Mandatory	Long Duration – Outbound	Verify that an outbound call is established for a minimum of 30 minutes.	
4.30	Optional	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.	
4.31	Optional	ShoreTel Mobility Router (SMR)	Verify that the SMR can be used with the SUT	

Table 1-5: Security

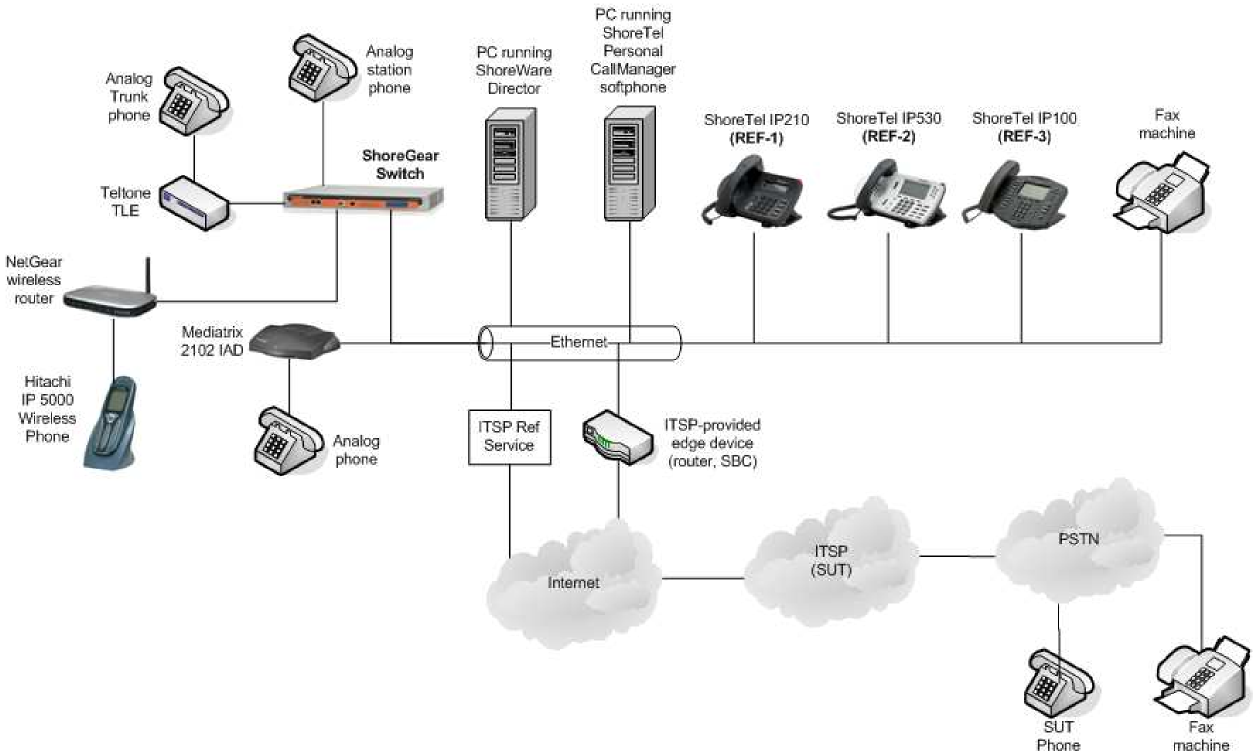
ID		Name	Description	Notes
5.1	Optional	Registration or Digest Authentication	Verify the SUT supports the use of registration or digest authentication for service access for inbound and outbound calls.	

2. Test Setup Requirements and Definition of Terms

2.1. Test Setup Diagram

The basic SIP ITSP ITP test setup is shown in Figure 1 below. Depending on specific test cases, this diagram may be changed to show additional components.

Figure 1: Basic ShoreTel SIP ITSP Test Setup



2.2. Definition of Terms

Term	Definition
Host VoIP platform	The IP PBX or call center equipment manufacturer's test lab setup, whether hosted at ShoreTel or made available remotely.
Reference SIP device	Permanent test lab phone chosen by the IP PBX or call center equipment manufacturer previous verified based on usage with the platform in the field and previous interoperability testing. The reference endpoints are used in this test plan primarily for lab verification as part of the quick-test process.
Vendor ITSP service	Vendor Internet Telephony Service Provider service being tested for interoperability with the host VoIP platform.
CODEC	COder DECoder: by specifying a preferred CODEC for an IP endpoint device – whether on the endpoint device itself or on the host VoIP platform for that device – you are choosing a standard compression method for reducing the bit rate (as well as the quality) of audio sent between devices.
SUT	System Under Test, A.K.A vendor ITSP service.
SUT Phone	An analog phone connected to the PSTN reachable through the SUT.
DN	Directory Number, the number dialed to reach a particular endpoint device or application. This number is typically configured on the host VoIP platform.

3. Quick Test Procedure

The Innovation Network Lab will verify the Host VoIP Platform environment's readiness to begin testing the SUT by performing a subset of the test plan with a Reference SIP phone. Once this set of tests has been completed, the Innovation Network Lab will then perform the same steps again with an SUT phone.

3.1. Purpose of Quick Test

The goal of this quick test is to:

1. Verify that the Host VoIP Platform has been configured correctly and is ready for the SUT
2. Verify that the SUT is able to perform the most basic functions in the host VoIP Platform
3. Identify warning flags early if fundamental compatibility issues exist with the SUT

3.2. Quick Test Procedure:

1. Log the following information:

   3rd Party ITSP service being tested: tw telecom

   3rd Party ITSP service version: 5.0

   ShoreTel version: 13.1

2. Perform the test cases (see section 0) listed below with **Reference SIP ITSP**, logging results into "Reference device" field in Table 3-1 below.

3. Perform the test cases (see section 0) listed below with **vendor SIP system** under test (SUT), logging results into "Vendor system (SUT)" field in Table 3-1 below.

Table 3-1: Quick Test Results

Test case:	Reference device:	Vendor (SUT) device:
1.1		
1.2*		
2.4**		
4.7***		

* ShoreTel IP-560 to SUT Phone
** Use IP-560, G.711 only. Test in-band and out-of-band (RFC2833) with SUT in both originating and terminating end of the DTMF transmission.
*** Call #1 Only - Transfer from ShoreTel phone to another phone.

IMPORTANT NOTE TO TESTER: SIP TRACE CAPTURES FOR SELECTED TEST CASES WILL NEED TO BE SAVED AND DELIVERED TO SHORETEL AT THE CONCLUSION OF THIS TEST. If the SIP Trace line in the test case table indicates “Yes”, then you must capture the SIP message trace(s) and rename the capture file using the following general guideline: ***[itsp vendor name]-[test case number]-[variation].cap***. For example, “XYZ-1.0-trunk to ST.cap” would be one capture for ITSP vendor XYZ for test case 1.0 in the trunk-to-ShoreTel direction.

4. Initialization and Basic Features Test Cases

Perform all test cases in this section, logging results and notes as appropriate.

ID	**1.0**
Name	**Configuration Application Note**
Mandatory or Optional:	**Mandatory**
Description	**The Innovation Network Lab will use the configuration application note provided by the vendor to configure the vendor’s product to work with the ShoreTel system.**
SIP Trace? No	Trace 1 filename: Trace 2 filename:
Test steps	1. Use the configuration application note provided by the vendor to configure the vendor’s product to work with the ShoreTel system 2. Comment on the accuracy and completeness of the configuration application note provided by the vendor.
Result	Pass
Notes	

ID	**1.1**
Name	**Setup and initialization**
Mandatory or Optional:	**Mandatory**
Description	**Verify successful setup and initialization of the SUT**
SIP Trace? No	Trace 1 filename: Trace 2 filename:
Test steps	1. Configure **ITSP SUT** as static IP; Configure **ShoreTel Individual Trunk** as static 2. Perform any **SUT** configuration necessary to register with the appropriate server(s). 3. Verify that **SUT** is ready to place and receive calls.
Result	Pass
Notes	SIP connectivity to the ITSP must be configured as a static trunk with static IP addresses used for both the ShoreTel side and the SUT.

ID	**1.2**
Name	**Outbound Call (Domestic)**
Mandatory or Optional:	**Mandatory**
Description	**Verify calls outbound placed through the SUT reach the external destination.**
SIP Trace? Yes 1 call	Trace 1 filename: TWTelecom-1.2-OutboundCall Trace 2 filename:
Test steps	Place two calls with normal dialing to each device shown in Table 4-1, column 1. 1. Place call from **ShoreTel phone** to **SUT phone**. 2. Confirm normal processing during the call including ringback to the caller, and that the called telephone rings. 3. Verify that the call is connected successfully, and that media is established. 4. Hang up the call from the **ShoreTel phone**. 5. Confirm proper call teardown at the end of the call. 6. Place call from **ShoreTel phone** to **SUT phone**. 7. Confirm normal processing during the call including ringback to the caller, and that the called telephone rings. 8. Verify that the call is connected successfully, and that media is established. 9. Hang up the call from the **SUT phone**. 10. Confirm proper call teardown at the end of the call.
Result	Log results into Table 4-1
Notes	Pass

ID	**1.3**
Name	**Inbound Call (Domestic)**
Mandatory or Optional:	**Mandatory**
Description	**Verify calls received by the SUT are routed to the default trunk group destination.**
SIP Trace? Yes 1 call	Trace 1 filename: TWTelecom-1.3-InboundCall Trace 2 filename:
Test steps	Receive calls with normal dialing from each device shown in Table 4-1, column 2. 1. Place call from **SUT phone** to **ShoreTel side**. 2. Confirm normal processing during the call including ringback to the caller, and that the called telephone rings. 3. Verify that the call is connected successfully, and that media is established. 4. Hang up the call from the **ShoreTel side**. 5. Confirm proper call teardown at the end of the call. 6. Place call from **SUT phone** to **ShoreTel side**. 7. Confirm normal processing during the call including ringback to the caller, and that the called telephone rings. 8. Verify that the call is connected successfully, and that media is established. 9. Hang up the call from the **SUT phone**. 10. Confirm proper call teardown at the end of the call.
Result	Log results into Table 4-1.
Notes	Pass

Table 4-1: Results from Test Steps 1.2 & 1.3

Device or Application	Test Step 1.2 ShoreTel Originating	Test Step 1.3 ShoreTel Terminating
ShoreTel IP565	OK	OK
ShoreTel IP230 **ShoreTel IP110**	OK	OK
Analog Phone	OK	OK
SIP Ref – can be any validated IN member SIP handset	N/A	N/A
Voicemail	OK	OK
Workgroup to available agent	OK	OK
Workgroup queue then to available agent	OK	OK

ID	**1.4**
Name	**Device restart –Power Loss**
Mandatory or Optional:	**Mandatory**
Description	**Verify that the SUT recovers after power loss to the SUT**
SIP Trace? No	Trace 1 filename: Trace 2 filename:
Test steps	1. With **SUT** in a READY/IDLE state, remove power from the **voice switch associated with the SUT**. 2. Wait several seconds, then re-apply power to the **voice switch**. 3. Verify that the **voice switch** re-initializes successfully and returns to a READY/IDLE state. 4. Place call from **REF-1** to **SUT phone**. 5. Place call from **SUT phone** to **REF-1**.
Result	Pass
Notes	

ID	**1.5**
Name	**Device restart – Network Loss**
Mandatory or Optional:	**Mandatory**
Description	**Verify the SUT recovers after loss of network link to the SUT.**
SIP Trace? No	Trace 1 filename: Trace 2 filename:
Test steps	1. With **SUT** in a READY/IDLE state, remove network connection to the device. 2. Wait several seconds, then re-apply network connection to the **SUT** device. 3. Verify that the device re-initializes successfully and returns to a READY/IDLE state. 4. Place call from **REF-1** to **SUT phone**. 5. Place call from **SUT phone** to **REF-1**.
Result	Pass
Notes	

ID	**1.6**
Name	**All Trunks Busy – Inbound Callers**
Mandatory or Optional:	**Mandatory**
Description	**Verify an inbound callers hears busy tone when all channels/trunks are in use**
SIP Trace? Yes	Trace 1 filename: TWTelecomm-1.6-All Trunks Busy – Inbound Callers Trace 2 filename:
Test steps	1. Start with all channels/trunks busy on the **SUT**. 2. Place a call from any other device to an **SUT phone**. 3. Verify that the caller hears busy tone.
Result	Pass
Notes	**Shoretel responses back with a 500 Server Internal Error**

ID	**1.7**
Name	**All Trunks Busy – Outbound Callers**
Mandatory or Optional:	**Mandatory**
Description	**Verify an outbound callers hears busy tone when all channels/trunks are in use**
SIP Trace? No	Trace 1 filename: Trace 2 filename:
Test steps	1. Start with all channels/trunks busy that are configured in the ShoreTel trunk group. 2. Place a call from an **SUT phone** to any other device. 3. Verify that the caller hears busy tone.
Result	Pass
Notes	**The ShoreTel user making the call must be configured with access to ONLY the SIP trunk group.**

ID	**1.8**
Name	**Incomplete Inbound Calls**
Mandatory or Optional:	**Mandatory**
Description	**Verify proper call progress tones are provided and proper call teardown for incomplete inbound calls.**
SIP Trace? Yes	Trace 1 filename: TWTelecomm-1.8-Incomplete Inbound Calls-C1 Trace 2 filename: TWTelecomm-1.8-Incomplete Inbound Calls-C2 Trace 3 filename: TWTelecomm-1.8-Incomplete Inbound Calls-C3
Test steps	**Call #1** 1. From **REF**, call the **SUT phone** device and verify that the call rings. 2. From **REF**, hang up the call. 3. Verify that the **SUT phone** device stops ringing. 4. Verify that picking up the **REF** again results in dial tone. **Call #2** 1. From **REF**, call the **SUT phone** device and verify that the call rings. 2. Do not answer the **SUT phone**. 3. Verify that the **SUT phone** device keeps ringing. 4. Verify that the **REF** keeps hearing ringing. 5. Verify that the **REF** times out after some point. **Call #3** 1. Take the **SUT phone** device off-hook (if more than 1 line is supported on the device, make sure all lines are busy) 2. From **REF**, call the **SUT phone** device. 3. Verify that the **REF** device hears a busy tone.
Result	Pass
Notes	

5. Media and DTMF Support Tests

Perform all test cases in this section, logging results and notes as appropriate.

ID	**2.1**
Name	**Media Support - ShoreTel Phone to SUT**
Mandatory or Optional:	**Mandatory**
Description	**Verify call connection and audio path from a ShoreTel phone to an external destination through the service provider using all supported codes with both sides set to a common codec.**
SIP Trace? Yes one for each codec supported	Trace 1 filename: TWTelecomm-2.1-OutboundG711-MediaSupport Trace 2 filename: TWTelecomm-2.1-InboundG711-MediaSupport
Test steps	1. For each CODEC supported by both **SUT** and **REF-x** devices and the server/switch, configure the devices for that CODEC and Intrasite CODEC. 2. Attach the packet analyzer in between the **SUT** and **REF-x** devices; do not yet enable capturing. 3. Re-initialize both **SUT** and **REF-x** devices and verify that they are in a READY/IDLE state. 4. Enable packet capturing on the analyzer. 5. Perform tests 1.2 Outbound Call (Domestic) and 1.3 Inbound Call (Domestic) for each supported CODEC with packet capturing enabled. 6. Disable packet capturing and find one decoded packet for both **SUT phone** and **REF-x** in the list of captured packets that specifies which CODEC was used for each call. 7. Verify that the CODEC specified in the decoded packet matches the configured CODEC for each respective device. There is no need to save packet traces unless there is a problem to report.
Result	**Pass with G711-Ulaw, TW only support G711**
Notes	If ITSP supports any additional codecs be sure to include them in the tests.

ID	**2.2**
Name	**Media Support – SIP Reference to SUT**
Mandatory or Optional:	**Mandatory**
Description	**Verify call connection and audio path from a SIP Reference phones to an external destination through the service provider using all supported codes with both sides set to a common codec.**
SIP Trace? No	Trace 1 filename: Trace 2 filename:
Test steps	1. For each CODEC supported by both **SUT** and SIP-Ref devices and the server/switch, configure the devices for that CODEC and Intrasite CODEC. 2. Attach the packet analyzer in between the **SUT** and SIP-Ref devices; do not yet enable capturing. 3. Re-initialize both **SUT** and SIP-Ref devices and verify that they are in a READY/IDLE state. 4. Enable packet capturing on the analyzer. 5. Perform tests 1.2 Outbound Call (Domestic) and 1.3 Inbound Call (Domestic) for each supported CODEC with packet capturing enabled.. 6. Disable packet capturing and find one decoded packet for both **SUT phone** and SIP-Ref in the list of captured packets that specifies which CODEC was used for each call. 7. Verify that the CODEC specified in the decoded packet matches the configured CODEC for each respective device. There is no need to save packet traces unless there is a problem to report.
Result	**Pass with G711-Ulaw, TW only support G711**
Notes	If ITSP supports any additional codecs be sure to include them in the tests.

ID	**2.3**
Name	**Codec Negotiation**
Mandatory or Optional:	**Mandatory**
Description	**Verify codec negotiation between the SUT and the calling device with each side configured for a different codec.**
SIP Trace? No	Trace 1 filename: Trace 2 filename:
Test steps	1. Configure **REF-x** device to G711, configure **SUT** to G729 2. Attach the packet analyzer in between the **REF-x** and **SUT** devices; do not yet enable capturing. 3. Re-initialize both **REF-x** and **SUT** devices and verify that they are in a READY/IDLE state. 4. Enable packet capturing on the analyzer. 5. Place call from **REF-x** device to **SUT** device. 6. Disable packet capturing and inspect the decoded packets to verify that the CODEC negotiation occurred and **SUT** device negotiated to G711 if supported. 7. Enable packet capturing on the analyzer. 8. Place call from **SUT** device to **REF-x** device. 9. Disable packet capturing and inspect the decoded packets to verify that the CODEC negotiation occurred and **SUT** device negotiated to G711 if supported. 10. Configure **REF-x** device to G729, configure **SUT** device G711. 11. Attach the packet analyzer in between the **REF-x** and **SUT** devices; do not yet enable capturing. 12. Re-initialize both **REF-x** and **SUT** devices and verify that they are in a READY/IDLE state. 13. Enable packet capturing on the analyzer. 14. Place call from **SUT** device to **REF-x** device. 15. Disable packet capturing and inspect the decoded packets to verify that the CODEC negotiation occurred and **SUT** device negotiated to G729 if supported. 16. Enable packet capturing on the analyzer. 17. Place call from **REF-x** device to **SUT** device. 18. Disable packet capturing and inspect the decoded packets to verify that the CODEC negotiation occurred and **SUT** device negotiated to G729 if supported.
Result	**Record results in** Table 5-1 **below**
Notes	• **Codecs Supported by ShoreTel: G711-Ulaw, G729** • **Negotiation will depend on how SIP device is configured in ShoreTel system, following the preferred codecs defined in the system for the different types of calls.** • **It is OK if the ITSP only supports G.711 codecs. In this case, do the negotiation with the ShoreTel set to a different codec.**

Table 5-1: Results from Test Step 2.3

Sequence #'s	ShoreTel REF-x codec setting	Direction of call	SUT codec setting	Results
1-6	G711	→	G729	G729 N/S by TW Telecom. Call established with G711
7-9	G711	←	G729	G729 N/S by TW Telecom. Call established with G711
10-15	G729	←	G711	G729 N/S by TW Telecom, G729 offered but call established with G711
16-18	G729	→	G711	G729 N/S by TW Telecom, G729 offered but call established with G711

ID	**2.4**
Name	**DTMF Transmission – Out of Band / In Band**
Mandatory or Optional:	**Mandatory**
Description	**Verify transmission of in-band and out-of-band digits per RFC 2833 for various devices connected to the SUT.**
SIP Trace? Yes 1 for each codec supported	Trace 1 filename: TWTelecomm-2.4-DTMF TransmissionRFC2833 Trace 2 filename:
Test steps	1. Configure the **SUT phone** device for out-of-band digit transmission **Calls originating with the SUT** 2. From **SUT phone**, place call to terminating device specified in Table 5-2. 3. Using the touch tone keypad, navigate through the application menus, or just press touch-tone digits. 4. For calls terminating at an application, verify that the touch tones are recognized and the menus are traversed as expected. 5. For calls terminating in a device, verify that touch-tones pressed in both directions are heard at the other end of the call. **Calls terminating with the SUT** 6. From each originating device (specified in Table 5-3) if capable, place call to **SUT phone**. 7. Using the touch tone keypad or application generate touch-tone digits. 8. Verify that touch-tones pressed in both directions are heard at the other end of the call.
Result	For calls originating with the SUT, log results into Table 5-2. For calls terminating with the SUT, log results into Table 5-3.
Notes	SIP INFO not supported by TW

Table 5-2: Results from Test Step 2.4 (Originating Device always = SUT phone)

Terminating Device or Application	DTMF transmission Type		
	G711 SIP Info (confirm inband DTMF)	G711 RFC 2833	G729 RFC 2833
ShoreTel IP230 ShoreTel IP110	N/S	Pass	N/S
ShoreTel IP565 ShoreTel IP560	N/S	Pass	N/S
Analog Phone	N/S	Pass	N/S
SIP Ref – Can be any validated IN member SIP handset	N/T	N/T	N/S
AutoAttendant	N/S	Pass	N/S
Backup AutoAttendant	N/T	N/T	N/S

Table 5-3: Results from Test Step 2.4 (Terminating Device always = SUT phone)

Originating Device or Application	DTMF transmission Type		
	G711 SIP Info (confirm inband DTMF)	G711 RFC 2833	G729 RFC 2833
ShoreTel IP230 ShoreTel IP110	N/S	Pass	N/S
ShoreTel IP565 ShoreTel IP560	N/S	Pass	N/S
Analog Phone	N/S	Pass	N/S
ShoreTel softphone	N/T	N/T	N/S
SIP Ref – Can be any validated IN member SIP handset	N/T	N/T	N/S

ID	**2.5**
Name	**Auto Attendant Menu**
Mandatory or Optional?	**Mandatory**
Description	**Verify that inbound calls are properly terminated on the ShoreTel Auto Attendant menu and that you can transfer to the desired extension.**
SIP Trace? **Yes**	Trace 1 Filename: TWTelecomm-2.5-All Auto Attendant Menu Trace 2 Filename:
Test steps	1. Configure a ShoreTel Auto Attendant menu "Multiple digits" parameter for "Transfer to Extension". 2. Define the Trunk Group destination as the AA menu configured in step 1, or assign a DNIS map or DID to the AA menu. 3. Place an inbound call. 4. Verify that you hear the AA menu prompt you for input. 5. Press the desired digits for the extension you want to be transferred to. You should hear "Please hold while I transfer your call". 6. Verify that the desired extension phone rings and the call is answered properly. 7. Verify two way audio.
Result	Pass
Notes	

ID	**2.6**
Name	**Auto Attendant Menu "Dial by Name"**
Mandatory or Optional?	**Mandatory**
Description	**Verify that inbound calls are properly terminated on the ShoreTel Auto Attendant menu and that you can transfer to the desired extension using the "Dial by Name" feature.**
SIP Trace? **No**	Trace 1 Filename: Trace 2 Filename:
Test steps	1. Configure a ShoreTel Auto Attendant menu option for "Dial by first name". 2. Define the Trunk Group destination as the AA menu configured in step 1, or assign a DNIS map or DID to the AA menu. 3. Place an inbound call. 4. Verify that you hear the AA menu prompt you for input. 5. Press the digit that corresponds to the "Dial by Name" feature. When prompted spell the first name of the user you want the call to go to and follow the prompts. 6. Verify that the desired extension phone rings and the call is answered properly. 7. Verify two way audio.
Result	Pass
Notes	

ID	2.7
Name	**Auto Attendant Menu checking Voice Mail mailbox**
Mandatory or Optional?	Mandatory
Description	**Verify that inbound calls are properly terminated on the ShoreTel Auto Attendant menu and that you can transfer to the Voice Mail Login Extension.**
SIP Trace? No	Trace 1 Filename: Trace 2 Filename:
Test steps	1. Configure a ShoreTel Auto Attendant menu option for " Go to menu" and define the destination as the Voice Mail Login extension. 2. Place an inbound call. 3. Verify that the you hear the AA menu prompt you for input. 4. Press the digit that corresponds to the "Go to menu" option. 5. Verify that the call is routed to the ShoreTel Voice Mail server and prompts you to enter an extension, enter the desired extension and password. 6. Navigate through the menu and listen to any messages, then change the user's Call Handling Mode to "In a Meeting" and verify that user's CHM is changed.
Result	Pass
Notes	

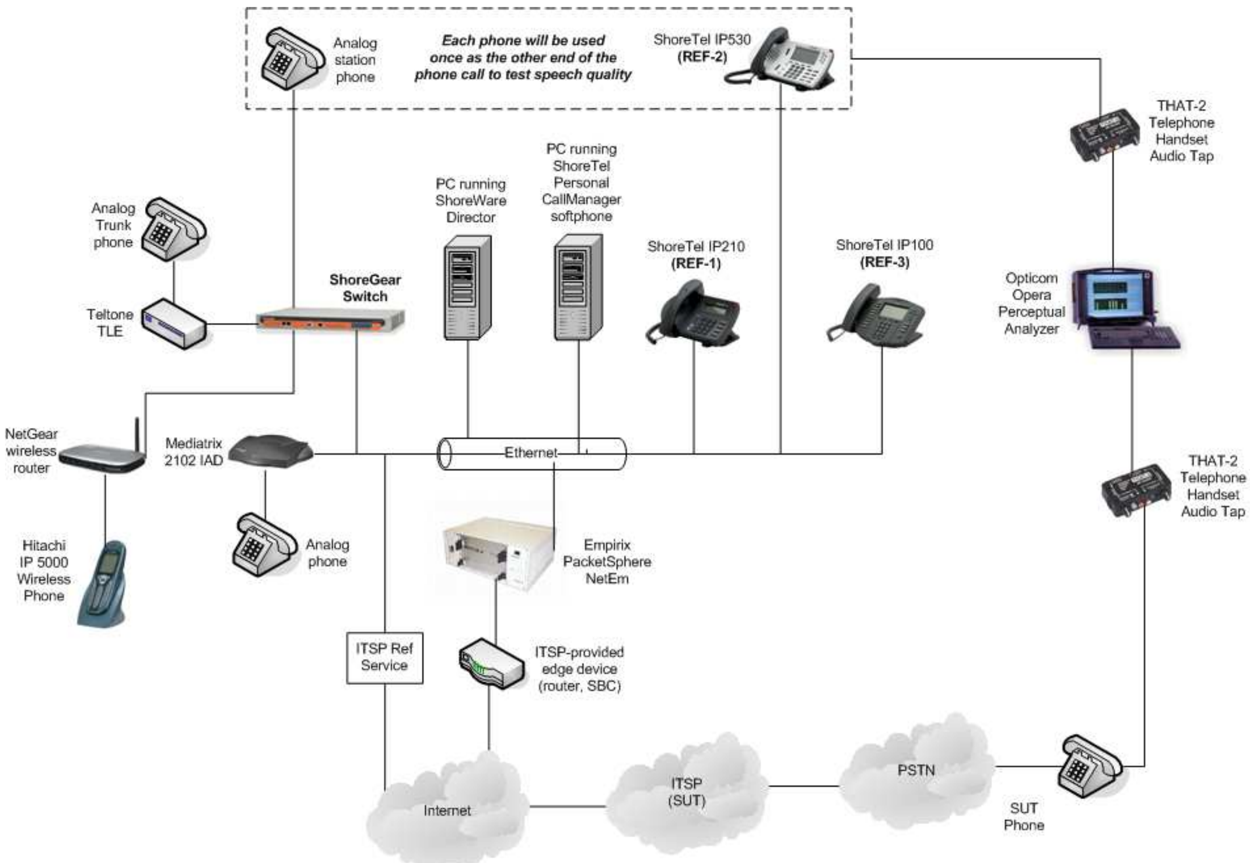
6. Quality of Service Tests

Perform all test cases in this section, logging results and notes as appropriate.

6.1. Quality of Service Test Setup Diagram

The basic Quality of Service SIP ITSP ITP test setup is shown in Figure 2 below. Depending on specific test cases, this diagram may be changed to show additional components.

Figure 2: Basic Performance ShoreTel SIP ITSP Test Setup



6.2. Quality of Service Test Cases

ID	**3.1**
Name	**Voice Quality Service Levels**
Mandatory or Optional:	**Optional**
Description	**Verify the SUT can provide a voice quality SLA across the WAN from the customer premises to the SUT SIP gateway.**
Test steps	1. Measure and record the SLA latency 2. Measure and record the SLA jitter
Result	SLA latency: SLA jitter:
Notes	**The number of trunks configured must match the available bandwidth based on codecs. For example, if 64K bandwidth with G.729, there only be 2 trunks configured. Bandwidth use by codec can be found in the ShoreTel Planning and Installation Guide.**

ID	**3.2**
Name	**Post Dial Delay**
Mandatory or Optional:	**Mandatory**
Description	**Verify that post dial delay is within acceptable limits.**
Test steps	1. Pick up the **SUT phone**. 2. Dial the DTMF digits for a ShoreTel extension. 3. Measure the time between the transmission of the last DTMF digit in the dial string and when ringback is received. 4. Verify that this value is less than 3 seconds.
Result	Pass
Notes	

ID	**3.3**
Name	**Billing Accuracy**
Mandatory or Optional:	**Mandatory**
Description	**Verify that all test calls made are accurately reflected in the SUT's CDR and billing reports.**
Test steps	1. Place 10 calls – noting number dialed and length of call. 2. Access or request the billing record from the SUT. 3. Verify accuracy of billing record.
Result	**Call 1:** number called:________1214-242-5915_____call length:_90 sec__ **Call 2:** number called:________19728522617 _____ call length:_104 sec__ **Call 3:** number called:_______ 19728522617______ call length:__87 sec___ **Call 4:** number called:_______ 19728053445 ______ call length:___195 sec_____ **Call 5:** number called:_______ 19728053445_______ call length:_75 sec____ **Call 6:** number called:_______19728522617________ call length:_95 sec_______ **Call 7:** number called:_______ 12144450311________call length:__100 sec______ **Call 8:** number called:________12142425915_______ call length:__75 sec______ **Call 9:** number called:________ 12144450310_______call length:_130 sec_______ **Call 10:** number called:_______1 9728053445_______call length:__30 sec______
Notes	If SUT doesn't provide access to billing records use ShoreTel Director reports.

7. Enhanced Services and Features Test Cases

Perform all test cases in this section, logging Results and Notes as appropriate.

ID	**4.1**
Name	**Caller ID Name and Number - Inbound**
Mandatory or Optional:	**Mandatory**
Description	**Verify that Caller ID name and number is received from SIP endpoint device**
SIP Trace? Yes	Trace 1 filename: TWTelecomm-4.1-Caller ID Name and Number - Inbound.pcap Trace 2 filename:-
Test steps	1. Activate VB Trunk Test tool. 2. From **SUT phone**, place call to **ShoreTel phone**. 3. Verify Caller ID number is received on **ShoreTel phone** from **SUT phone**. Note: if CallerID is not available on the phone, the verification can be done in the VB Trunk Test tool by checking what the Inside-ID says in the SIP messages for the call.
Result	Pass
Notes	

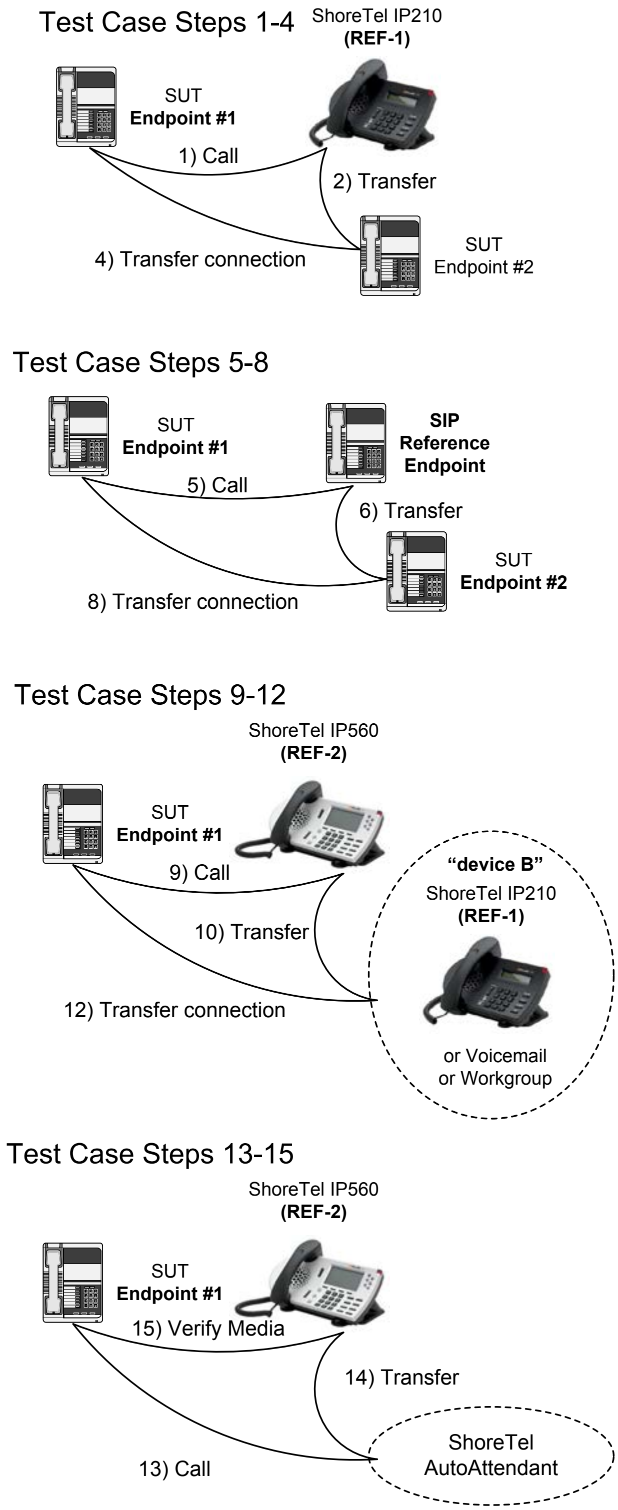
ID	**4.2**
Name	**Caller ID Name and Number - Outbound**
Mandatory or Optional:	**Mandatory**
Description	**Verify that Caller ID name and number is sent from SIP endpoint device**
SIP Trace? Yes	Trace 1 filename: Trace 2 filename:
Test steps	1. From **ShoreTel phone**, place call to **SUT phone**. 2. Verify Caller ID number is received from **ShoreTel phone** 3. From **ShoreTel phone,** place call to an Analog Trunk Phone. 4. Verify Caller ID number is received from **ShoreTel phone**. Note: if CallerID is not available on the phone,the verification can be done in the VB Trunk Test tool by checking what the Inside-ID says in the SIP messages for the call.
Result	Pass
Notes	

ID	**4.3**
Name	**Hold from SUT to SIP Reference**
Mandatory or Optional:	**Mandatory**
Description	**Verify successful hold and resume of connected call**
SIP Trace? Yes	Trace 1 filename: TWTelecomm-4.3-Hold from SUT to SIP Reference.pcap Trace 2 filename:
Test steps	1. From **SUT phone**, place call to **SIP-Ref**. 2. Answer incoming call from **SUT phone**. 3. Verify that the call is connected successfully. 4. From **SUT phone**, place **SIP-Ref** on hold. 5. Verify that the audio path is dropped. 6. Verify that the **external** device appropriately indicates a held call still in progress. 7. From **SUT phone**, resume the held call. 8. Verify that the audio path is restored and that the call proceeds as expected. 9. From **SIP-Ref**, place **SUT phone** on hold. 10. Verify that the audio path is dropped. 11. Verify that the **SIP-Ref** device appropriately indicates a held call still in progress. 12. From **SIP-Ref**, resume the held call. 13. Verify that the audio path is restored and that the call proceeds as expected.
Result	Pass
Notes	Verify with Music On Hold (MOH)

ID	**4.4**
Name	**Call Forward - SUT**
Mandatory or Optional:	**Mandatory**
Description	**Verify outbound calls that are being forwarded by the SUT are redirected and connected to the appropriate destination.**
SIP Trace? No	Trace 1 filename: Trace 2 filename:
Test steps	1. Configure the **SUT phone** device to forward all calls to **ShoreTel phone**. 2. From **REF-x**, place call to **SUT phone**. 3. Verify that the call is immediately forwarded to **ShoreTel phone**. 4. Answer incoming call. 5. Verify that the call is connected successfully and there is two-way audio path. 6. Disconnect the call from the ShoreTel phone. 7. Verify that all call legs are properly disconnected.
Result	Pass
Notes	

ID	**4.5**
Name	**Call Forward – External PSTN Number**
Mandatory or Optional:	**Mandatory**
Description	**Verify outbound calls that are being forwarded by the SUT are redirected and connected to the appropriate destination.**
SIP Trace? Yes	Trace 1 filename: Trace 2 filename:
Test steps	1. Configure the **SUT phone** device to forward all calls to an external **PSTN telephone** number. 2. From **REF-x**, place call to **SUT phone**. 3. Verify that the call is immediately forwarded to **PSTN** destination. 4. Answer incoming call. 5. Verify that the call is connected successfully and there is two-way audio path. 6. Disconnect the call from the originating party. 7. Verify that the call is properly disconnected.
Result	Pass
Notes	

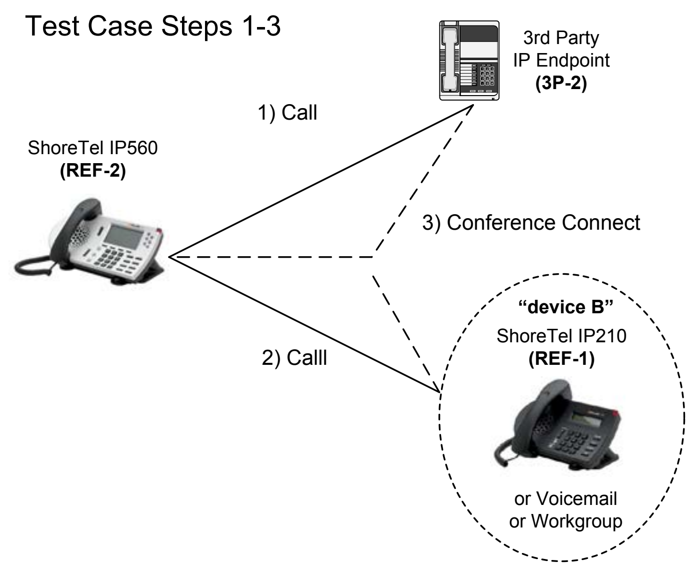
Figure 3: Transfer Scenarios Drawing (for Test Cases 4.6 and 4.7)



<table>
<tr><td>ID</td><td>4.6</td></tr>
<tr><td>Name</td><td>Call Transfer – blind</td></tr>
<tr><td>Mandatory or Optional:</td><td>Mandatory</td></tr>
<tr><td>Description</td><td>Verify a call connected from the SUT to the ShoreTel phone can be transferred to an alternate destination.</td></tr>
<tr><td>SIP Trace?
Yes for Call #1</td><td>Trace 1 filename: TWTelecomm-4.6-Call Transfer Blind
Trace 2 filename:</td></tr>
<tr><td>Test steps</td><td>Call #1 – Originate Transfer from a ShoreTel phone
1. From SUT endpoint #1, call the ShoreTel 230 (REF-1) device and verify that the call is connected successfully.
2. From ShoreTel 230 (REF-1), transfer the call to SUT endpoint #2 device and hang up before answering the new incoming call.
3. After hanging up the ShoreTel 230 (REF-1) device, verify that the SUT endpoint #2 device is still ringing.
4. Answer the incoming call and verify that the SUT endpoint#1 and SUT endpoint #2 devices have been connected successfully.
Call #2 – Originate transfer from a SIP reference
5. From SUT endpoint #1, call the Reference phone device and verify that the call is connected successfully.
6. From Reference phone, transfer the call to SUT endpoint #2 device and hang up before answering the new incoming call.
7. After hanging up the Reference phone device, verify that the SUT endpoint #2 device is still ringing.
8. Answer the incoming call and verify that the SUT endpoint#1 and SUT endpoint #2 devices have been connected successfully.
Call #3 – Originate transfer from a SIP reference to “device B”
9. From SUT endpoint #1, call the ShoreTel 230 (REF-1) device and verify that the call is connected successfully.
10. From ShoreTel 230 (REF-1), transfer the call to “device B” and hang up before answering the new incoming call.
11. After hanging up the ShoreTel 230 (REF-1) device, verify that “device B” device is still ringing.
12. Answer the incoming call and verify that the SUT endpoint#1 and “device B” have been connected successfully.
Call #4 – Dial through AutoAttendant to extension
13. From SUT endpoint #1, call the ShoreTel AutoAttendant and enter the desired extension.
14. Verify transfer to correct destination.
15. Verify media in both directions.</td></tr>
<tr><td>Result</td><td>Call #1 Pass
Call #2 Pass
Call #3a – using ShoreTel IP 230 as “device B” Pass
Call #3b – using ShoreTel voicemail as “device B” Pass
Call #3c – using ShoreTel Workgroup as “device B” Pass
Call #4</td></tr>
<tr><td>Notes</td><td>See Figure 3: Transfer Scenarios Drawing (for Test Cases 4.6 and 4.7) (above)</td></tr>
</table>

ID	**4.7**
Name	**Call Transfer – Consultative**
Mandatory or Optional:	**Mandatory**
Description	**Verify a call connected from the SUT to the ShoreTel phone can be transferred to an alternate destination.**
SIP Trace? Yes for Call #1	Trace 1 filename: TWTelecomm-4.7-Call Transfer Consultative.pcap Trace 2 filename:
Test steps	**Call #1 – Originate Transfer from a ShoreTel phone** 1. From **SUT endpoint #1**, call the **ShoreTel 230 (REF-1)** device and verify that the call is connected successfully. 2. From **ShoreTel 230 (REF-1)**, transfer the call to the **SUT endpoint #2** device; do not hang up the **ShoreTel 230 (REF-1)** device. 3. From **SUT endpoint #2**, answer the incoming call and verify that only **ShoreTel 230 (REF-1)** and **SUT endpoint #2** are connected. 4. Complete the call transfer and verify that **SUT endpoint #1** and **SUT endpoint #2** are successfully connected. **Call #2 – Originate transfer from a SIP reference** 5. From **SUT endpoint #1**, call the **Reference phone** device and verify that the call is connected successfully. 6. From **Reference phone**, transfer the call to the **SUT endpoint #2** device; do not hang up the **Reference phone** device. 7. From **SUT endpoint #2**, answer the incoming call and verify that only **Reference phone** and **SUT endpoint #2** are connected. 8. Complete the call transfer and verify that **SUT endpoint #1** and **SUT endpoint #2** are successfully connected. **Call #3 – Originate transfer from a SIP reference to "device B"** 9. From **SUT endpoint #1**, call the **ShoreTel 230 (REF-1)** device and verify that the call is connected successfully. 10. From **ShoreTel 230 (REF-1)**, transfer the call to **"device B"**; do not hang up the **ShoreTel 230 (REF-1)** device. 11. From **"device B"**, answer the incoming call and verify that only **ShoreTel 230 (REF-1)** and **"device B"** are connected. 12. Complete the call transfer and verify that **SUT endpoint#1** and **"device B"** have been connected successfully.
Result	**Call #1 Pass** **Call #2 Pass** **Call #3a – using ShoreTel IP 210 as "device B" Pass** **Call #3b – using ShoreTel voicemail as "device B" Pass** **Call #3c – using ShoreTel Workgroup as "device B" Pass**
Notes	See Figure 3: Transfer Scenarios Drawing (for Test Cases 4.6 and 4.7) (above)

Figure 3: Conference Scenario Drawing (for Test Case 4.8)



ID	**4.8**
Name	**Conference – ad hoc**
Mandatory or Optional:	**Mandatory**
Description	**Verify successful ad hoc conference of three parties**
SIP Trace? Yes for Call 1a and 1c	Trace 1 filename: TWTelecomm-4.8-Conference-ad hoc.pcap Trace 2 filename:
Test steps	**Call #1 – Three party conference call** 1. From **ShoreTel IP560 (REF-2)**, call **SUT Endpoint #1** and verify that the call is connected successfully. 2. From **ShoreTel IP560 (REF-2)**, place **SUT Endpoint #1** on hold and call **"device B"**. 3. Complete the conference and verify the audio path between all 3 devices. **Call #2 -** repeat Call #1 using the SIP reference phone instead of the ShoreTel phone.
Result	**Call #1a – using ShoreTel IP 230 as "device B"** **Pass** **Call #1b – using ShoreTel IP 560 as "device B"** **Pass** **Call #1c – using External PSTN Numb4er as "device B"** **Pass** **Call #2a – using ShoreTel IP230 as "device B"** **Pass** **Call #2b – using ShoreTel IP560 as "device B"** **Pass** **Call #2c – using External PSTN Number as "device B"** **Pass**
Notes	See Figure 3: Conference Scenario Drawing (for Test Case 4.8) (above). You may need to enable "SIP Media Proxy" on the SIP Trunk ShoreGear switch.

ID	**4.9**
Name	**Inbound DID/DNIS**
Mandatory or Optional:	**Mandatory**
Description	**Verify the SUT provides inbound "dialed number information" and is correctly routed to the configured destination.**
SIP Trace? Yes	Trace 1 filename: Trace 2 filename:
Test steps	1. Dial from PSTN through the trunk. 2. Verify correct CallerID is displayed.
Result	Pass
Notes	

ID	**4.10**
Name	**Outbound 911**
Mandatory or Optional:	**Mandatory**
Description	**Verify that outbound calls to 911 are routed to the correct PSAP for the calling location and that caller ID information is delivered.**
SIP Trace? Yes	Trace 1 filename: TWTelecomm-4.10-Outbound 911.pcap Trace 2 filename:
Test steps	1. Check routing for 911 from SUT and ensure that it points to a controlled destination, not live 911 services. 2. Place call from **ShoreTel phone** to **911**. 3. Ensure that call routes properly to controlled destination.
Result	Pass
Notes	**Based on service provider's support of emergency calls.**

ID	**4.11**
Name	**Operator Assisted**
Mandatory or Optional:	**Mandatory**
Description	**Verify that 0+ calls are routed to an operator for calling assistance.**
SIP Trace? Yes	Trace 1 filename: Trace 2 filename:
Test steps	1. Place call from **ShoreTel phone** to **0+**. 2. Ensure that call routes properly to controlled 0+ destinations.
Result	Pass
Notes	**Based on service provider's support of operator services.**

ID	**4.12**
Name	**Inbound / Outbound call with Blocked Caller ID**
Mandatory or Optional:	**Mandatory**
Description	**Verify that calls with Blocked Caller ID route properly and the answering phone does not display any Caller ID information.**
SIP Trace? Yes	Trace 1 Filename: Trace 2 Filename: TWTelecomm-4.12-Outbound call with Blocked Caller ID.pcap
Test steps	1. Place an inbound call to the SUT and block the Caller ID on the initiating caller 2. Verify that the call routes properly and the answering Reference phone displays “Unknown” for Caller ID. 3. Place an outbound call, via SUT, to an external device (i.e. mobile phone) and block Caller ID. 4. Verify that the call routes properly and the destination phone rings and doesn’t display any Caller ID
Result	Pass
Notes	

ID	**4.13**
Name	**Inbound call to a Hunt Group**
Mandatory or Optional:	**Mandatory**
Description	**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.**
SIP Trace? Yes	Trace 1 Filename: TWTelecomm-4.13-Inbound call to a Hunt GroupC1.pcap Trace 2 Filename: TWTelecomm-4.13-Inbound call to a Hunt GroupC2.pcap
Test steps	1. Configure the ShoreTel system to route incoming calls to a Hunt Group, this can be the default Trunk Group destination or via DNIS / DID. Place an inbound call. 2. Verify that the call routes to the proper Hunt Group and is answered by an available agent and two way communication is available. 3. Blind transfer call to another extension and verify that the call rings the other extension and is answered successfully with two way communication 4. Configure the No Answer / Call Stack Full destination for the Hunt Group (in step 1) to be another Hunt Group extension, 5. Place an inbound call and do not answer the call and allow the call to route to the second Hunt Group. Verify that that the call first routes to the first Hunt Group and then to the second Hunt Group, ringing all available agents. Answer the call from an available member in the second hunt group. 6. Perform a Consultive Transfer to another extension, verify that the destination extension rings and the call is successfully transferred with two way communication.
Result	G.729 – N/S G.711 - Pass
Notes	

ID	**4.14**
Name	**Inbound call to a Workgroup**
Mandatory or Optional:	**Mandatory**
Description	**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.**
SIP Trace? **Yes**	Trace 1 Filename: TWTelecomm-4.14-Inbound call to a WorkgroupC1.pcap Trace 2 Filename: TWTelecomm-4.14-Inbound call to a WorkgroupC2.pcap
Test steps	1. Configure the ShoreTel system to route incoming calls to a Workgroup, this can be the Trunk Group default destination or via DNIS / DID. Place an inbound call. 2. Verify that the call routes to the proper Workgroup and rings all available workgroup agents. Answer the call and verify two way communications. 3. Blind transfer the call to another extension, verify that the call rings and is successfully answered with two way communication. 4. Configure the No Answer destination for the Workgroup (in step 1) to be another Workgroup. 5. Place an inbound call, verify that the call routes to the first Workgroup, rings all available agents and is then routed to the second Workgroup. Answer the call from an available workgroup agent and verify two way communication. 6. Perform a Consultive Transfer to another extension, verify that the extension rings and the call is successfully transferred with two way communication.
Result	G.729 – N/S G.711 – Pass
Notes	

ID	**4.15**
Name	**Inbound call to DID / DNIS and leave a voice mail message**
Mandatory or Optional?	**Mandatory**
Description	**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.**
SIP Trace? **No**	Trace 1 Filename: Trace 2 Filename:
Test steps	1. Place an inbound call to a user extension, via DID / DNIS, that has a mailbox. 2. Don't answer the call and allow it to roll to voice mail after three rings. 3. Leave a message for the user. 4. Verify that the user's phone immediately illuminates the Message Waiting Indicator, then log into the user's mailbox and listen to the message, verify that the audio is acceptable, both in level and quality.
Result	Pass
Notes	

ID	**4.16**
Name	**Call Forward – “FindMe”**
Mandatory or Optional?	**Mandatory**
Description	**Verify that inbound calls are forwarded to a user’s “FindMe” destination.**
SIP Trace? **Yes**	Trace 1 Filename: Trace 2 Filename:
Test steps	1. Configure a ShoreTel extension with an External PSTN “FindMe” destination. 2. Place an inbound call to the user extension defined in step 1, don’t answer the call and allow it to go to voice mail. 3. Once you hear the user’s greeting, press the digit 1, this should place an outbound call to the desired external number. 4. Verify that the external destination rings, answer the call. 5. Verify that you are prompted to press 1 to accept the call and 2 to decline, press DTMF 1 to accept the call. 6. Verify that the there’s two way communication. 7. Hangup the call from the originating party and verify that the destination hangs up.
Result	Pass
Notes	

ID	**4.17**
Name	**Call Forward Always**
Mandatory or Optional?	**Mandatory**
Description	**Verify that inbound calls are immediately automatically forwarded to a user’s external destination.**
SIP Trace? **Yes**	Trace 1 Filename: TWTelecomm-4.17-Call Forward Always Trace 2 Filename:
Test steps	1. Configure a ShoreTel user extension’s call handling mode to always forward to an external number. 2. Place an inbound call to the user configured in step 1. 3. Verify that the call is immediately forwarded out the SUT and the external destination rings. 4. Answer the call and verify two way audio. 5. Disconnect the call from called party and verify that the calling party is disconnected.
Result	Pass
Notes	

ID	**4.18**
Name	**Inbound / Outbound Fax calls**
Mandatory or Optional?	**Mandatory**
Description	**Verify that inbound / outbound fax calls complete successfully.**
SIP Trace? No	Trace 1 Filename: Trace 2 Filename:
Test steps	1. Configure a ShoreTel extension for analog and connect a fax machine to the ShoreGear switch port. 2. Place an inbound call directed to the fax extension defined in step 1. 3. Send a 5 page fax and verify successful transmission of the fax. 4. Place an outbound fax call from the fax extension defined in step 1. 5. Send a 10 page fax and verify successful transmission of the fax.
Result	G.711 – OK T.38 -- N/S
Notes	Test for both T.38 and G.711

ID	**4.19**
Name	**ShoreTel Service Appliance Unified Communication System**
Mandatory or Optional?	**Optional**
Description	**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.**
SIP Trace? **No**	Trace 1 Filename: Trace 2 Filename:
Test steps	1. Configure the ShoreTel Service Appliance for a reservation-less conference and note the participant access code. 2. Place an inbound call to SUT and direct the call to the Service Appliance, once prompted to enter your access code, be sure to input the proper participant access code. 3. Verify that the Service Appliance properly detects the DTMF digits and allows you access into the conference call. From another ShoreTel extension dial the Service Appliance and verify that you can communicate properly with the external party. 4. Using the host controls, initiate an outbound call to an external number. 5. Verify that the call is properly placed and the phone rings. Answer the call and verify audio path for all parties involved.
Result	N/T, additional hardware required
Notes	The ShoreTel Service Appliance supports several codecs, but not the G.729 codec, test all codecs supported by the ITSP.

ID	**4.20**
Name	**Inbound call to Bridged Call Appearance (BCA) extension**
Mandatory or Optional?	**Mandatory**
Description	**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.**
SIP Trace? **No**	Trace 1 Filename: Trace 2 Filename:
Test steps	2. Configure the ShoreTel system with a Bridged Call Appearance (BCA), and defined a configurable button on two of the ShoreTel IP Phones (i.e. IP212K/230/560 or BB24). 3. Configure the Trunk Group destination as the BCA extension, or define the BCA extension with a DNIS map or assign it a DID. 4. Place an inbound call 5. Verify that the call is directed to the BCA extension and all of the phones with the programmable BCA button are showing the call. 6. Answer the call and verify two way communications. 7. Place the call on-hold 8. Take call off hold from one of the other extensions that are monitoring the BCA extension, verify that there's two way communication. 9. Blind transfer the call to another ShoreTel extension. Verify that the destination phone is ringing and the call is properly answered with two way communication.
Result	Pass
Notes	

ID	**4.21**
Name	**Inbound call to a Group Pickup extension**
Mandatory or Optional?	**Mandatory**
Description	**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.**
SIP Trace? **No**	Trace 1 Filename: Trace 2 Filename:
Test steps	1. Configure the ShoreTel system with a Pickup Group extension and defined a programmable button on two of the ShoreTel IP Phones (i.e. IP212K/230/560 or BB24). 2. Configure the Trunk Group destination as the one of the extensions defined in the Extension List for the Pickup Group extension or assign a DNIS map or DID to the desired extension. 3. Place an inbound call 4. Verify that the call is directed to the desired user extension. 5. From one of the phones that has a programmable button configured for the Pickup Group extension, answer the call and verify two way communication. 6. Place the call on-hold 7. Take call off hold , verify that there's two way communication. 8. Blind transfer the call to another ShoreTel extension. Verify that the destination phone is ringing and the call is properly answered with two way communication.
Result	Pass
Notes	

ID	**4.22**
Name	**Office Anywhere External**
Mandatory or Optional?	**Mandatory**
Description	**Verify that inbound calls are properly presented to the Office Anywhere External PSTN destination.**
SIP Trace? Yes	Trace 1 Filename: TWTelecomm-4.22-Office Anywhere External.pcap Trace 2 Filename:
Test steps	1. Configure a ShoreTel extension with a DID from the SUT. 2. Configure the ShoreTel extension to use External Assignment and define a PSTN number, and set the activation to "accept call by answering". 3. Place an inbound call to the DID defined in step 1. 4. Verify that the phone configured for External Assignment rings and that the calling party hears ringback tone. 5. Answer the call, verify two way audio path. 6. Hang up at the called party side and verify that the calling party hangs up. 7. Configure the extension's External Assignment activation to "accept call by pressing '1'". 8. Repeat steps 3 and 4. 9. Answer the call and accept by pressing DTMF '1'. 10. Verify two way audio communication. 11. Hang-up up the call at the calling party side and verify that the called party hangs up.
Result	Pass
Notes	

ID	**4.23**
Name	**Simul Ring**
Mandatory or Optional?	**Mandatory**
Description	**Verify that inbound calls are properly presented to the desired extension and the "Additional Phones" destinations.**
SIP Trace? Yes	Trace 1 Filename: TWTelecomm-4.23-Simul Ring.pcap Trace 2 Filename:
Test steps	1. Configure a ShoreTel extension with a DID from the SUT. 2. Enable "SIP Media Proxy" resources on the SIP Trunk ShoreGear Switch. 3. Configure the ShoreTel extension to use the "Additional Phones" feature and define two separate external PSTN numbers. 4. Place an inbound call to the DID defined in step 1. 5. Verify that the destination extension rings, and that the "Additional Phones" destinations also ring. 6. Answer one of the configured "Additional Phones" PSTN numbers, verify two-way audio path. 7. Go to the main ShoreTel extension and select the "Move" SoftKey, verify that the call is transferred to the ShoreTel IP Phone and that there's two-way audio path. 8. Hang-up the call from the calling party side, verify that the ShoreTel IP Phone goes on-hook.
Result	Pass
Notes	

ID	**4.24**
Name	**MakeMe Conference**
Mandatory or Optional?	**Mandatory**
Description	**Verify that an inbound call can be conferenced with three (or more) additional parties**
SIP Trace? **No**	Trace 1 Filename: Trace 2 Filename:
Test steps	1. Configure four (or more) "MakeMe" conference ports on a ShoreGear switch. 2. Place an inbound call to a DID for a ShoreTel IP Phone. 3. Press the "Conference" button and dial an external PSTN number, after dialing a valid number, press the "Conference" SoftKey. 4. Verify that the external PSTN number rings, answer the call and verify that all parties get audio. 5. Using the original called party ShoreTel IP Phone, once again press the "Conference" button and dial an internal ShoreTel extension, after dialing a valid extension, press the "Conference" SoftKey. 6. Verify that all parties get audio in all directions. 7. Add additional parties (both internal and external) until six parties is reached (if you have configured sufficient "MakeMe" conference port resources), verify audio with all parties in all directions.
Result	Pass
Notes	

ID	**4.25**
Name	**Park / Unpark**
Mandatory or Optional?	**Mandatory**
Description	**Verify that an inbound call can be parked and unparked**
SIP Trace? **No**	Trace 1 Filename: Trace 2 Filename:
Test steps	1. Place an inbound call to a DID for a ShoreTel IP Phone. 2. Answer the call and verify two-way audio path. 3. Press the "Park" SoftKey and enter another valid user extension, the press the "Park" SoftKey again. 4. Go to the ShoreTel IP Phone extension where you parked the call and pickup the handset. Verify that the call is active and you have two-way audio. 5. Press the "Park" SoftKey and enter the original called extension, then press the "Park" SoftKey again. 6. The call will now be parked at the original extension, from another ShoreTel IP Phone, press the "Unpark" SoftKey and enter the extension where the call is parked, then press the "Unpark" SoftKey again, then select the extension where the call is parked. 7. Verify two-way audio. 8. Hangup the call from the calling party, verify that the ShoreTel extension goes on-hook.
Result	Pass
Notes	

ID	**4.26**
Name	**Call Recording**
Mandatory or Optional?	**Mandatory**
Description	**Verify that external calls can be recorded via the SIP Trunk using ShoreTel Communicator**
SIP Trace? **Yes**	Trace 1 Filename: Trace 2 Filename:
Test steps	1. Configure a ShoreTel extension to allow recording of own calls and associate with ShoreTel Communicator. 2. Enable "SIP Media Proxy" resources on the SIP Trunk ShoreGear switch. 3. Using Communicator place an outbound call to an external PSTN number. 4. Verify that you hear ringback and that the destination phone rings. 5. Answer the call and verify two-way audio path. 6. Using Communicator record the call, verify that you don't get any errors. 7. Hangup the call and verify that you receive a new voice mail message (which is the call recording). 8. Listen to the call recording and verify that you have all audio and there are no drop outs or missing audio. 9. Place an inbound call to the extension and repeats steps 4 through 8.
Result	Pass
Notes	Retested by Craig and Santosh on 6/5/14 and passed inbound & outbound call recording

ID	**4.27**
Name	**Silent Monitor / Barge-In / Whisper Page**
Mandatory or Optional?	**Mandatory**
Description	**Verify that external calls can be silently monitored, barged-in and whisper paged via the SUT.**
SIP Trace? **No**	Trace 1 Filename: Trace 2 Filename:
Test steps	1. Configure a ShoreTel extension to allow Silent Monitor / Barge-In and Whisper Page of other extensions. 2. Enable "SIP Media Proxy" resources on the SIP Trunk ShoreGear switch. 3. Place an external PSTN call from an extension that can be monitored, answer the called party destination and verify two-way audio path. 4. Using the extension configured in step 1 initiate a Silent Monitor, verify that you can hear both sides of the conversation and that the two parties cannot hear you. Hang-up the Silent Monitor call. 5. Using the extension configured in step 1 initiate a Barge-in, verify that you can hear both parties and that they can hear you. Hang-up the Barge-In call. 6. Using the extension configured in step 1 initiate a Whisper Page, verify that only the ShoreTel extension can hear you. Hang-up the Whisper Page call. 7. Disconnect the call. 8. Place an inbound call to the extension that can be monitored and verify two-way audio path. Perform steps 4 through 7.
Result	Pass
Notes	

ID	**4.28**
Name	**Long Duration - Inbound**
Mandatory or Optional?	**Mandatory**
Description	**Verify that an inbound call is established for a minimum of 30 minutes.**
SIP Trace? **No**	Trace 1 Filename: Trace 2 Filename:
Test steps	1. Place an inbound call to a ShoreTel IP Phone via the SUT. 2. Verify two-way communication. 3. Leave the call active for a minimum of 30 minutes, verify that there's two-way communication for the entire duration of the call. 4. Disconnect the call from the Calling Party side. 5. Verify that the Called Party is also disconnected.
Result	Pass
Notes	

ID	**4.29**
Name	**Long Duration - Outbound**
Mandatory or Optional?	**Mandatory**
Description	**Verify that an outbound call is established for a minimum of 30 minutes.**
SIP Trace? **No**	Trace 1 Filename: Trace 2 Filename:
Test steps	1. Place an outbound external PSTN call via the ShoreTel Reference phone. 2. Verify two-way communication. 3. Leave the call active for a minimum of 30 minutes, verify that there's two-way communication for the entire duration of the call. 4. Disconnect the call from the Called Party side. 5. Verify that the Calling Party is also disconnected.
Result	Pass
Notes	

ID	**4.30**
Name	**Contact Center**
Mandatory or Optional?	**Optional**
Description	**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.**
SIP Trace? **Yes**	Trace 1 Filename: Trace 2 Filename:
Test steps	1. Configure the Route Point (that is defined as the ECC IRN) with a DID from the SUT. 2. Configure the ECC IRN to play announcements (mandatory & first) and to queue calls when no agents are available. 3. Log in a single ECC Agent and place in wrap-up. 4. Place an inbound call to the DID configured in step 1. 5. Verify that you hear the entire mandatory and first announcements. 6. Verify that call goes into queue. 7. Take the ECC Agent out of wrap-up. 8. Answer the call using the Agent Tool Bar and verify two-way communication. 9. Place the call on-hold for 60 seconds, resume the call and verify two-way communication. 10. Blind transfer the call to another ShoreTel extension. 11. Verify that the destination phone rings and the calling party hears ringback tone. Upon answering verify two-way communication. 12. Disconnect the call from the Called Party side and verify that the Calling Party also disconnects.
Result	N/T. Agent Tool Bar not available.
Notes	

ID	**4.31**
Name	**ShoreTel Mobility Router (SMR)**
Mandatory or Optional?	**Optional**
Description	**Verify that the SMR can be used with the SUT**
SIP Trace? **Yes**	Trace 1 Filename: Trace 2 Filename:
Test steps	1. You will need four DIDs from the SUT for SMR, one for the extension, one for access, one for handover and one for reverse dial. The actual tests are TBD.
Result	N/T.
Notes	Test not defined

8. Security Tests

Perform all test cases in this section, logging Results and Notes as appropriate.

8.1. Security Test Setup Diagram

The security SIP ITSP ITP test setup is the same as the basic test setup shown in Figure 1: Basic ShoreTel SIP ITSP Test Setup

8.2. Security Test Cases

ID	**5.1**
Name	**Registration or Digest Authentication**
Mandatory or Optional:	**Optional**
Description	**Verify the SUT supports the use of registration or digest authentication for service access for inbound and outbound calls.**
SIP Trace? Yes	Trace 1 filename: Trace 2 filename:
Test steps	1. Configure ShoreTel system as needed with or without registration or digest authentication. 2. Make an inbound call and verify that it completes properly. 3. Make an outbound call and verify that it completed properly.
Result	N/T. Registration not configured
Notes	**Based on service provider's support of registration or digest authentication.**

End of Test Plan