IVT Test Plan and Report
for
Tesira SVC-2 (SIP) Endpoint
with
ShoreTel 13.1

Innovation Network

ShoreTel®
Biamp Tesira SVC-2

ShoreTel Platform Information:

	[Component #1]
Machine Type:	Shoregear 120/24
OS Type and SP Level:	Windows 2008 R2 SP 1
Software & Version	ShoreWare 13.1 Build 18.23.2412.0
RAM	4 GB

[Tesira SVC-2] Vendor Information:

Vendor company name:	**Biamp Systems**
Vendor rep present (if any)**:**	Name: Jing Jiang Title: Engineer Phone: 5036417287 Email: jjiang@biamp.com
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Description of Product Tested

The Tesira SVC-2 is a modular Voice over Internet Protocol (VoIP) card for use with Tesira SERVER and SERVER-IO devices. The SVC-2 allows a Tesira system to connect directly to IP-based telephone systems. When used in conjunction with Automatic Echo Cancellation processing, Tesira becomes an extraordinarily powerful, flexible and affordable conferencing platform.

VoIP Endpoint product information:

Product name:	Tesira SVC-2
Product type:	SIP endpoint
Product model no.:	Tesira SVC-2
Product serial no:	TesiraServer 91101307
Product release:	1.1.2.6
Software release (if any)**:**	None
Options installed (if any)**:**	None

Executive Summary:

The following summarizes tekVizion's findings:

- Test Cases Failures:
  - DUT cannot access ShoreTel Voice mail (VM) and features related to VM. DUT and ShoreTel VM are configured to work with Out-of-Band DTMF. When the DUT calls the ShoreTel VM. Call is answered by VM, but DUT is replied with DTMF not Supported. However, DTMF works fine between DUT and ShoreTel phones. The issue has been reported to Shoretel and was identified to be an issue with ShoreTel version 13.1 build 18.23.2412.0 which was subsequently resolved in ShoreTel 13.2 build 18.42.1100.0.

- Features Not Supported:
  - Date and time update support
  - Forward from SIP DUT
  - Missed call notification
  - Callback
  - Headset
  - Ring selection
  - Fax handling
  - Mid-call codec negotiation.

- Test cases Not Tested:
  - Calls to and from SIT ATA Mediatrix 2102.
  - Speech quality – Minimal impairment.

- Observations:
  - The DUT can only have 2 other phones in a conference, while DUT is initiating the call.
  - DUT cannot initiate a call transfer; refer to test case (TC) 3.5 and 3.6.
  - When a DUT calls a hunt group, the Call cannot not be transferred to second hunt group, refer to steps 5-6 of TC 3.18.
  - When a DUT calls a work group, the Call cannot not be transferred to second work group, refer to steps 5-6 of TC 3.19.

1. Test Cases Overview

This section presents an overview of all the VoIP Endpoint test cases available in this test plan template. The Notes field should be used to indicate if the test is required or optional, and any other explanation that might be pertinent to this test case.

Table 1-1: Basic Feature Test Cases

ID	Optional?	Name	Description	Notes
1.1	Mandatory	Device initialization with static IP address	Verify successful startup and initialization of the device up to a READY/IDLE state using a static IP address	Pass
1.2	Mandatory	Device reset – idle (for static configurations)	Verify successful re-initialization of device after power loss while device is idle	Pass
1.3	Mandatory	Device initialization with DHCP	Verify successful startup and initialization of the device up to a READY/IDLE state using DHCP	Pass
1.4	Mandatory	Device reset – idle (for dynamic configurations)	Verify successful re-initialization of device after power loss while device is idle	Pass
1.5	Mandatory	Verify Diffserv Code Point support	Verify the ability to set Diffserv Code Point from SIP DUT	Pass
1.6	Optional	Verify Date and Time Update support	Verify setting of Date and Time Update on SIP DUT	N/S
1.7	Mandatory	Place call	Verify successful call placement with normal dialing to a variety of terminating phones	Pass
1.8	Mandatory	Receive call	Verify successful reception of calls with normal dialing from a variety of calling phones	Pass
1.9	Optional	Place call – re-dial	Verify successful call placement using re-dial to SIP Reference	Pass
1.10	Optional	Place call – speed dial	Verify successful call placement using programmed speed dial	Pass
1.11	Mandatory for G.711, Optional for other CODECs	CODEC support – common (from DUT to ShoreTel Phone, REF-x)	Verify successful call connection and audio path using all supported CODECs (G.711-Ulaw and G.729)	Pass

ID	Optional?	Name	Description	Notes
1.12	Mandatory for G.711, Optional for other CODECs	CODEC support – common (from DUT to SIP Reference Phone, SIP-Ref)	Verify successful call connection and audio path using all supported CODECs (G.711-Ulaw and G.729)	Pass
1.13	Mandatory (only if more than 1 CODEC is supported)	CODEC support – negotiated	Verify successful negotiation between devices configured with different default CODECs (G.711-Ulaw and G.729)	Pass
1.14	Mandatory	Hold from DUT to SIP Reference	Verify successful hold and resume of connected call	Pass
1.15	Mandatory	Hold from DUT to ShoreTel Phone	Verify successful hold and resume of connected call	Pass
1.16	Mandatory	Forward	Verify successful forwarding of incoming calls	Pass
1.17	TBD	Forward from SIP DUT	Verify successful forwarding of incoming calls	N/S
1.18	Optional	Mute	Verify device's mute function	Pass
1.19	Mandatory	Out-of-band / In-band DTMF Transmission	Verify successful transmission of in-band and out-of-band digits (RFC2833) for calls placed to and from the DUT with a variety of other devices	Pass
1.20	Optional	Missed call notification	Verify that device notifies the user about missed calls	N/S
1.21	Optional	Volume	Verify the device's volume adjustment function	Pass

Table 1-2: Performance Test Cases

ID		Name	Description	Notes
2.1	Mandatory	Speech quality – Minimal impairment	Verify acceptable voice quality between two parties with minimal network impairment condition	N/T
2.2	Mandatory	Speech quality – Moderate Impairment	Verify acceptable voice quality between two parties with low-to-moderate artificial network impairment condition	N/T
2.3	Mandatory	Speech quality – High Impairment	Verify acceptable voice quality between two parties with moderate-to-high artificial network impairment condition	N/T

Table 1-3: Extended Feature Test Cases

ID		Name	Description	Notes
3.1	Mandatory	Call waiting	Verify appropriate notification and successful connection of incoming call while busy with another party	Pass
3.2	TBD	Park	Verify successful park and retrieval of connected call	Pass
3.3	Optional	Extended forward	Verify extended call forwarding options – busy forwarding, no-answer forwarding	Pass
3.4	Optional	Extended forward from SIP DUT	Verify extended call forwarding options – busy forwarding, no-answer forwarding	Pass
3.5	Mandatory	Transfer – blind	Verify successful blind transfer of connected call	Cannot be transferor
3.6	Mandatory	Transfer – monitored	Verify successful monitored transfer of connected call	Cannot be transferor
3.7	Mandatory	Conference – ad hoc	Verify successful ad hoc conference of three parties	Pass DUT can only have 2 other phones, while initiating the call.
3.8	Optional	Place call – secondary line	Verify successful call placement using secondary line	Pass
3.9	Optional	Receive call – secondary line	Verify successful connection of incoming call on secondary line	Pass
3.10	Optional	Callback	Verify successful connection of a call using the missed-call callback feature of the device	N/S
3.11	Optional	Headset	Verify the device's support for external headsets (using headsets supplied by the 3P phone vendor)	N/S
3.12	Optional	Ring selection	Verify the device's ability to change the ring type	N/S
3.13	Mandatory	Caller ID Name and Number	Verify that Caller ID name and number is sent and received from SIP endpoint device	Pass
3.14	Optional	SIP Device Generates Busy Tone	Verify that SIP DUT generates busy tone when calling a busy extension	Pass
3.15	TBD	POTS Analog Gateway supports the transfer operation by “flashing”	Verify that the POTS Analog Gateway can support the transfer operation by “flashing”	N/A
3.16	Mandatory	Verify handling of “911”	Verify dialing “911” on DUT could connect with “911” services	N/A

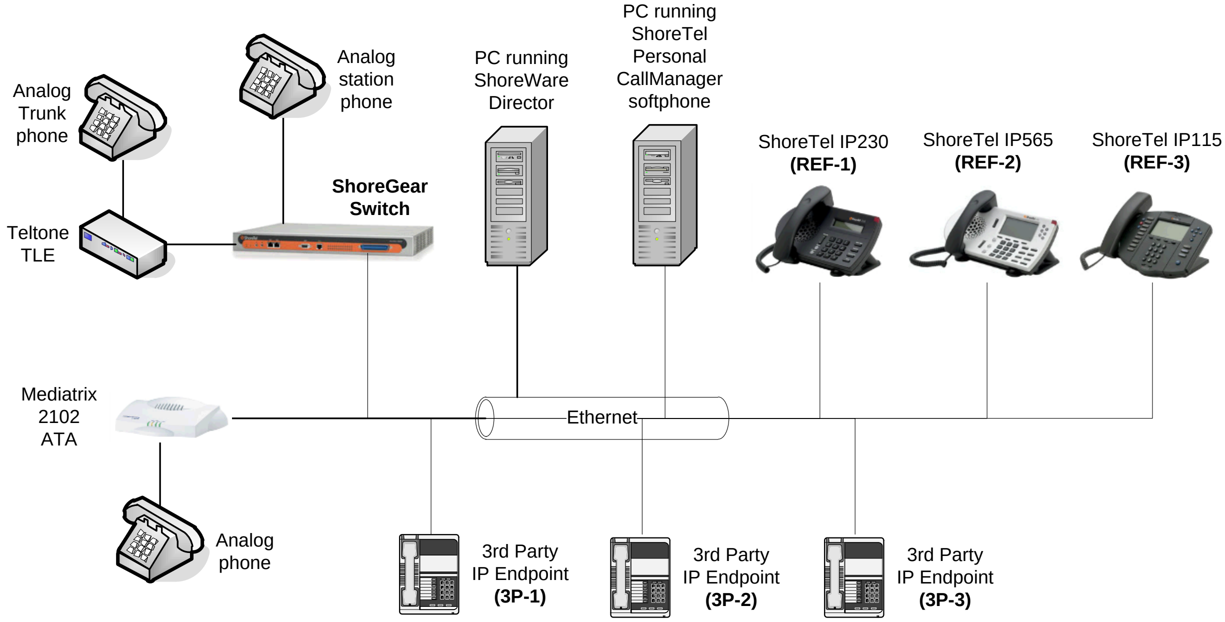
ID		Name	Description	Notes
3.17	Mandatory	Verify Fax Handling	Verify that fax can be sent and received through DUT	N/S
3.18	Mandatory	Call to Hunt Group	Verify that calls route to the proper Hunt Group and are answered by an available hunt group member with audio in both directions using G.729 and G.711 codecs.	Limitation: DUT cannot initialize a transfer. DUT cannot get into the second hunt group
3.19	Mandatory	Call to Work Group	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.	DUT cannot initialize a transfer. DUT cannot get into the second work group.
3.20	Mandatory	Hunt Group Member	Verify that the DUT can receive calls while configured to be a hunt group member.	Pass
3.21	Mandatory	Workgroup Agent	Verify that DUT can receive calls while configured to be a workgroup agent.	Pass
3.22	Mandatory	Call Forward – FindMe	Verify that callers are forwarded to the “FindMe” destination.	Pass
3.23	Mandatory	Communicator	Verify that a call can be initiated from ShoreTel Communicator, that it rings the DUT, and once answered, a call is placed to the desired destination	Pass
3.24	Mandatory	Simultaneous Ring	Verify that an inbound call to a user’s primary ShoreTel IP Phone also rings the DUT.	Pass

2. Test Setup Requirements and Definition of Terms

2.1. *Test Setup Diagram*

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

Figure 1: Basic ShoreTel VoIP Endpoint ITP Test Setup



Number of 3rd Party VoIP Endpoint devices required for this test plan = **3**

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 CT Labs or made available remotely.
Reference VoIP endpoint device	Permanent test lab phone chosen by the IP PBX or call center equipment manufacturer 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 VoIP endpoint device	Vendor product being tested for interoperability with the host VoIP platform. This can be an IP desk phone, soft phone, wireless IP phone, or like device.
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.
DUT	Device Under Test, A.K.A vendor VoIP Endpoint Device
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

CT Labs will verify the Host VoIP Platform environment's readiness to begin testing the DUT by performing a subset of the test plan with a Reference VoIP Endpoint device. Once this set of tests has been completed, CT Labs will then add the DUT to the test environment and perform the same steps again with it.

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 DUT,
2. Verify that the DUT is able to perform the most basic functions in the host VoIP Platform, and
3. Raise a warning flag early in the testing process if a DUT has fundamental problems operating in the Host VoIP Platform environment.

3.2. Quick Test Procedure:

1. Log the following information:

3rd Party Device being tested:	03/22/2013
3rd Party Device version:	1.1.2.6
ShoreTel version:	Version 13.1 , Build:18.23.2412.0

2. Perform the test cases (see section 4) listed below with **lab Reference VoIP Endpoint device**, logging results into “Reference device” field in Table 3-1 below.

3. Perform the test cases (see section 4) listed below with **vendor VoIP Endpoint device** under test (DUT), logging results into “Vendor device (DUT)” field in Table 3-1 below.

Table 3-1: Quick Test Results (PASSED OR FAILED)

Test case:	Reference device:	Vendor device (DUT):
1.1	Pass	Pass
1.3	Pass	Pass
1.7	Pass	Fail
1.8	Pass	Pass
1.11	Pass	Pass
1.15	Pass	Pass
1.19	Pass	Fail

4. 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	**CT Labs will use the configuration application note provided by the vendor to configure the vendor's product to work with the ShoreTel system.**
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	Application Note shows how to configure ShoreTel 13.1 system to work with Biamp Tesira SVC-2 card

ID	**1.1**
Name	**Device initialization with static IP address**
Mandatory or Optional:	**Mandatory**
Description	**Verify successful startup and initialization of the device up to a READY/IDLE state using a static IP address**
Test steps	1. Configure phone as static IP device; Configure ShoreTel Individual Trunk as static 2. Start **3P-1** and perform any local configuration necessary to register with the appropriate server(s). 3. Verify that **3P-1** initializes properly and is ready to place and receive calls. Configured DN and/or device state should be displayed.
Result	**Pass**
Notes	Status display is not showed if the device is registered correctly. However if the device is not registered, the display will show the error.

ID	**1.2**
Name	**Device reset – idle (for static configurations)**
Mandatory or Optional:	**Mandatory**
Description	**Verify successful re-initialization of device after power loss while device is idle**
Test steps	1. Use the static setup from Test Step 1.1 2. With **3P-1** in a READY/IDLE state, remove power from the device. 3. Wait several seconds, then re-apply power to the **3P-1** device. 4. Verify that the device re-initializes successfully and returns to a READY/IDLE state. 5. Place call from **REF-1** to **3P-1**. 6. Place call from **3P-1** to **REF-1**.
Result	**Pass**
Notes	

ID	**1.3**
Name	**Device initialization with DHCP**
Mandatory or Optional:	**Mandatory**
Description	**Verify successful startup and initialization of the device up to a READY/IDLE state using DHCP for the Ethernet Interface to the IP endpoint, NOT SIP Proxy/registration.**
Test steps	1. With power removed from **3P-1**, perform all necessary configuration changes on server/switch to prepare the device for normal operation with DHCP. 2. Start **3P-1** and perform any local configuration necessary to register with the appropriate server(s). 3. Verify that **3P-1** initializes properly and is ready to place and receive calls. Configured DN and/or device state should be displayed.
Result	**Pass**
Notes	

ID	**1.4**
Name	**Device reset – idle (for dynamic configurations)**
Mandatory or Optional:	**Mandatory**
Description	**Verify successful re-initialization of device after power loss while device is idle**
Test steps	1. Use the dynamic setup from test step 1.3 2. With **3P-1** in a READY/IDLE state, remove power from the device. 3. Wait several seconds, then re-apply power to the **3P-1** device. 4. Verify that the device re-initializes successfully and returns to a READY/IDLE state. 5. Place call from **REF-1** to **3P-1**. 6. Place call from **3P-1** to **REF-1**.
Result	**Pass**
Notes	

ID	**1.5**
Name	**Verify Diffserv Code Point support**
Mandatory or Optional:	**Mandatory**
Description	**Verify the ability to set Diffserv Code Point from SIP DUT**
Test steps	1. Change the Diffserv value on the **DUT** (can set to either 64 or 184) (refer to vendor documentation for instructions). 2. Verify via packet capture that the setting has been made. (check the Differentiated Services Field, and verify that the binary # translates correctly. Note 64 = 1000000 and 184 = 10111000)
Result	**Pass**
Notes	

ID	**1.6**
Name	**Verify Date and Time Update support**
Mandatory or Optional:	**Optional**
Description	**Verify setting of Date and Time Update on SIP DUT**
Test steps	1. Configure **DUT** to point to the Time server over the Internet 2. Verify the Time and Date on the **DUT** has changed to match the time of the Time server on the ShoreWare Director.
Result	**Not Supported**
Notes	This info is shown only on an incoming call as part of caller info.

ID	**1.7**
Name	**Place call**
Mandatory or Optional:	**Mandatory**
Description	**Verify successful call placement with normal dialing to a variety of terminating phones**
Test steps	Place a call with normal dialing to each device shown in Table 4-1, column 1. 1. From originating device, call the terminating device. 2. Verify that ringback is heard. 3. Answer incoming call from **3P-1**. 4. Verify that the call is connected successfully, and that media is established.
Result	Results logged into Table 4-1
Notes	When the DUT calls the ShoreTel Voice mail (VM). ShoreTel VM answers the call, but the ShoreTel replies to DUT as DTMF is Not Supported. Note: Since VM is using Out-of -band DTMF, DUT is also configured to use Out-of-Band DTMF.

ID	**1.8**
Name	**Receive call**
Mandatory or Optional:	**Mandatory**
Description	**Verify successful reception of calls with normal dialing from a variety of calling phones**
Test steps	Receive a call with normal dialing from each device shown in Table 4-1, column 2. 1. From originating device, call the terminating device. 2. Verify that ringback is heard. 3. Answer incoming call from endpoint device. 4. Verify that the call is connected successfully, and that media is established.
Result	Results logged into Table 4-1.
Notes	

Table 4-1: Results from Test Steps 1.7 & 1.8

Device or Application	Test Step 1.7 3rd Party DUT Originating	Test Step 1.8 3rd Party DUT Terminating
ShoreTel IP115	Pass	Pass
ShoreTel IP565	Pass	Pass
ShoreTel IP230	Pass	Pass
Analog Phone	Pass	Pass
ShoreTel softphone	Pass	Pass
Trunk phone	Pass	Pass
SIP ATA Mediatrix 2102	N/T	N/T
Voicemail	Fail	N/A
AutoAttendant	Fail	Pass
Backup AutoAttendant	Fail	Pass
Workgroup to available agent	N/A	N/A
Workgroup queue then to available agent	N/A	N/A
Office anywhere (Test for OB2)	N/A	N/A

ID	**1.9**
Name	**Place call – re-dial**
Mandatory or Optional:	**Optional**
Description	**Verify successful call placement using re-dial to SIP Reference**
Test steps	1. From **3P-1**, place call to **Sip-Ref** using the re-dial function. 2. Answer incoming call from **3P-1**. 3. Verify that the call is connected successfully.
Result	**Pass**
Notes	

ID	**1.10**
Name	**Place call – speed dial**
Mandatory or Optional:	**Optional**
Description	**Verify successful call placement using programmed speed dial**
Test steps	1. On the **3P-1** device, program a speed dial number for **SIP-Ref**. 2. From **3P-1**, place call to **SIP-Ref** using the speed dial function. 3. Answer incoming call from **3P-1**. 4. Verify that the call is connected successfully.
Result	**Pass**
Notes	

ID	**1.11**
Name	**CODEC support – common (from DUT to ShoreTel Phone, REF-x)**
Mandatory or Optional:	**Mandatory for G.711, Optional for other CODECs**
Description	**Verify successful call connection and audio path using all supported CODECs (G.711-Ulaw and G.729)**
Test steps	1. For each CODEC supported by both **3P-1** 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 **3P-1** and **REF-x** devices; do not yet enable capturing. 3. Re-initialize both **3P-1** and **REF-x** devices and verify that they are in a READY/IDLE state. 4. Enable packet capturing on the analyzer. 5. Perform tests 1.7 Place call and 1.8 Receive call for each supported CODEC with packet capturing enabled. 6. Disable packet capturing and find one decoded packet for both **3P-1** 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**
Notes	

ID	**1.12**
Name	**CODEC support – common (from DUT to SIP Reference Phone, SIP-Ref)**
Mandatory or Optional:	**Mandatory for G.711, Optional for other CODECs**
Description	**Verify successful call connection and audio path using all supported CODECs (G.711-Ulaw and G.729)**
Test steps	1. For each CODEC supported by both **3P-1** 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 **3P-1** and SIP-Ref devices; do not yet enable capturing. 3. Re-initialize both **3P-1** and SIP-Ref devices and verify that they are in a READY/IDLE state. 4. Enable packet capturing on the analyzer. 5. Perform tests 1.7 Place call and 1.8 Receive call for each supported CODEC with packet capturing enabled. 6. Disable packet capturing and find one decoded packet for both **3P-1** 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**
Notes	

ID	**1.13**
Name	**CODEC support – negotiated**
Mandatory or Optional:	**Mandatory (only if more than 1 CODEC is supported)**
Description	**Verify successful negotiation between devices configured with different default CODECs (G.711-Ulaw and G.729)**
Test steps	1. Configure **REF-x** device to G.711, configure **3P-1** device G.729 2. Attach the packet analyzer in between the **REF-x** and **3P-1** devices; do not yet enable capturing. 3. Re-initialize both **REF-x** and **3P-1** 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 **3P-1** device. 6. Disable packet capturing and inspect the decoded packets to verify that the CODEC negotiation occurred and **3P-1** device negotiated to G.711 if supported. 7. Enable packet capturing on the analyzer. 8. Place call from **3P-1** device to **REF-x** device. 9. Disable packet capturing and inspect the decoded packets to verify that the CODEC negotiation occurred and **3P-1** device negotiated to G.711 if supported. 10. Configure **REF-x** device to G.729, configure **3P-1** device G.711. 11. Attach the packet analyzer in between the **REF-x** and **3P-1** devices; do not yet enable capturing. 12. Re-initialize both **REF-x** and **3P-1** devices and verify that they are in a READY/IDLE state. 13. Enable packet capturing on the analyzer. 14. Place call from **3P-1** device to **REF-x** device. 15. Disable packet capturing and inspect the decoded packets to verify that the CODEC negotiation occurred and **3P-1** device negotiated to G.729 if supported. 16. Enable packet capturing on the analyzer. 17. Place call from **REF-x** device to **3P-1** device. 18. Disable packet capturing and inspect the decoded packets to verify that the CODEC negotiation occurred and **3P-1** device negotiated to G.729 if supported.
Result	**Pass**
Notes	

Table 4-2: Results from Test Step 1.13

Sequence #'s	ShoreTel REF-x codec setting	Direction of call	3P-1 codec setting	Results
1-6	**G.711**	➡	**G.729**	Pass
7-9	**G.711**	⬅	**G.729**	Pass
10-15	**G.729**	⬅	**G.711**	Pass
16-18	**G.729**	➡	**G.711**	Pass

ID	**1.14**
Name	**Hold from DUT to SIP Reference**
Mandatory or Optional:	**Mandatory**
Description	**Verify successful hold and resume of connected call**
Test steps	1. From **3P-1**, place call to **SIP-Ref**. 2. Answer incoming call from **3P-1**. 3. Verify that the call is connected successfully. 4. From **3P-1**, place **SIP-Ref** on hold. 5. Verify that the audio path is dropped. 6. Verify that the **3P-1** device appropriately indicates a held call still in progress. 7. From **3P-1**, resume the held call. 8. Verify that the audio path is restored and that the call proceeds as expected. 9. From **SIP-Ref**, place **3P-1** 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	

ID	**1.15**
Name	**Hold from DUT to ShoreTel Phone**
Mandatory or Optional:	**Mandatory**
Description	**Verify successful hold and resume of connected call**
Test steps	1. From **3P-1**, place call to **REF-x**. 2. Answer incoming call from **3P-1**. 3. Verify that the call is connected successfully. 4. From **3P-1**, place **REF-x** on hold. 5. Verify that the audio path is dropped. 6. Verify that the **3P-1** device appropriately indicates a held call still in progress. 7. From **3P-1**, resume the held call. 8. Verify that the audio path is restored and that the call proceeds as expected. 9. From **REF-x**, place **3P-1** on hold. 10. Verify that the audio path is dropped. 11. Verify that the **REF-x** device appropriately indicates a held call still in progress. 12. From **REF-x**, resume the held call. 13. Verify that the audio path is restored and that the call proceeds as expected.
Result	**Pass**
Notes	

ID	**1.16**
Name	**Forward**
Mandatory or Optional:	**Mandatory**
Description	**Verify successful forwarding of incoming calls**
Test steps	1. Configure the **REF-x** device to forward all calls to **3P-3**. 2. From **3P-2**, place call to **REF-x**. 3. Verify that the call is immediately forwarded to **3P-3**. 4. Answer incoming call from **3P-2**. 5. Verify that the call is connected successfully.
Result	**Pass**
Notes	

ID	**1.17**
Name	**Forward from SIP DUT**
Mandatory or Optional:	**TBD**
Description	**Verify successful forwarding of incoming calls**
Test steps	1. Configure the **3P-1** device to forward all calls to **3P-3**. 2. From **3P-2**, place call to **3P-1**. 3. Verify that the call is immediately forwarded to **3P-3**. 4. Answer incoming call from **3P-2**. 5. Verify that the call is connected successfully. 6. From **REF-x**, place call to **3P-1**. 7. Verify that the call is immediately forwarded to **3P-3**. 8. Answer incoming call from **REF-x**. 9. Verify that the call is connected successfully.
Result	**Not Supported.**
Notes	**Call forwarding is NOT SUPPORTED from end-point.**

ID	**1.18**
Name	**Mute**
Mandatory or Optional:	**Optional**
Description	**Verify device's mute function**
Test steps	1. From **3P-2**, place call to **REF-x**. 2. Answer incoming call from **3P-2**. 3. Verify the two-way audio path between both user devices. 4. Press the Mute button on the **3P-2** device. 5. Verify that incoming audio is still heard on **3P-2** but outgoing audio is muted. 6. Un-mute the call on the **3P-2** device. 7. Verify the two-way audio path between both user devices. 8. Press the Mute button on the **REF-x** device. 9. Verify that incoming audio is still heard on **REF-x** but outgoing audio is muted. 10. Un-mute the call on the **REF-x** device. 11. Verify the two-way audio path between both user devices.
Result	**Pass**
Notes	

ID	**1.19**
Name	**Out-of-band / In-band DTMF Transmission**
Mandatory or Optional:	**Mandatory**
Description	**Verify successful transmission of in-band and out-of-band digits (RFC2833) for calls placed to and from the DUT with a variety of other devices**
Test steps	1. Configure the **3P-1** device for out-of-band digit transmission **Calls originating with the DUT** 2. From **3P-1**, place call to terminating device specified in Table 4-3. 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 DUT** 6. From each originating device (specified in Table 4-4) if capable, place call to **3P-1**. 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	**Fail**
Notes	For calls originating with the DUT, log results into Table 4-3. For calls terminating with the DUT, log results into Table 4-4 Refer to table for details.

Table 4-3: Results from Test Step 1.19

(Originating Device always = 3rd Party DUT)

Terminating Device or Application	DTMF transmission Type		
	G.711 SIP Info (confirm inband DTMF)	G.711 RFC 2833	G.729 RFC 2833
3rd Party DUT	Pass	Pass	Pass
ShoreTel IP115	Pass	Pass	Pass
ShoreTel IP565	Pass	Pass	Pass
Analog Phone	Pass	Pass	Pass
ShoreTel softphone	Pass	Pass	Pass
Trunk phone	Pass	Pass	Pass
SIP ATA Mediatrix 2102	N/T	N/T	N/T
AutoAttendant	N/A	Fail	Fail
Backup AutoAttendant	N/A	Fail	Fail
Office Anywhere (For OB2 Release)	N/A	Fail	Fail

Table 4-4: Results from Test Step 1.19 (Terminating Device always = 3rd Party DUT)

Originating Device or Application	DTMF transmission Type		
	G.711 SIP Info (confirm inband DTMF)	G.711 RFC 2833	G.729 RFC 2833
3rd Party DUT	Pass	Pass	Pass
ShoreTel IP115	Pass	Pass	Pass
ShoreTel IP565	Pass	Pass	Pass
Analog Phone	Pass	Pass	Pass
ShoreTel softphone	Pass	Pass	Pass
Trunk phone	Pass	Pass	Pass
SIP ATA Mediatrix 2102	N/T	N/T	N/T
AutoAttendant	N/A	N/A	N/A
Backup AutoAttendant	N/A	N/A	N/A
Office Anywhere (For OB2 Release)	N/A	N/A	N/A

ID	**1.20**
Name	**Missed call notification**
Mandatory or Optional:	**Optional**
Description	**Verify that device notifies the user about missed calls**
Test steps	1. From **3P-2**, place a call to **3P-1** and do not answer the incoming call. 2. Abandon the call after one ring. 3. Verify that the **3P-1** device displays a notification that a call was missed. 4. Browse the call list and view the missed call. 5. Verify that the notification is removed once the call list has been browsed.
Result	**Not supported.**
Notes	

ID	**1.21**
Name	**Volume**
Mandatory or Optional:	**Optional**
Description	**Verify the device's volume adjustment function**
Test steps	1. From **3P-2**, place a call to **REF-x**. 2. Answer the incoming call. 3. Maintain audio between both user devices while adjusting the volume on the 3P-2 device. 4. Verify that the volume is adjusted appropriately; note any negative characteristics of the audio present at lower-than-normal or higher-than-normal volumes. 5. Maintain audio between both user devices while adjusting the volume on the **REF-x** device. 6. Verify that the volume is adjusted appropriately; note any negative characteristics of the audio present at lower-than-normal or higher-than-normal volumes. 7. If **DUT** has an outbound gain setting set the gain to the lowest setting and repeat steps 1-6; then set the gain to the highest setting and repeat steps 1-6. Verify gain settings will change the gain
Result	**Pass**
Notes	

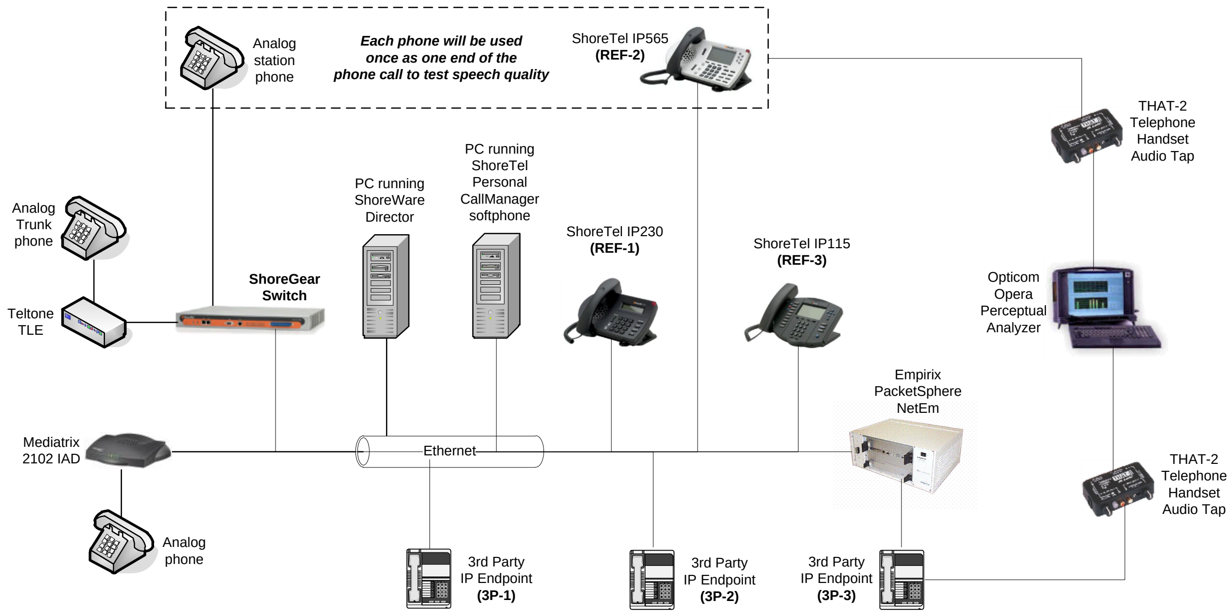
5. Performance Tests

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

5.1. Performance Test Setup Diagram

The basic performance VoIP Endpoint ITP test setup is shown in Figure 1 below. Depending on specific test cases, this diagram may be changed to show additional components or VoIP Endpoint types.

Figure 2: Basic Performance ShoreTel VoIP Endpoint ITP Test Setup



5.2. Performance Test Cases

ID	**2.1**
Name	**Speech quality – Minimal impairment**
Mandatory or Optional:	**Mandatory**
Description	**Verify acceptable voice quality between two parties with minimal network impairment condition**
Test steps	1. Verify "Minimal Impairment" settings on the network emulator (E.G. Empirix PacketSphere NetEm): ▪ 0 % packet-loss ▪ 40 ms (avg) normal distributed latency ▪ 10 ms jitter; 1 ms standard deviation 2. Configure the devices for each CODEC (G.711,G.729) if supported and run steps 3-8. 3. Place call between **3P-3** and **REF-2** 4. Verify two way audio is not degraded. Log the PSQM score if measured. 5. Hangup 6. Place call between **3P-3**and ShoreTel Analog Station Phone. 7. Verify two way audio is not degraded. 8. Hangup
Result	**Not Tested.**
Notes	

ID	**2.2**
Name	**Speech quality – Moderate Impairment**
Mandatory or Optional:	**Mandatory**
Description	**Verify acceptable voice quality between two parties with low-to-moderate artificial network impairment condition**
Test steps	1. Verify "Moderate Impairment" settings on the network emulator (E.G. Empirix PacketSphere NetEm) ▪ 1 % random packet-loss ▪ 100 ms (avg) normal distributed latency ▪ 15 ms jitter; 1 ms standard deviation 2. Configure the devices for each CODEC (G.711,G.729) if supported and run steps 3-8. 3. Place call between **3P-3**and **REF-2** 4. Verify two way audio is only moderately degraded. Log the PSQM score if measured. 5. Hangup 6. Place call between **3P-3**and ShoreTel Analog Station Phone. 7. Verify two way audio is not degraded. 8. Hangup
Result	**Not Tested.**
Notes	

ID	**2.3**
Name	**Speech quality – High Impairment**
Mandatory or Optional:	**Mandatory**
Description	**Verify acceptable voice quality between two parties with moderate-to-high artificial network impairment condition**
Test steps	1. Verify “High Impairment” settings on the network emulator (E.G. Empirix PacketSphere NetEm) ▪ 2.5 % random packet-loss ▪ 150 ms (avg) normal distributed latency ▪ 20 ms jitter; 1 ms standard deviation 2. Configure the devices for each CODEC (G.711,G.729) if supported and run steps 3-8. 3. Place call between **3P-3**and **REF-2** 4. Verify two way audio is still reasonable, even in the presence of this high level of WAN impairments. Log the PSQM score if measured. 5. Hangup 6. Place call between **3P-3**and ShoreTel Analog Station Phone. 7. Verify two way audio is not degraded. 8. Hangup
Result	**Not Tested.**
Notes	

6. Extended Features Test Cases

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

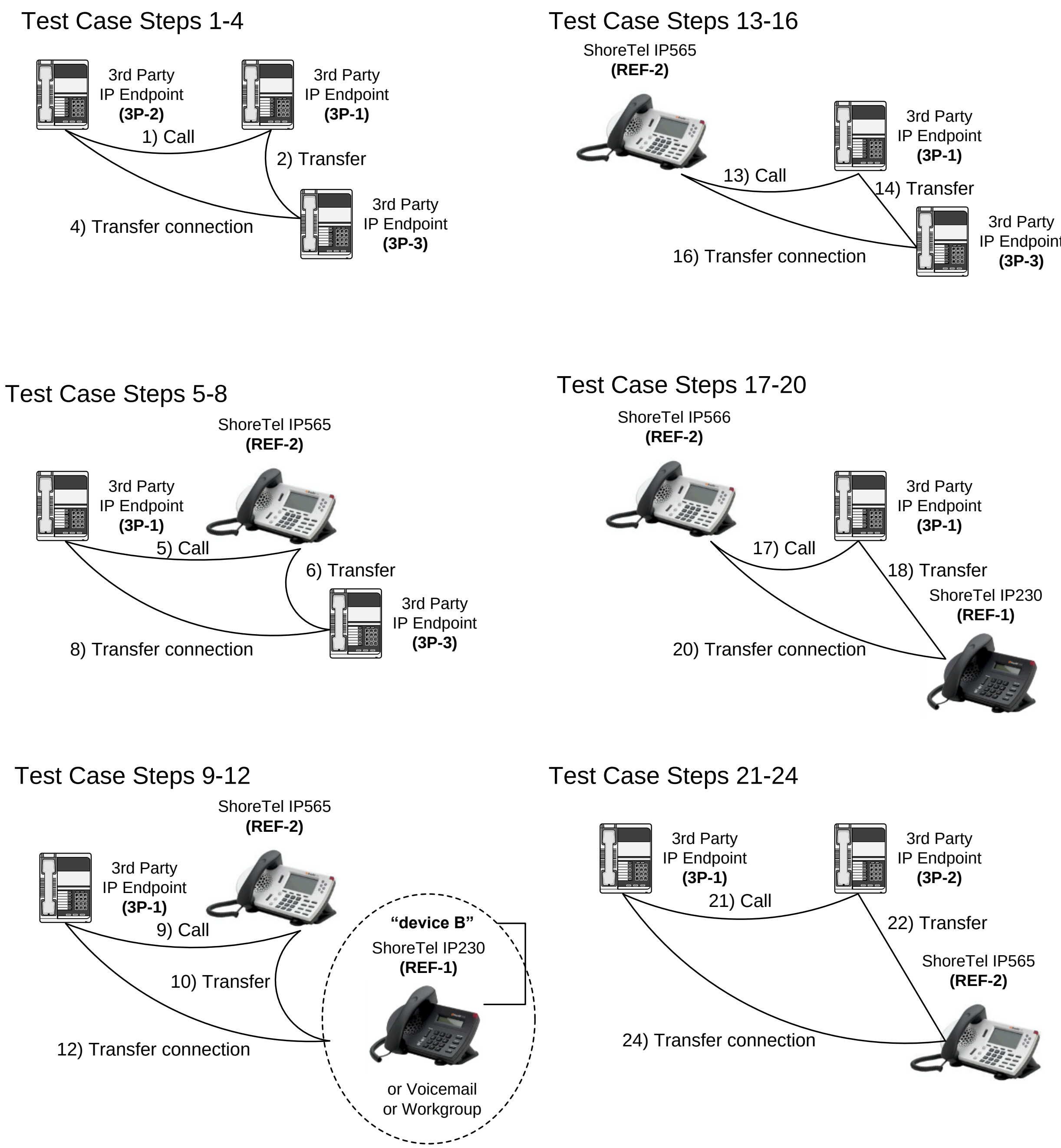
ID	**3.1**
Name	**Call waiting**
Mandatory or Optional:	**Mandatory**
Description	**Verify appropriate notification and successful connection of incoming call while busy with another party**
Test steps	1. From **3P-2**, call the **3P-1** device and verify that the call is connected successfully. 2. Leave this call active. 3. From **3P-3**, call the **3P-1** device. 4. Verify appropriate notification of the new incoming call to **3P-1**. 5. Answer the new incoming call and verify that the call is connected successfully. 6. Verify that **3P-2** has been placed on hold (no audio to or from the **3P-2** device). 7. Terminate the new call and verify that the original call with **3P-2** may be resumed successfully. 8. From a reference SIP phone, call the **3P-1** device. 9. Verify appropriate notification of the new incoming call to **3P-1**. 10. Answer the new incoming call and verify that the call is connected successfully. 11. Verify that **3P-2** has been placed on hold (no audio to or from the **3P-2** device). 12. Terminate the new call and verify that the original call with **3P-2** may be resumed successfully.
Result	**Pass**
Notes	

ID	**3.2**
Name	**Park**
Mandatory or Optional:	**TBD**
Description	**Verify successful park and retrieval of connected call**
Test steps	1. From **REF-1**, call the **3P-1** device and verify that the call is connected successfully. 2. From **REF-1**, park the call and note which park number the call may be retrieved from. 3. From **REF-2**, call the noted park number and verify that the parked call may be retrieved successfully.
Result	**Pass**
Notes	

ID	3.3
Name	**Extended forward**
Mandatory or Optional:	**Optional**
Description	**Verify extended call forwarding options – busy forwarding, no-answer forwarding**
Test steps	1. For the **REF-x** device, configure "no-answer" and "busy" call forwarding. No-answer forwarding should route to the **3P-2** device and busy forwarding should route to the **3P-3** device. 2. From **3P-3**, call **REF-x** and do not answer the incoming call. 3. Verify that the call is forwarded to the **3P-2** device. 4. Answer the incoming call from the **3P-2** device and leave this call active. 5. From **3P-1**, call **REF-x** and verify that the call is forwarded to the **3P-3** device.
Result	**Pass**
Notes	

ID	3.4
Name	**Extended forward from SIP DUT**
Mandatory or Optional:	**Optional**
Description	**Verify extended call forwarding options – busy forwarding, no-answer forwarding**
Test steps	1. For the **3P-1** device, configure "no-answer" and "busy" call forwarding. No-answer forwarding should route to the **3P-2** device and busy forwarding should route to the **3P-3** device. 2. From **3P-3**, call **3P-1** and do not answer the incoming call. 3. Verify that the call is forwarded to the **3P-2** device. 4. Answer the incoming call from the **3P-2** device and leave this call active. 5. From **REF-x**, call **3P-1** and verify that the call is forwarded to the **3P-3** device.
Result	**Pass**
Notes	

Figure 3: Transfer Scenarios Drawing (for Test Cases 3.5 and 3.6)



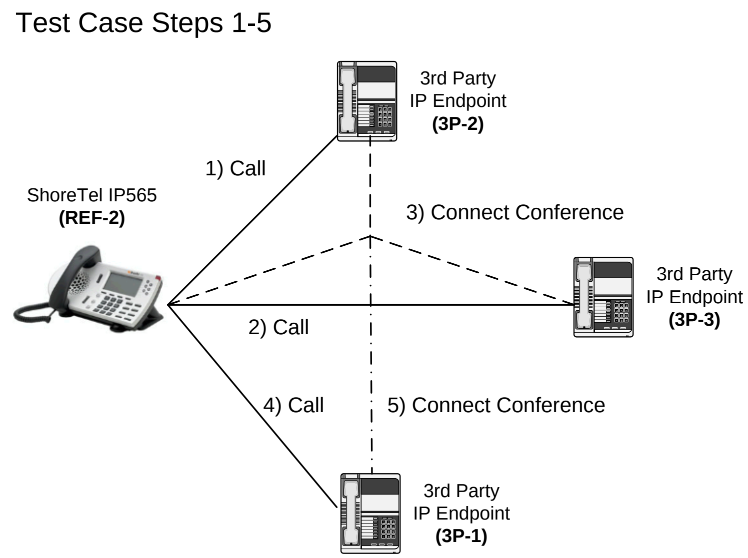
ID	**3.5**
Name	**Transfer – blind**
Mandatory or Optional:	**Mandatory**
Description	**Verify successful blind transfer of connected call**
Test steps	**Call #1** 1. From **3P-2**, call the **3P-1** device and verify that the call is connected successfully. 2. From **3P-1**, transfer the call to the **3P-3** device and hang up before answering the new incoming call. 3. After hanging up the **3P-1** device, verify that the **3P-3** device is still ringing. 4. Answer the incoming call and verify that the **3P-2** and **3P-3** devices have been connected successfully. **Call #2** 5. From **3P-1**, call **REF-2** and verify that the call is connected successfully. 6. From **REF-2**, transfer the call to the **3P-3** device and hang up before answering the new incoming call. 7. After hanging up **REF-2**, verify that the **3P-3** device is still ringing. 8. Answer the incoming call and verify that the **3P-1** and **3P-3** devices have been connected successfully **Call #3-5** 9. From **3P-1**, call **REF-2** and verify that the call is connected successfully. 10. From **REF-2**, transfer the call to “**device B**” (see notes) and hang up before answering the new incoming call. 11. After hanging up **REF-2**, verify “**device B**” is still ringing. 12. Answer the incoming call and verify that the **3P-1** and “**device B**” have been connected successfully. **Call #6** 13. From **REF-2**, call the **3P-1** device and verify that the call is connected successfully. 14. From **3P-1**, transfer the call to the **3P-3** device and hang up before answering the new incoming call. 15. After hanging up the **3P-1** device, verify that the **3P-3** device is still ringing. 16. Answer the incoming call and verify that **REF-2** and **3P-3** devices have been connected successfully. **Call #7** 17. From **REF-2**, call the **3P-1** device and verify that the call is connected successfully. 18. From **3P-1**, transfer the call to **device B** and hang up before answering the new incoming call. 19. After hanging up the **3P-1** device, verify that **device B** is still ringing. 20. Answer the incoming call and verify that **REF-2** and **device B** have been connected successfully. **Call #8** 21. From **3P-1**, call the **3P-2** device and verify that the call is connected successfully. 22. From **3P-2**, transfer the call to **REF-2** and hang up before answering the new incoming call. 23. After hanging up the **3P-2** device, verify that the **REF-2** is still ringing. 24. Answer the incoming call and verify that the **3P-1** and **REF-2** have been connected successfully

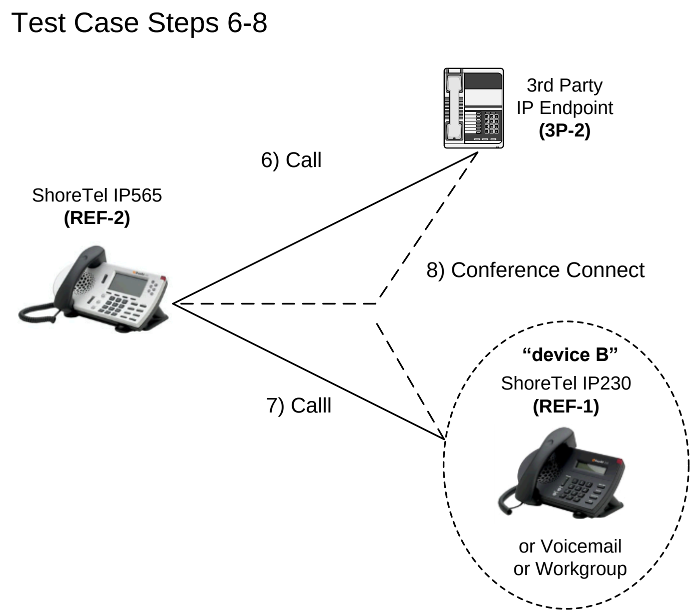
ID	**3.5**
Name	**Transfer – blind**
Result	**Call #1: Not Supported.** **Call #2: Pass.** **Call #3: Pass.** **Call #4: Pass.** **Call #5: Pass.** **Call #6: Not Supported.** **Call #7: Not Supported.** **Call #8: Not Supported.**
Notes	See Figure 3: Transfer Scenarios Drawing (for Test Cases 3.5 and 3.6) (above) For Call #3, "**device B**" is REF-1 (ShoreTel IP530) For Call #4, "**device B**" is voicemail. For Call #5, "**device B**" is workgroup.

ID	**3.6**
Name	**Transfer – monitored**
Mandatory or Optional:	**Mandatory**
Description	**Verify successful monitored transfer of connected call**
Test steps	**Call #1** 1. From **3P-2**, call the **3P-1** device and verify that the call is connected successfully. 2. From **3P-1**, transfer the call to the **3P-3** device; do not hang up the **3P-1** device. 3. From **3P-3**, answer the incoming call and verify that only **3P-1** and **3P-3** are connected. 4. Complete the call transfer and verify that **3P-2** and **3P-3** are successfully connected. **Call #2** 5. From **3P-1**, call **REF-2** and verify that the call is connected successfully. 6. From **REF-2**, transfer the call to the **3P-3** device; do not hang up **REF-2**. 7. From **3P-3**, answer the incoming call and verify that only **REF-2** and **3P-3** are connected. 8. Complete the call transfer and verify that **3P-1** and **3P-3** are successfully connected. **Call #3-5** 9. From **3P-1**, call **REF-2** and verify that the call is connected successfully. 10. From **REF-2**, transfer the call to **device B** (see notes); do not hang up **REF-2**. 11. From **device B**, answer the incoming call and verify that only **REF-2** and **device B** are connected. 12. Complete the call transfer and verify that **3P-1** and **device B** are successfully connected. **Call #6** 13. From **REF-2**, call the **3P-1** device and verify that the call is connected successfully. 14. From **3P-1**, transfer the call to the **3P-3** device; do not hang up the **3P-1** device. 15. From **3P-3**, answer the incoming call and verify that only **3P-1** and **3P-3** are connected. 16. Complete the call transfer and verify that **REF-2** and **3P-3** are successfully connected. **Call #7** 17. From **REF-2**, call the **3P-1** device and verify that the call is connected successfully. 18. From **3P-1**, transfer the call to the **device B**; do not hang up the **3P-1** device. 19. From **device B**, answer the incoming call and verify that only **3P-1** and **device B** are connected. 20. Complete the call transfer and verify that **REF-2** and **device B** are successfully connected. **Call #8** 21. From **3P-1**, call the **3P-2** device and verify that the call is connected successfully. 22. From **3P-2**, transfer the call to **REF-2** – DO NOT hang up the **3P-2** device. 23. From **REF-2**, answer the incoming call and verify that only **3P-2** and **REF-2** are connected. 24. Complete the call transfer and verify that **3P-1** and **REF-2** are successfully connected.

ID	**3.6**
Name	**Transfer – monitored**
Result	**Call #1 – Not Supported.** **Call #2 – Pass.** **Call #3 – Pass.** **Call #4 – Pass.** **Call #5 – Pass.** **Call #6 – Not Supported.** **Call #7 – Not Supported.** **Call #8 – Not Supported.**
Notes	See Figure 3: Transfer Scenarios Drawing (for Test Cases 3.5 and 3.6) (above) For Call #3, "**device B**" is REF-1 (ShoreTel IP530) For Call #4, "**device B**" is voicemail. For Call #5, "**device B**" is workgroup.

Figure 4: Conference Scenario Drawing (for Test Case 3.7)





ID	3.7
Name	**Conference – ad hoc**
Mandatory or Optional:	**Mandatory**
Description	**Verify successful ad hoc conference of three parties**
Test steps	**Call #1** 1. From **REF-2**, call **3P-2** and verify that the call is connected successfully. 2. From **REF-2**, place **3P-2** on hold and call **3P-3**. 3. Complete the conference and verify the audio path between all 3 devices. 4. From **REF-2**, place conference on hold and call **3P-1** 5. Complete the conference and verify the audio path between all 4 devices **Call #2-4** 1. From **REF-2**, call **3P-2** and verify that the call is connected successfully. 2. From **REF-2**, place **3P-2** on hold and call "**device B**"(see notes). 3. Complete the conference and verify the audio path between all 3 devices **Call #5** 1. From **3P- 2**, call **REF-2** and verify that the call is connected successfully. 2. From **3P-2**, place **REF-2** on hold and call **3P-3**. 3. Complete the conference and verify the audio path between all 3 devices. 4. From **3P-2**, place conference on hold and call **3P-1** 5. Complete the conference and verify the audio path between all 4 devices **Call #6-8** 1. From **3P-2**, call **REF-2** and verify that the call is connected successfully. 2. From **3P-2**, place **REF-2** on hold and call "**device B**"(see notes). 3. Complete the conference and verify the audio path between all 3 devices
Result	**Call #1: Pass** **Call #2: Pass** **Call #3: Pass** **Call #4: Pass** **Call #5: N/S (The 3rd party phone can only have 2 other phones in conference.)** **Call #6: Pass** **Call #7: Pass (select only one codec from the list of codecs in the 3rd party phone. if all the codecs are selected, conference results in noise.)** **Call #8: Pass**
Notes	See Figure 4: Conference Scenario Drawing (for Test Case 3.7) (above) For Call #2, "**device B**" is REF-1 (ShoreTel IP530) For Call #3, "**device B**" is voicemail. For Call #4, "**device B**" is workgroup. Note: Conference might have to be initiated from a non SIP endpoint.

ID	**3.8**
Name	**Place call – secondary line**
Mandatory or Optional:	**Optional**
Description	**Verify successful call placement using secondary line**
Test steps	1. Configure a secondary line on the **3P-1** device. 2. From **3P-1**'s primary line, call **3P-2**. 3. Verify that the incoming call notification on the **3P-2** device indicates an incoming call from **3P-1**'s primary line. 4. Answer the incoming call and verify that the call is connected successfully. 5. Put the call on hold and access the secondary line on **3P-1**. 6. From **3P-1**'s secondary line, call **3P-2**. 7. Verify that the incoming call notification on the **3P-2** device indicates an incoming call from **3P-1**'s secondary line. 8. Answer the incoming call and verify that the call is connected successfully.
Result	**Pass**
Notes	Same extension number is used for Line 1 and Line 2.

ID	**3.9**
Name	**Receive call – secondary line**
Mandatory or Optional:	**Optional**
Description	**Verify successful connection of incoming call on secondary line**
Test steps	1. Make sure the ShoreTel system is configured with static IP for individual trunks. 2. Make a call from **3P-3** to **3P-1** and verify successful connection. 3. From **3P-2**, call **3P-1**’s secondary line. 4. Verify that the incoming call notification on the **3P-1** device indicates an incoming call on **3P-1**'s secondary line. 5. Answer the incoming call and verify that the call is connected successfully.
Result	**Pass**
Notes	Same extension number. Second call appearance is verified.

ID	**3.10**
Name	**Callback**
Mandatory or Optional:	**Optional**
Description	**Verify successful connection of a call using the missed-call callback feature of the device**
Test steps	1. From **3P-2**, call **3P-1** and do not answer the incoming call. 2. Terminate the call attempt from **3P-2**. 3. Browse the "missed calls" list on the **3P-1** device use the callback feature from this list to return the call to **3P-2**. 4. Answer the incoming call and verify that the call is connected successfully.
Result	**Not Supported.**
Notes	

ID	**3.11**
Name	**Headset**
Mandatory or Optional:	**Optional**
Description	**Verify the device's support for external headsets (using headsets supplied by the 3P phone vendor)**
Test steps	1. Connect a headset to the **3P-1** device's headset jack. 2. From **3P-1**, place call to **3P-2**. 3. Answer the incoming call. 4. Verify that the incoming audio is heard through the headset. 5. From the **3P-2** device, verify that the headset microphone audio quality is acceptable (note characteristics of audio below if the quality is poor)
Result	**Not Supported.**
Notes	

ID	**3.12**
Name	**Ring selection**
Mandatory or Optional:	**Optional**
Description	**Verify the device's ability to change the ring type**
Test steps	1. Change the default ring type of the **3P-1** device. 2. From **3P-2**, place a call to **3P-1** and do not answer the incoming call. 3. Verify that the new ring type is played on the **3P-1** device. 4. Abandon the call attempt. 5. Change the ring selection on the **3P-1** device back to the default ring type. 6. From **3P-2**, place a call to **3P-1** and do not answer the incoming call. 7. Verify that the default ring type is played on the **3P-1** device.
Result	**Not Supported.**
Notes	

ID	**3.13**
Name	**Caller ID Name and Number**
Mandatory or Optional:	**Mandatory**
Description	**Verify that Caller ID name and number is sent and received from SIP endpoint device**
Test steps	1. Activate VBTRunkTest tool. 2. From **3P-1**, place call to **3P-2**. 3. Verify Caller ID number is sent and received from SIP endpoint devices. Note: the verification can be done in the VBTrunkTest tool by checking what the Inside-ID says in the SIP messages for the call. 4. From **3P-1**, place call to **REF-2**. 5. Verify Caller ID number is sent and received from **3P-1** and **REF-2** 6. From **REF-2**, place call to **3P-1**. 7. Verify Caller ID number is sent and received from **3P-1** and **REF-2** 8. From Analog Trunk Phone, place call to Auto Attendant which forwards to **3P-1**. 9. Verify Caller ID number is sent and received from Analog Trunk Phone and **3P-1**
Result	**Pass**
Notes	

ID	**3.14**
Name	**SIP Device Generates Busy Tone**
Mandatory or Optional:	**Optional**
Description	**Verify that SIP DUT generates busy tone when calling a busy extension**
Test steps	Note: If Call Waiting is enabled for **3P-1**, you will need to place two calls instead of just one in order to "busy out" the device. 1. From **3P-1**, place call to **3P-3** and verify connection is established 2. From **3P-1** put call on hold and access secondary line. 3. From **3P-1**, place call to **3P-2** and verify connection is established. 4. From **REF-2**, place call to **3P-1** 5. Verify **REF-2** receives busy tone in the audio path, and hang up the call. Note: The ShoreTel REF phones are set up to have a call stack of 8 calls. Go into the ShoreTel Operator Call Manager, select Options and Configure ShoreTel System, and under the Telephony tab, change the call stack size to 1. 6. From **REF-2**, place call to **3P-2**. 7. From **3P-1** place call to **REF-2** 8. Verify **3P-1** receives busy tone in the audio path.
Result	**Pass**
Notes	In Biamp architecture, each interface can be configured as two phones. On the interface LAN card configured with 2 phones: we have to set the call stack on ShoreTel Director for that Individual User to 3 or more to receive incoming calls on the 1st phone show in the Biamp Interface. When the call stack is set to 3, 2 outgoing calls can be made (3rd call fails) and 1 incoming call can be made.

ID	**3.15**
Name	**POTS Analog Gateway supports the transfer operation by “flashing”**
Mandatory or Optional:	**TBD**
Description	**Verify that the POTS Analog Gateway can support the transfer operation by “flashing”**
Test steps	1. Establish call across Analog Gateway 2. From Analog device connected to POTS Analog Gateway generate “flash” 3. Verify transfer operation is supported
Result	**Not Applicable.**
Notes	Applies to POTS analog gateways only

ID	**3.16**
Name	**Verify handling of “911”**
Mandatory or Optional:	**Mandatory**
Description	**Verify dialing “911” on DUT could connect with “911” services**
Test steps	1. Activate VBTrunkTest tool 2. Select line 3. Dial “911” digits on phone 4. Verify could connect with “911” services (check Teltone TLE display for “911” as the dial string coming out of the ShoreTel system) 5. Verify that Caller ID information is sent. (check in the VBTrunkTest tool), Not possible with a GW that has an analog Trunk for outbound calls
Result	**Not Applicable.**
Notes	An additional GW with a trunk for outbound calls needs to be configured in the ShoreGear. This test does not support Caller ID information.

ID	**3.17**
Name	**Verify Fax Handling**
Mandatory or Optional:	**Mandatory**
Description	**Verify that fax can be sent and received through DUT**
Test steps	1. Configure DUT to not use T38 for fax. 2. Set up a User account for an internal redirect fax machine (ext 199) 3. Enable “always forward” in the Personal Options section of the account to always forward to **3P-1** (where you will have the fax machine attached) 4. Go to “sites” and select Headquarters. Set the FAX redirect extension (199) to be the extension you set up for the user account in step 2. Fax #1 – analog trunk to **3P-1** 5. Attach a fax machine to the analog trunk on the Teltone TLE. 6. Attach another fax machine to the **3P-1**. 7. Send a fax from the fax machine on the analog trunk to the fax machine on **3P-1**. 8. Verify that the fax is sent and received successfully. Fax #2 9. Leave fax machines attached as they were for Fax #1 10. Send a fax from the fax machine on the analog trunk to **REF-x** 11. Answer **REF-x** 12. The call should be automatically transferred to the site fax machine 13. Verify that the fax is sent and received successfully. Fax #3 – **3P-1**to **3P-2** 14. Leave one fax machine attached to **3P-1** 15. Attach the other fax machine to **3P-2** 16. Send a fax from **3P-1** to **3P-2** 17. Verify that the fax is sent and received successfully.
Result	Fax #1: **Not Supported.** Fax #2: **Not Supported.** Fax #3: **Not Supported.**
Notes	Perform only on DUTs that normally handle fax calls.

ID	3.18
Name	**Call to 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.**
Test steps	1. Initiate a call to a ShoreTel Hunt Group from the DUT. 2. Verify that the call routes to the proper Hunt Group and is answered by an available hunt group member and there's two way communication. 3. Blind transfer the call (from the hunt group member) 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 dialed in step 1, to be another Hunt Group extension. 5. Place another call to the Hunt Group extension, do not answer the call and allow it to route to the second Hunt Group. Verify that the call first routes to the initial Hunt Group and then to the second Hunt Group, ringing all available members. Answer the call from an available member in the second hunt group. 6. Perform a Consultative Transfer (from the hunt group member) to another extension, verify that the destination extension rings and the call is successfully transferred with two way communication.
Result	**Pass***
Notes	Steps 1-3 are working correctly, but when no one in hunt group 1 is available to answer the call, the call has to transfer from hunt group 1 to hunt group 2. When the call is made from a 3rd party phone the call is not transferred. Display in the 3rd party phone shows connected to hunt group 1.

ID	**3.19**
Name	**Call to 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.**
Test steps	1. Initiate a call to a ShoreTel Workgoup from the DUT. 2. Verify that the call routes to the proper Workgroup and is answered by an available workgroup agent and there's two way communication. 3. Blind transfer the call (from the workgroup agent) 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 Workgroup dialed in step 1, to be another Workgroup extension. 5. Place another call to the Workgroup extension, do not answer the call and allow it to route to the second Workgroup. Verify that the call first routes to the initial Workgroup and then to the second Workgroup, ringing all available workgroup agents. Answer the call from an available agent in the second workgroup. 6. Perform a Consultative Transfer (from the workgroup agent) to another extension, verify that the destination extension rings and the call is successfully transferred with two way communication.
Result	**Pass***
Notes	The call cannot be transferred to workgroup. Steps 5-6 are Not Supported.

ID	**3.20**
Name	**Hunt Group Member**
Mandatory or Optional:	**Mandatory**
Description	**Verify that the DUT can receive calls while configured to be a hunt group member**
Test steps	1. Configure the DUT to be a hunt group member. 2. Initiate a call to the Hunt Group. 3. Verify that the DUT rings and you can successfully answer the call and there's two way communication. 4. From the DUT, place the call on-hold for 30 seconds, resume the call and verify two way audio path. 5. From the DUT, blind transfer the call to another extension and verify that the call rings the other extension and is answered successfully with two way audio path.
Result	**Pass**
Notes	Transfer from the DUT is Not Supported.

ID	**3.21**
Name	**Workgroup Agent:**
Mandatory or Optional:	**Mandatory**
Description	**Verify that DUT can receive calls while configured to be a workgroup agent.**
Test steps	1. Configure the DUT to be a workgroup agent. 2. Initiate a call to the Workgroup. 3. Verify that the DUT rings and you can successfully answer the call and there's two way audio path. 4. From the DUT, place the call on-hold for 30 seconds, resume the call and verity two way audio path. 5. From the DUT, transfer the call consultatively to another extension and verify that the call rings the other extension and is answered successfully. Verify that the originating party cannot hear the consultative call, complete the transfer. Verify that the call was successfully transferred and the DUT is now idle (on-hook).
Result	**Pass**
Notes	Transfer from the DUT is Not Supported.

ID	**3.22**
Name	**Call Forward – FindMe**
Mandatory or Optional:	**Mandatory**
Description	**Verify that callers are forwarded to the "FindMe" destination.**
Test steps	1. Configure an existing ShoreTel extension "FindMe" destination to be the DUT extension number. 2. Place a 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 a call to the desired DUT extension number. 4. Verify that the DUT rings, answer the call and verify that the there's two way communication.
Result	**Pass**
Notes	

ID	**3.23**
Name	**Communicator**
Mandatory or Optional:	**Mandatory**
Description	**Verify that a call can be initiated from ShoreTel Communicator, that it rings the DUT, and once answered, a call is placed to the desired destination.**
Test steps	1. Configure the DUT to be managed by ShoreTel Communicator. 2. Place a call using Communicator to another ShoreTel extension or external PSTN number 3. Verify that the DUT rings, answer the phone 4. Verify that the destination device rings, answer the called party 5. Verify there's two way audio path. 6. Hang-up from the called party and verify that the DUT hangs up
Result	**Pass**
Notes	

ID	**3.24**
Name	**Simultaneous Ring**
Mandatory or Optional:	**Mandatory**
Description	**Verify that an inbound call to a user's primary ShoreTel IP Phone also rings the DUT**
Test steps	1. Configure the DUT to be an "Additional Phone" to a user's extension 2. Verify that inbound calls ring both phones (ShoreTel IP Phone and DUT). 3. Answer the call the call on the DUT, verify two way audio path. 4. On the ShoreTel IP Phone press the "Move" softkey and verify that the call goes to the main phone. Verify that the DUT rings, answer the phone 5. Once again, press the "Move" softkey and verify that the call goes back to DUT 6. Hang-up the call from the DUT and verify the calling party is disconnected
Result	**Pass**
Notes	

End of Test Plan