



## SIP ATA Interoperability Test Case

## Document Revision

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1.1	Added expected values for correct tones	03/05/2012	Rajesh Naidu
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# 1 Scope and Purpose

## 1.1 Document Purpose

This document describes the test cases to validate the ATA to VSP interoperability

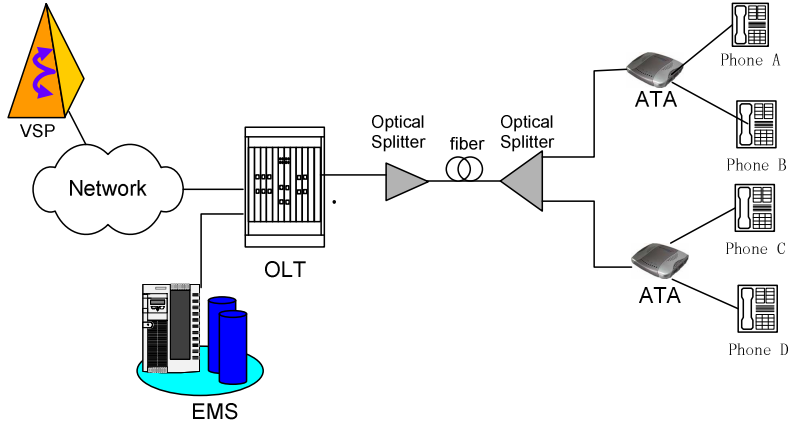
## 1.2 Scope

The present document specifies test cases to validate the interoperability between a VSP and ATA based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP) to enable an IP based telephony service.

# 2 Test Case

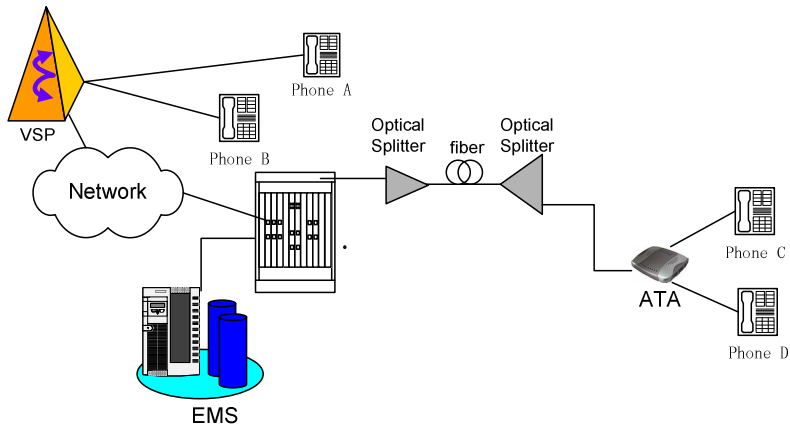
## Basic Calling Services

### 2.1 SIP Registration (Two users under same ATA)

<p><b>Purpose</b></p>	<p>To verify the registration and dial tone feature of the ATA</p>
<p><b>Test setup</b></p>	 <p style="text-align: center;">Figure 1</p>
<p><b>Prerequisite</b></p>	<p>1. Network is created as per the figure 1.</p>

	2. Configure user A and B in the same ATA
<b>Procedure</b>	1. Verify that both phone A and phone B has the correct NZ dial tone.
<b>Expected result</b>	1. Both phones A and B has registered with the softswitch and correct NZ dial tone ( <b>400 Hz continuous</b> ) can be heard
<b>Test result</b>	
<b>Pass/Fail</b>	
<b>Date/Signature</b>	
<b>Remarks</b>	

**2.2 Basic Call On-Net (ATA call the other ATA User )**

<b>Purpose</b>	To verify the on-net calling feature of the ATA
<b>Test setup</b>	 <p style="text-align: center;">Figure 2</p>
<b>Prerequisite</b>	<ol style="list-style-type: none"> <li>1. Network is created as per figure 2 above</li> <li>2. Make sure that the ATA works in the normal mode with correct</li> </ol>



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	dial tones.
<b>Procedure</b>	<ol style="list-style-type: none"><li>1. Phone C makes a outbound call to Phone D</li><li>2. Phone D receives local ring tone</li><li>3. Phone D answers the call</li><li>4. Capture the packets to verify local bridging is not happening</li></ol>
<b>Expected result</b>	<ol style="list-style-type: none"><li>1. ON-Net Call can be completed successfully.</li><li>2. The call is not bridged locally, the traffic goes upstream and back.</li></ol>
<b>Test result</b>	
<b>Pass/Fail</b>	
<b>Date/Signature</b>	
<b>Remarks</b>	

### 2.3 Basic Call Off-Net Inbound/outbound ( Calling to and from mobile phone )

<b>Purpose</b>	To verify the off-net calling feature of the ATA
<b>Test setup</b>	Same as figure2
<b>Prerequisite</b>	<ol style="list-style-type: none"><li>1. Network is created as per figure 2 above. \</li><li>2. Make sure that the ATA works in the normal mode with correct dial tones.</li></ol>
<b>Procedure</b>	<ol style="list-style-type: none"><li>1. Phone C makes a call to a local mobile number.</li><li>2. Verify that correct ring back tone is generated</li><li>3. Local mobile makes a call to Phone C</li><li>4. Verify that correct ringing cadence is received at Phone C.</li></ol>
<b>Expected result</b>	<ol style="list-style-type: none"><li>1. Outbound call is made successfully</li><li>2. Correct ring back tone is heard at Phone C</li><li>3. Inbound call is received successfully</li><li>4. Correct ring cadence(<b>400 Hz, modulated at 16 2/3 Hz, interrupted,400 ms on, 200 ms off,400 ms on, 2 sec.off, all repeated</b>)is played locally at phone C.</li></ol>



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Test result	
Pass/Fail	
Date/Signature	
Remarks	

### 2.4 Call Waiting (Hook flash)

Purpose	To verify the call waiting feature of the ATA
Test setup	Same as figure2
Prerequisite	<ol style="list-style-type: none"><li>1. Network is created as per figure 2 above</li><li>2. Make sure that the ATA works in the normal mode with correct dial tones</li></ol>
Procedure	<ol style="list-style-type: none"><li>1. Inbound call is made to phone C</li><li>2. Phone C goes off-hook and starts conversation</li><li>3. A 2nd inbound call is made to phone C</li><li>4. Phone C hears a call waiting tone</li><li>5. Phone C does a hookflash to answer the 2nd call</li><li>6. Phone C does a hook flash to get back to the 1st call</li><li>7. Verify that both the calls are terminated correctly</li></ol>
Expected result	<ol style="list-style-type: none"><li>1. Correct call waiting tone (<b>400 Hz interrupted,200 ms on, 3 sec. off,200 ms on, 3 sec. off,200 ms on, 3 sec. off,200 ms on, 3 sec. off,200 ms on, not repeated</b>) is heard on Phone C</li><li>2. Hook flash does the call swaps correctly</li><li>3. Both the calls terminate correctly.</li></ol>
Test result	
Pass/Fail	
Date/Signature	
Remarks	

### 2.5 Call Forwarding (CFNA/CFB/CFA)

<b>Purpose</b>	To verify the call forwarding feature of the ATA
<b>Test setup</b>	Same as figure2
<b>Prerequisite</b>	<ol style="list-style-type: none"> <li>1. Network is created as per figure 2 above</li> <li>2. Make sure that the ATA works in the normal mode with correct dial tones</li> </ol>
<b>Procedure</b>	<ol style="list-style-type: none"> <li>1. Phone C is configured for CFNA</li> <li>2. An inbound call is made to Phone C</li> <li>3. Phone C is configured for CFB</li> <li>4. An inbound call is made to Phone C</li> <li>5. Phone C is configured for CFA</li> <li>6. An inbound call is made to Phone C</li> </ol>
<b>Expected result</b>	<ol style="list-style-type: none"> <li>1. Call Forward No Answer (CFNA) works normally。</li> <li>2. Call Forward Busy (CFB) works normally</li> <li>3. Call Forward All (CFA) works normally</li> </ol>
<b>Test result</b>	
<b>Pass/Fail</b>	
<b>Date/Signature</b>	
<b>Remarks</b>	

### 2.6 Basic Call (Unobtainable Number)

<b>Purpose</b>	To verify the calling feature of the ATA
<b>Test setup</b>	Same as figure 1
<b>Prerequisite</b>	<ol style="list-style-type: none"> <li>1. Network is created as per figure 1 above</li> <li>2. Make sure that the ATA works in normal mode with correct dial tones</li> </ol>
<b>Procedure</b>	<ol style="list-style-type: none"> <li>1. Caller A makes a call to a number that is unobtainable</li> <li>2. Observe the correct tone played by the ATA</li> </ol>
<b>Expected result</b>	<ol style="list-style-type: none"> <li>1. Correct tone(400 Hz interrupted,75 ms on, 100 ms off,75 ms on, 100 ms off,75 ms on, 100 ms off,75 ms on, 400 ms off, all</li> </ol>





	<b>repeated)</b> is heard by the caller after dialling an unobtainable number
<b>Test result</b>	
<b>Pass/Fail</b>	
<b>Date/Signature</b>	
<b>Remarks</b>	

### 2.7 Basic Call (incomplete phone number)

<b>Purpose</b>	To verify the calling feature of the ATA
<b>Test setup</b>	Same as figure1
<b>Prerequisite</b>	<ol style="list-style-type: none"><li>1. Network is created as per figure 1 above</li><li>2. Make sure that the ATA works in normal mode with correct dial tones.</li></ol>
<b>Procedure</b>	<ol style="list-style-type: none"><li>1. Caller A makes a call to an incomplete number.</li></ol>
<b>Expected result</b>	<ol style="list-style-type: none"><li>1. Unobtainable tone (<b>400 Hz interrupted,75 ms on, 100 ms off,75 ms on, 100 ms off,75 ms on, 100 ms off,75 ms on, 400 ms off, all repeated</b>) is heard by the caller after waiting for timeout</li></ol>
<b>Test result</b>	
<b>Pass/Fail</b>	
<b>Date/Signature</b>	
<b>Remarks</b>	

### 2.8 Basic Call (Callee Busy)

<b>Purpose</b>	To verify the calling feature of the ATA
<b>Test setup</b>	Same as figure2

<b>Prerequisite</b>	<ol style="list-style-type: none"> <li>1. Network is created as figure 2 above.</li> <li>2. Make sure that the ATA works in normal mode and with correct dial tones</li> </ol>
<b>Procedure</b>	<ol style="list-style-type: none"> <li>1. Phone B is talking with Phone C. At this time, Phone A calls Phone B.</li> </ol>
<b>Expected result</b>	<ol style="list-style-type: none"> <li>1. In step 1, Phone A can hear the busy (<b>400 Hz interrupted,500 ms on, 500 ms off, repeated</b>) tone.</li> </ol>
<b>Test result</b>	
<b>Pass/Fail</b>	
<b>Date/Signature</b>	
<b>Remarks</b>	

## 2.9 DHCP function

<b>Purpose</b>	To verify the DHCP function of the ATA
<b>Test setup</b>	Same as figure1
<b>Prerequisite</b>	<ol style="list-style-type: none"> <li>1. Follow the network diagram to set up the test system.</li> <li>2. DHCP server works in the normal state</li> <li>3. Enable the DHCP function on the vlan interface</li> </ol>
<b>Procedure</b>	<ol style="list-style-type: none"> <li>1. Configure the signaling IP and media IP user DHCP mode</li> <li>2. Configure the SIP interface attribute use DHCP mode</li> <li>3. User A calls user B</li> </ol>
<b>Expected result</b>	<ol style="list-style-type: none"> <li>1. In step 1,Check SIP interface ,can get signaling IP and media</li> <li>2. In step 2, The ATA users can hear a dial tone (<b>400 Hz continuous</b>).</li> </ol>

	3. In step 3, The call between User A calls user B is normal
<b>Test result</b>	
<b>Pass/Fail</b>	
<b>Date/Signature</b>	
<b>Remarks</b>	

### 2.10 Fax (Transparent Transmission Mode)

<b>Purpose</b>	To verify the FoIP function in the G711 pass-through mode of the fax service
<b>Test setup</b>	Same as figure1
<b>Prerequisite</b>	<ol style="list-style-type: none"> <li>1. Follow the network diagram to set up the test system.</li> <li>2. Make sure that the ATA works in the normal state and user can hear the dial tone.</li> </ol>
<b>Procedure</b>	<ol style="list-style-type: none"> <li>1. Fax machine A calls fax machine B and transmits a fax.</li> <li>2. Trace the message flow.</li> </ol>
<b>Expected result</b>	<ol style="list-style-type: none"> <li>1. In step 1, Fax transmits successfully.</li> <li>2. In step 2, The packet capture proves that the medium stream is transmitted in the G.711 coding scheme.</li> </ol>
<b>Test result</b>	
<b>Pass/Fail</b>	
<b>Date/Signature</b>	
<b>Remarks</b>	

### 2.11 Fax (T38)

<b>Purpose</b>	To verify that the ATA supports T38 faxing function
<b>Test setup</b>	Same as figure1
<b>Prerequisite</b>	<ol style="list-style-type: none"> <li>1. The devices are properly connected according to the network diagram.</li> <li>2. The VSP and the ATA are in the normal state.</li> <li>3. FAX A and FAX B connect to the two POTS ports A and B respectively.</li> </ol>
<b>Procedure</b>	<ol style="list-style-type: none"> <li>1. Configure fax parameter for T38</li> <li>2. Call FAX B from FAX A for a fax transmitting/receiving test.</li> </ol>
<b>Expected result</b>	1. In step 2, Trace the faxing call signaling, the coding/decoding mode is T38
<b>Test result</b>	
<b>Pass/Fail</b>	
<b>Date/Signature</b>	
<b>Remarks</b>	

### 2.12 Modem Service

<b>Purpose</b>	To verify the modem service supported by the ATA
<b>Test setup</b>	Same as figure1
<b>Prerequisite</b>	<ol style="list-style-type: none"> <li>1. The devices are properly connected according to the network diagram.</li> </ol>

	2. Make sure that the ATA works in the normal state and users can hear the correct dial tone
<b>Procedure</b>	1. Configure the modem service transmission mode as the transparent transmission mode. 2. Enable the automatic answering function on the called modem. Use the calling modem to call the called modem. 3. After the connection between the calling modem and the called is set up, transmit files between two PCs.
<b>Expected result</b>	1. In step 2, the calling modem can call the called modem successfully. 2. In step 3, the two PCs can transmit files to each other.
<b>Test result</b>	
<b>Pass/Fail</b>	
<b>Date/Signature</b>	
<b>Remarks</b>	

### 2.13 Basic Call (DTMF)

<b>Purpose</b>	To verify the capability of the ATA to support DTMF tones )
<b>Test setup</b>	Same as figure2
<b>Prerequisite</b>	1. The devices are properly connected according to the network diagram. 2. Make sure that the ATA works in the normal state and users can hear the correct dial tone
<b>Procedure</b>	1. Phone C makes an outbound call to an IVR number <b>0800 257 777 ( IRD IVR system)</b> 2. Phone C presses different keys to navigate through the IVR

	system
<b>Expected result</b>	1. The Called IVR system recognises the DTMF tones sent by the ATA
<b>Test result</b>	
<b>Pass/Fail</b>	
<b>Date/Signature</b>	
<b>Remarks</b>	

#### 2.14 Call Feature (Distinctive ringing)

<b>Purpose</b>	To verify the user distinctive ringing function by the ATA
<b>Test setup</b>	Same as figure2
<b>Prerequisite</b>	<ol style="list-style-type: none"> <li>1. The devices are properly connected according to the network diagram.</li> <li>2. Make sure that the ATA works in the normal state and the users can hear the correct dial tone</li> </ol>
<b>Procedure</b>	<ol style="list-style-type: none"> <li>1. Configure the user defined-ring ,map ring name to alert info</li> <li>2. User A calls user B, VSP send INVITE to B with Alert-info:&lt;http://127.0.0.0.1/ Bellcore-dr4&gt;</li> </ol>
<b>Expected result</b>	In step 2,User B can listen ring mode <b>(400 ms on, 800 ms off,400 ms on, 1400 ms off, and repeated )</b> Bellcore-DA4 cadence
<b>Test result</b>	
<b>Pass/Fail</b>	



<b>Date/Signature</b>	
<b>Remarks</b>	

### 2.15 Basic Call (Digitmap)

<b>Purpose</b>	To verify the digit configuration of the ATA
<b>Test setup</b>	Same as figure2
<b>Prerequisite</b>	<ol style="list-style-type: none"><li>1. The devices are properly connected according to the network diagram.</li><li>2. Make sure that the ATA works in the normal state and the users can hear the correct dial tones.</li><li>3. Make sure that the emergency services are notified that testing is being conducted.</li></ol>
<b>Procedure</b>	<ol style="list-style-type: none"><li>1. Make an out bound call to a local number.</li><li>2. Observe the delay</li><li>3. Make an outbound call to a national number</li><li>4. Observe the delay</li><li>5. Make an outbound call to emergency number</li><li>6. Observe the delay</li></ol>
<b>Expected result</b>	<ol style="list-style-type: none"><li>1. Invite message is sent straight away after the appropriate digits are punched in</li></ol>
<b>Test result</b>	
<b>Pass/Fail</b>	
<b>Date/Signature</b>	
<b>Remarks</b>	

**2.16 Call feature (Caller Line Identification Presentation)**

<b>Purpose</b>	To verify the function of presenting the calling line identification
<b>Test setup</b>	Same as figure1
<b>Prerequisite</b>	<ol style="list-style-type: none"> <li>1. The devices are properly connected according to the network diagram.</li> <li>2. The VSP and the ATA are in the normal state.</li> <li>3. MG interface data and user data are configured at the VSP side.</li> <li>4. The phone supports the Call ID.</li> </ol>
<b>Procedure</b>	<ol style="list-style-type: none"> <li>1. Configure the CLIP of phone A on the VSP.</li> <li>2. Do not configure any new service of phone B on the VSP.</li> <li>3. Phone B calls phone A.</li> </ol>
<b>Expected result</b>	In step 3, the phone A displays the phone number of phone B.
<b>Test result</b>	
<b>Pass/Fail</b>	
<b>Date/Signature</b>	
<b>Remarks</b>	

**2.17 Call feature (Caller Line Identification Restricted)**

<b>Purpose</b>	To verify the function of the calling line identification restriction
<b>Test setup</b>	Same as figure1
<b>Prerequisite</b>	<ol style="list-style-type: none"> <li>1. The devices are properly connected according to the network diagram.</li> </ol>



	<ol style="list-style-type: none"> <li>2. The VSP and the ATA are in the normal state.</li> <li>3. MG interface data and user data are configured at the VSP side.</li> <li>4. Phone A and Phone B support Call ID</li> </ol>
<b>Procedure</b>	<ol style="list-style-type: none"> <li>1. Configure the CLIP of phone A on the VSP.</li> <li>2. Configure the function of the CLIR of phone B on the VSP, while other new services are not configured.</li> <li>3. Phone B calls phone A.</li> </ol>
<b>Expected result</b>	<ol style="list-style-type: none"> <li>1. In step 3, the phone of phone A rings but cannot display the phone number of phone B. The calling line identification restriction is valid.</li> </ol>
<b>Test result</b>	
<b>Pass/Fail</b>	
<b>Date/Signature</b>	
<b>Remarks</b>	

### 2.18 Call feature (MWI)

<b>Purpose</b>	To verify the function of Message waiting indication
<b>Test setup</b>	Same as figure1
<b>Prerequisite</b>	<ol style="list-style-type: none"> <li>1. The devices are properly connected according to the network diagram.</li> <li>2. The VSP and the ATA are in the normal state.</li> <li>3. MG interface data and user data are configured at the VSP side.</li> <li>4. Telephone support message waiting indication function</li> </ol>

<b>Procedure</b>	<ol style="list-style-type: none"> <li>1. Configure the waiting indication function services of phone A on the VSP.</li> <li>2. Phone B leave message for phone A, Check phone A light state</li> <li>3. Phone A listens to message, Check phone A light state</li> </ol>
<b>Expected result</b>	<p>In step 2, phone A has the message light turn on and stutter dial tone (<b>400 Hz interrupted, 100 ms on, 100 ms off, repeated for 2.5 secs then continuous until it times out</b>) is heard when user picks up the phone</p> <p>In step 3 , the message light goes off and the stutter tone is cleared</p>
<b>Test result</b>	
<b>Pass/Fail</b>	
<b>Date/Signature</b>	
<b>Remarks</b>	

### 2.19 Dual Homing (SIP)

<b>Purpose</b>	To verify the dual homing function(SIP) of the ATA
<b>Test setup</b>	Same as figure1
<b>Prerequisite</b>	<ol style="list-style-type: none"> <li>1. Follow the network diagram to set up the test system.</li> <li>2. Make sure that the ATA works in the normal state</li> </ol>
<b>Procedure</b>	<ol style="list-style-type: none"> <li>1. Configure two available proxies with proxy1 as the primary one and proxy2 as the secondary one.</li> <li>2. Register the ATA with proxy1 successfully.</li> </ol>

	3. Disconnect the ATA and proxy1. 4. The ATA transmits messages to proxy2 to register.
<b>Expected result</b>	1. In step 1, The ATA can register with proxy2 successfully.
<b>Test result</b>	
<b>Pass/Fail</b>	
<b>Date/Signature</b>	
<b>Remarks</b>	

## 2.20 Additional VSP Feature testing (SIP)

<b>Purpose</b>	To verify additional testing that the VSP requires such as short codes etc
<b>Test setup</b>	Same as figure1
<b>Prerequisite</b>	As required by the VSP
<b>Procedure</b>	As required by the VSP
<b>Expected result</b>	As required by the VSP
<b>Test result</b>	
<b>Pass/Fail</b>	
<b>Date/Signature</b>	
<b>Remarks</b>	

### 3 Annex (normative): Test Completion Sheet

This section currently records the summary status of test case completion for the document. More details are required in order to use this page as a sign-off of testing completion.

Test case	Complete	Pass/Fail	Comments

### 4 History

Document history		
<Version>	<Date>	<Milestone>