



Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya Aura® Session Manager R7.0 and Avaya Aura® Communication Manager R7.0 to interoperate with Zenitel Turbine - Issue 1.0

Abstract

These Application Notes describe the configuration steps required for Zenitel Turbine to interoperate with Avaya Aura® Session Manager R7.0 and Avaya Aura® Communication Manager R7.0. The Zenitel Turbine is an IP Intercom that supports voice transmission using the Session Initiation Protocol (SIP).

Readers should pay attention to section 2, in particular the scope of testing as outlined in Section 2.1 as well as the observations noted in Section 2.2, to ensure that their own use cases are adequately covered by this scope and results.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the configuration steps required for Zenitel Turbine IP Intercom Substation to interoperate with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. The Zenitel Turbine IP Intercom Substations is a communicator that supports voice transmission using the Session Initiation Protocol (SIP) in harsh environments in sectors like Maritime, Oil&Gas, Heavy Industry, Transportation, Building security and Public safety. In the compliance testing, the Zenitel Turbine IP Intercom Substation was set up as a SIP user on Avaya Aura® Session Manager and underwent testing of various call scenarios with other Avaya telephones and Zenitel Turbine IP Intercom Substations.

2. General Test Approach and Test Results

The general test approach was to place calls to and from Stentonfon and exercise basic telephone operations. For serviceability testing, failures such as cable pulls and hardware resets were performed.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

2.1. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability testing. The feature testing was to verify that:

- TCIS 1-3, TCIS 4-5, TCIV-3/TCIV6, TFIS 1-2 and TMIS-1 models were tested.
- Successfully registers with Session Manager using IP address and FQDN.
- Turbine successfully establishes audio calls with Avaya H.323, SIP and digital endpoints registered to Session Manager and Communication Manager.
- Turbine successfully establishes audio calls with PSTN.
- Turbine IP successfully negotiates the appropriate audio codec.
- DTMF tones could be passed successfully to energize relay on Turbine unit and switch audio direction.
- Turbine successfully calls multiple destinations using a cover answer group.
- Turbine successfully calls a variety of endpoints in its call list.
- Correct handling of forwarded calls, cover paths and cover answer groups.
- Video was tested on the TCIV-3 model.

The serviceability testing focused on verifying the ability of Turbine to recover from adverse conditions, such as disconnecting/reconnecting the Ethernet cable to the devices and denying service on Session Manager.

2.2. Test Results

All test cases passed successfully.

2.3. Support

Technical support on Zenitel Turbine can be obtained through the following:

- **Phone:** +47 4000 2700
- **Web:** <https://www.zenitel.com/customer-service>

3. Reference Configuration

Figure 1 illustrates a test configuration that was used to compliance test the interoperability of Turbine with Session Manager and Communication Manager. The configuration consists of Communication Manager, System Manager and Session Manager. Communication Manager has connections to 96x1 IP (H323) deskphones. Session Manager has SIP registrations with, Turbine and 96x1 IP (SIP) deskphones. An ISDN-PRI trunk connects Communication Manager to the PSTN.

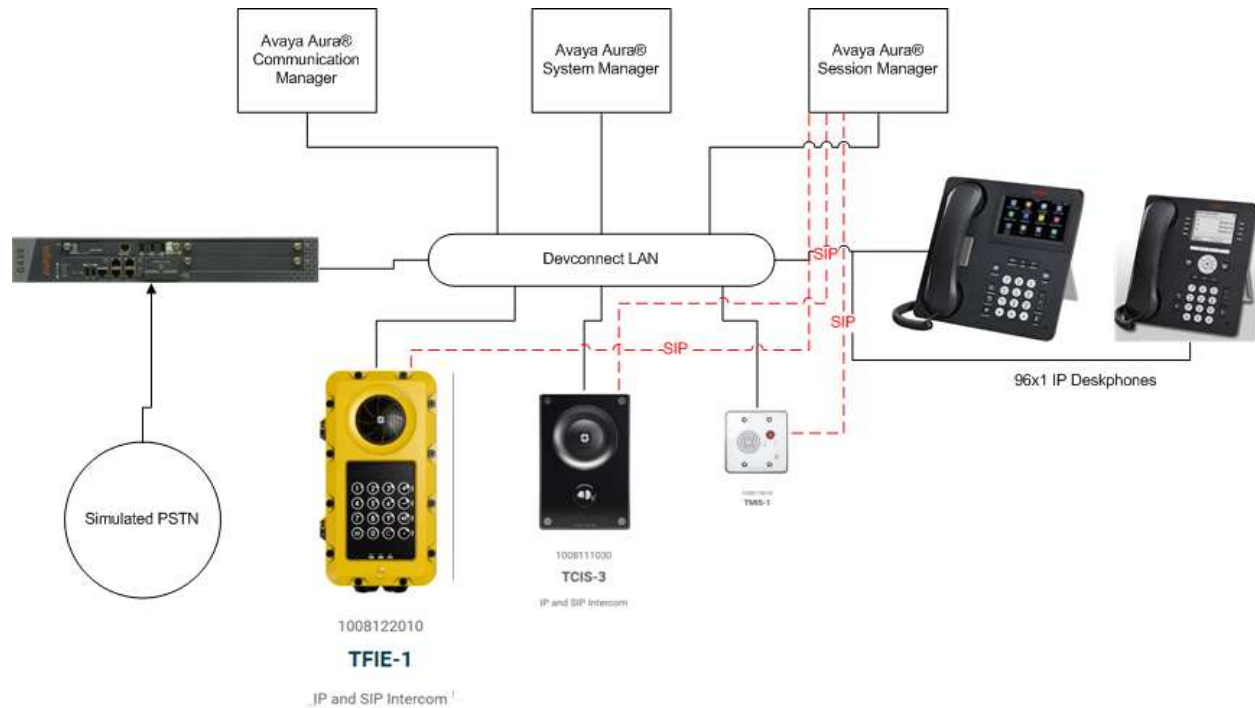


Figure 1: Avaya Aura® Session Manager and Avaya Aura® Communication Manager with Zenitel Turbine Configuration

4. Equipment and Software Validated

The following equipment and software were used for the sample configuration provided:

Equipment/Software	Release/Version
Avaya Aura® System Manager running on VMware Virtual Machine	R7.0.0.1 Build 7.0.0.0.16266-7.0.9.7001011 Software Update Revision 7.0.0.1.4212
Avaya Aura® Communication Manager running on VMware Virtual Machine	R7.0 SP1 Build 7.0.0.1.0.441.22477 Software Update Revision PLAT-rhel6.5-0010
Avaya G430 Media Gateway	37.21.0
Avaya Aura® Session Manager running on VMware Virtual Machine	R7.0 SP1 7.0.0.1.700102
Avaya 9611G IP Telephone Avaya 9641G IP Telephone	Release 6.6029 Release 6.6029 Release 7.0 Release 7.0
Zenitel Turbine	4.2.3.9

5. Configure Avaya Aura® Communication Manager

The configuration changes in this section for Communication Manager are performed through the Site Administration tool and via the System Manager web interface. Except where stated, the parameters in all steps are the default settings and are supplied for reference. For all other provisioning information such as provisioning of the trunks, call coverage, extensions, and voicemail, please refer to the Avaya product documentation in **Section 9**.

The procedures fall into the following areas:

- Configure IP Codec Set
- Configure SIP User
- Configure Endpoints for IP Video

5.1. Configure IP Codec Set

The IP Codec set must be configured with the codecs for use by IP endpoints. Enter the command **change ip-codec-set x** where **x** is the relevant codec set and set the **Audio Codec** to be used on **Page 1**. In the example below, codecs **G.722-64K** and **G.711A** are configured.

change ip-codec-set 1

Page 1 of 2

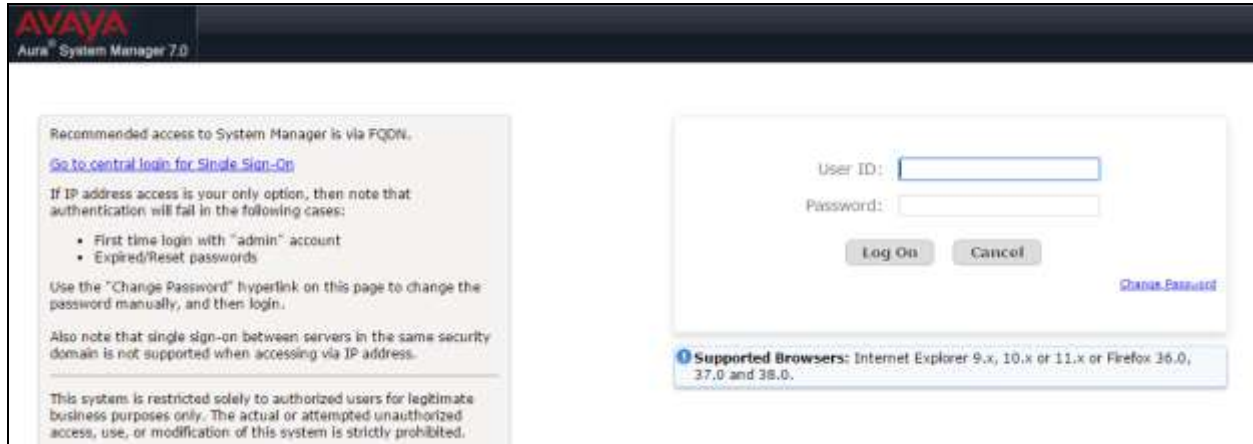
IP Codec Set

Codec Set: 1

	Audio	Silence	Frames	Packet
	Codec	Suppression	Per Pkt	Size(ms)
1:	G.722-64K		2	20
2:	G.711A	n	2	20
3:				
4:				
5:				
6:				
7:				

5.2. Configure SIP User

A SIP user must be added for each Turbine endpoint required. Navigate to the System Manager web interface, in this case <https://192.168.16.146/SMGR> and login with the relevant credentials.



The screenshot shows the Avaya Aura System Manager 7.0 login interface. On the left, there is a text box with instructions: "Recommended access to System Manager is via FQDN. Go to central login for Single Sign-On. If IP address access is your only option, then note that authentication will fail in the following cases: First time login with 'admin' account, Expired/Reset passwords. Use the 'Change Password' hyperlink on this page to change the password manually, and then login. Also note that single sign-on between servers in the same security domain is not supported when accessing via IP address. This system is restricted solely to authorized users for legitimate business purposes only. The actual or attempted unauthorized access, use, or modification of this system is strictly prohibited." On the right, there is a login form with fields for "User ID:" and "Password:", and buttons for "Log On" and "Cancel". A "Change Password" link is also present. At the bottom, a blue box lists "Supported Browsers: Internet Explorer 9.x, 10.x or 11.x or Firefox 36.0, 37.0 and 38.0."

From the Dashboard select **Users → User Management → Manage Users → New**



The screenshot shows the Avaya Aura System Manager 7.0 User Management interface. The top navigation bar includes "Home" and "User Management". The left sidebar lists "User Management", "Manage Users", "Public Contacts", "Shared Addresses", "System Presence", "ACLs", "Communication", "Profile Password", and "Policy". The main content area shows the "User Management" page with a search bar and a "Help" link. Below the search bar, there is a "Users" section with a table of users. The table has columns for "Last Name", "First Name", "Display Name", "Login Name", "SIP Handle", and "Last Login". The table shows 26 items, with a "Show 15" dropdown. There are also buttons for "Show", "Add", "Item", "Duplicate", "Delete", and "More Actions". A "Filter: Enable" link is visible on the right.

On the Identity tab enter an identifying **Last Name** and **First Name**, enter an appropriate **Login Name**, set **Authentication Type** to **Basic** and administer a password in the **Password** and **Confirm Password** fields.

The screenshot shows the 'New User Profile' form in the Avaya Aura System Manager 7.0 interface. The form is divided into four tabs: Identity, Communication Profile, Membership, and Contacts. The Identity tab is currently selected. The form contains the following fields:

- User Provisioning Rule:** A dropdown menu.
- Identity:**
 - Last Name:** Station
 - Last Name (Latin Translation):** Station
 - First Name:** New
 - First Name (Latin Translation):** New
 - Middle Name:**
 - Description:**
 - Login Name:** 8279999@devconnect.local
 - User Type:** Basic
 - Password:** (masked with asterisks)
 - Confirm Password:** (masked with asterisks)
 - Localized Display Name:**
 - Endpoint Display Name:**
 - Title:**

The form also includes a 'Commit & Continue' button and a 'Commit' button.

Click on the **Communication Profile** tab and enter and confirm a **Communication Profile Password**, this is used when logging in the SIP endpoint. and under **Communication Address** click **New**, select **Avaya SIP** from the **Type** drop down box and enter the **Fully Qualified Address** of the new SIP user. Click **Add** when done.

AVAYA
Aura® System Manager 7.0

Home / User Management / Manage Users

New User Profile

Identity * **Communication Profile** Membership Contacts

Communication Profile *

Communication Profile Password: *****

Confirm Password: *****

Communication Address *

Type	Handle	Domain
No Records found		
Type: Avaya SIP	Fully Qualified Address: 8279999	devconnect.local

Add Cancel

Continue to scroll down on the same page, enter the **Primary Session Manager, Origination Application Sequence, Termination Application Sequence** and **Home Location** relevant to the implementation.

Session Manager Profile

SIP Registration

* Primary Session Manager

SM71676

Secondary Session Manager

Survivability Server

Max. Simultaneous Devices

1

Block New Registration When
Maximum Registrations Active?

Primary	Secondary	Maximum
22	0	22

Application Sequences

Origination Sequence

CM1627_seq

Termination Sequence

CM1627_seq

Call Routing Settings

* Home Location

Devconnect

Conference Factory Set

(None)

Call History Settings

Enable Centralized Call History?

Scroll down the page to the **CM Endpoint Profile** section. Select the Communication Manager system from the **System** drop down box, select **Endpoint** as the **Profile Type**, enter the **Extension** number you wish to use, select **9620SIP_DEFAULT_CM_7_0** as the **Template** and ensure **IP** is configured as the **Port**, click Commit (not shown) when done. Repeat this for every SIP extension required.

☒ **CM Endpoint Profile** ▼

* System

CM71627 ▼

* Profile Type

Endpoint ▼

Use Existing Endpoints

☐

* Extension

8279999

Endpoint Editor

* Template

9620SIP_DEFAULT_CM_7_0 ▼

Set Type

9620SIP

Security Code

••••••

Port

IP

Voice Mail Number

Preferred Handle

8279999@devconnect.local ▼

Calculate Route Pattern

☐

Sip Trunk

aar

Enhanced Callr-Info display for 1-line phones

☐

Delete Endpoint on Unassign of Endpoint from User or on Delete User.

☒

Override Endpoint Name and Localized Name

☒

Allow H.323 and SIP Endpoint Dual Registration

☐

6. Configure Zenitel Turbine

The following steps detail the configuration for Turbine using the Web Interface. The steps include the following areas:

- Launch Web Interface
- Administer SIP Settings
- Configure Direct Access Key

6.1. Launch Web Interface

Access the Turbine web interface, enter **http://<ipaddress>** in an Internet browser window, where **<ipaddress>** is the IP address of Turbine. Log in with the appropriate credentials. The **IP-StationWeb** screen is shown.

The screenshot displays the VINGTOR STENTOFON WEB CONFIGURATION interface. The top navigation bar includes tabs for Station Main, SIP Configuration, Station Administration, Advanced SIP, and Advanced Network. The left sidebar shows a tree view with 'Station Information' expanded and 'Main Settings' as a sub-option. The main content area is divided into two sections: 'Station Information' and 'Station Status'.

Description	Information
Station IP:	10.10.16.131
Subnet Mask:	255.255.255.0
Default Gateway:	10.10.16.1
DNS Server 1:	10.10.16.10
DNS Server 2:	
Hardware Type:	8121
Hardware Version:	1
Software Versions:	List
Image Package Version:	4.2.3.9 (sti)
MAC Address:	00:13:cb:08:05:7a
System Model Name:	Stentofon Turbine Compact - Mini
Hardware Revision:	0000
Kernel Version:	3.10.0[st_develop_ab78065]+ #6 PREEMPT Wed Apr 13 12:06:43 CEST 2016
Devicetree Version:	01
Boot/Environment Version:	2015.04.30/2015.04.21

Description	Status
Station Mode:	SIP
Display Name:	TMIS1
Directory Number (SIP ID):	8279999
Server Domain (SIP):	devconnect.local, Registered - Thu Jan 1 19:19:13 1970
Backup Domain (SIP):	
Backup Domain 2 (SIP):	
Outbound Proxy:	10.10.16.77

6.2. Administer SIP Settings

Select **Main Settings** from the left menu and select **Use SIP**. From the **Model:** drop down menu choose **TCIS 1-3, TCIS 4-5, TCIV-3/TCIV6, TFIS 1-2** or **Mini (TMIS-1)** depending on the model tested. click **Save** when done. A screen will appear (not shown) to confirm the setting, click Apply and Turbine will reboot.

The screenshot displays the 'SIP Configuration' page of a web interface. The top navigation bar includes 'Station Main', 'SIP Configuration', 'Station Administration', 'Advanced SIP', and 'Advanced Network'. The left sidebar shows 'Station Information' and 'Main Settings'. The main content area is divided into three sections: 'Station Mode' with radio buttons for 'Use Alphacom', 'Use Exigo', 'Use SIP' (selected), 'Use Pulse', and 'Use Pulse Server'; 'Product Model And Accessory' with a 'Model:' dropdown set to 'Mini (TMIS-1)'; and 'IP Settings' with radio buttons for 'DHCP' (selected) and 'Static IP'. Below these are input fields for IP address, subnet mask, gateway, and DNS servers, each split into four boxes. The IP address is 10.5.11.19, subnet mask is 255.255.255.0, gateway is 10.5.11.1, and both DNS servers are 0.0.0.0. The 'Hostname' field contains 'zenitel08057a' and the 'Read IP Address' checkbox is checked. A 'Save' button is at the bottom.

Field	Box 1	Box 2	Box 3	Box 4
IP-address:	10	5	11	19
Subnet-mask:	255	255	255	0
Gateway:	10	5	11	1
DNS Server 1:	0	0	0	0
DNS Server 2:	0	0	0	0

Hostname: zenitel08057a

Read IP Address: ☒

Save

Click on **SIP Configuration** → **SIP Settings** and configure the following in the **Account Settings** section:

- **Display name:** Enter the desired name.
- **Directory Number (SIP ID):** Enter a user extension administered from **Section 5.2**.
- **Server Domain (SIP):** Enter the Domain of Session Manager.
- **Authentication User Name:** Enter a user extension administered from **Section 5.2**.
- **Authentication Password:** Enter the **Communication Profile Password** from **Section 5.2**.
- **Outbound Proxy (optional):** Enter the IP address of Session Manager and **5060** as the **Port**.

Station Main	SIP Configuration	Station Administration	Advanced SIP	Advanced Network																										
<div> <div> <div>▼ SIP Settings</div> <div> <div>▶ Audio Settings</div> <div>▶ Direct Access Key Settings</div> <div>▶ Relay Settings</div> <div>▶ Time Settings</div> <div>▶ I/O Settings</div> <div>▶ Keyboard Settings</div> <div>▶ Script Configuration</div> <div>▶ Script Events</div> <div>▶ Script Upload</div> </div> </div> <div> <h3>Account Settings</h3> <table> <thead> <tr> <th>Description</th> <th>Configuration</th> </tr> </thead> <tbody> <tr> <td>Display Name:</td> <td>TMIS1</td> </tr> <tr> <td>Directory Number (SIP ID):</td> <td>8279999</td> </tr> <tr> <td>Server Domain (SIP):</td> <td>devconnect.local</td> </tr> <tr> <td>Backup Domain (SIP):</td> <td></td> </tr> <tr> <td>Backup Domain 2 (SIP):</td> <td></td> </tr> <tr> <td>Registration Method:</td> <td>Parallel ▼</td> </tr> <tr> <td>Authentication User Name:</td> <td>8279999</td> </tr> <tr> <td>Authentication Password:</td> <td>*****</td> </tr> <tr> <td>Register Interval:</td> <td>600 (Minimum 60 seconds)</td> </tr> <tr> <td>Outbound Proxy [optional]:</td> <td>10.10.16.77 Port: 5060</td> </tr> <tr> <td>Outbound Backup Proxy [optional]:</td> <td>Port: 5060</td> </tr> <tr> <td>Outbound Backup Proxy 2 [optional]:</td> <td>Port: 5060</td> </tr> </tbody> </table> </div> </div>					Description	Configuration	Display Name:	TMIS1	Directory Number (SIP ID):	8279999	Server Domain (SIP):	devconnect.local	Backup Domain (SIP):		Backup Domain 2 (SIP):		Registration Method:	Parallel ▼	Authentication User Name:	8279999	Authentication Password:	*****	Register Interval:	600 (Minimum 60 seconds)	Outbound Proxy [optional]:	10.10.16.77 Port: 5060	Outbound Backup Proxy [optional]:	Port: 5060	Outbound Backup Proxy 2 [optional]:	Port: 5060
Description	Configuration																													
Display Name:	TMIS1																													
Directory Number (SIP ID):	8279999																													
Server Domain (SIP):	devconnect.local																													
Backup Domain (SIP):																														
Backup Domain 2 (SIP):																														
Registration Method:	Parallel ▼																													
Authentication User Name:	8279999																													
Authentication Password:	*****																													
Register Interval:	600 (Minimum 60 seconds)																													
Outbound Proxy [optional]:	10.10.16.77 Port: 5060																													
Outbound Backup Proxy [optional]:	Port: 5060																													
Outbound Backup Proxy 2 [optional]:	Port: 5060																													

In the **Call Settings** section, configure as required the **DTMF Method** as **SIP INFO** or RFC 2833 (not shown), this allows DTMF tones to be either sent in-band or using SIP INFO messaging. Configure other options as required.

Call Settings

Description	Configuration
Enable Auto Answer:	<input type="checkbox"/>
Auto Answer Delay:	<input type="text" value="0"/> seconds. Max 30 seconds.
Delay Call Setup:	<input type="text" value="0"/> seconds. Max 60 seconds. Only for Input Buttons.
Disable Disconnect By Button:	<input type="checkbox"/>
Overlap dialing:	<input type="checkbox"/>
DTMF method:	<input type="text" value="SIP INFO"/>
Call LED Off During Ringing:	<input type="checkbox"/>
RTP Timeout value:	<input type="text" value="0"/> seconds. 0 = RTP Timeout Disabled.
IP Heavy Duty:	<input type="checkbox"/>
Choose Relay To Configure:	<input type="text" value="Relay 1"/>

In the **Relay 1 Settings** section select a digit from the drop down box for **Remote Digit for Timed Relay On**. When this digit is pushed by a called party, the relay in the Turbine will be energized. Retain the default values for the remaining fields. Click **Save** when done. A screen will appear (not shown) to confirm the setting, click Reboot and Turbine will reboot

Relay 1 Settings

Description	Configuration
Remote Digit For Relay On:	<input type="text" value="-"/>
Remote Digit For Relay Off:	<input type="text" value="-"/>
Remote Digit For Relay Slow Flash :	<input type="text" value="-"/>
Remote Digit For Relay Fast Flash:	<input type="text" value="-"/>
Remote Digit For Relay Toggle:	<input type="text" value="-"/>
Remote Digit For Timed Relay On:	<input type="text" value="6"/>
Timed Relay Duration:	<input type="text" value="3"/> seconds.
Outgoing Ringing:	<input type="text" value="-"/>
Incoming Ringing:	<input type="text" value="-"/>
Outgoing Call:	<input type="text" value="-"/>
Incoming Call:	<input type="text" value="-"/>
Group Call (Pulse mode only):	<input type="text" value="-"/>
Idle:	<input type="text" value="-"/>
Error:	<input type="text" value="-"/>

Save

6.3. Configure Direct Access Key

Select **SIP Configuration** → **Direct Access Key Settings** from the left menu and select **Direct Access Key 1** to configure it. In the **Value** field enter the extension to be called when the **Direct Access Key 1** is pushed. Select **Ringlist 1** under **Option**. Select **End Call** under **Function(In Call)** in the Direct Access Key Settings (In Call) section for **Direct Access Key 1**. Configure the **Ringlist Settings** and enter an extension number to be dialed in the **Value 1**. In this example **Ringlist 1** is configured to call **8270002**.

Station Main

SIP Configuration

Station Administration

Advanced SIP

Advanced Network

▶ SIP Settings

▶ Audio Settings

▼ Direct Access Key Settings

▶ Relay Settings

▶ Time Settings

▶ I/O Settings

▶ Keyboard Settings

▶ Script Configuration

▶ Script Events

▶ Script Upload

Direct Access Key Settings

	Function (idle)	Value	Option
Direct Access Key 1	Call To ▼	8270002	Ringlist 1 ▼
Input 1	Call To		Unused ▼
Input 2	Call To		Unused ▼
Input 3	Call To		Unused ▼
Input 4	Call To		Unused ▼
Input 5	Call To		Unused ▼
Input 6	Call To		Unused ▼

Save

Direct Access Key Settings (In Call)

	Function (in call)	Activated	Deactivated
Direct Access Key 1	End Call ▼		
Input 1	Do Nothing ▼		
Input 2	Do Nothing ▼		
Input 3	Do Nothing ▼		
Input 4	Do Nothing ▼		
Input 5	Do Nothing ▼		
Input 6	Do Nothing ▼		

Save

Ringlist Settings

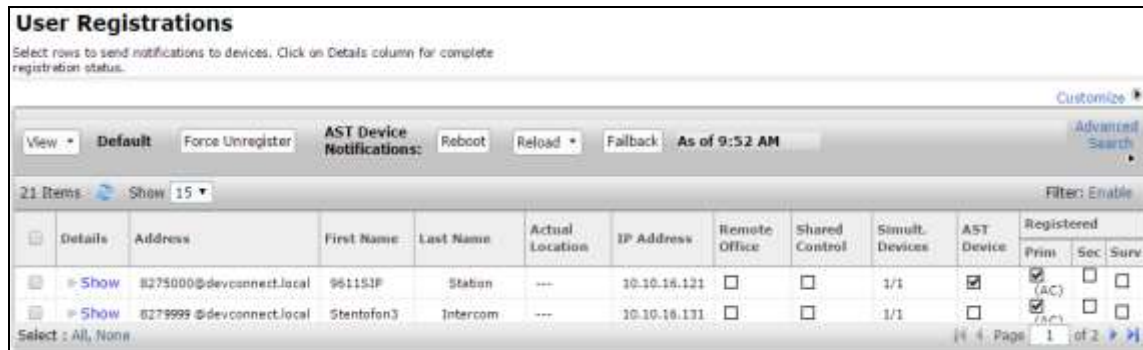
	Ringlist 1	With Previous	Ringlist 2	With Previous	Ringlist 3	With Previous
Value 1	8270002	<input type="checkbox"/>		<input type="checkbox"/>		<input type="checkbox"/>

7. Verification Steps

This section provides the tests that can be performed to verify correct configuration of Session Manager and Turbine.

7.1. Verify Avaya Aura® Session SIP Endpoint Registration

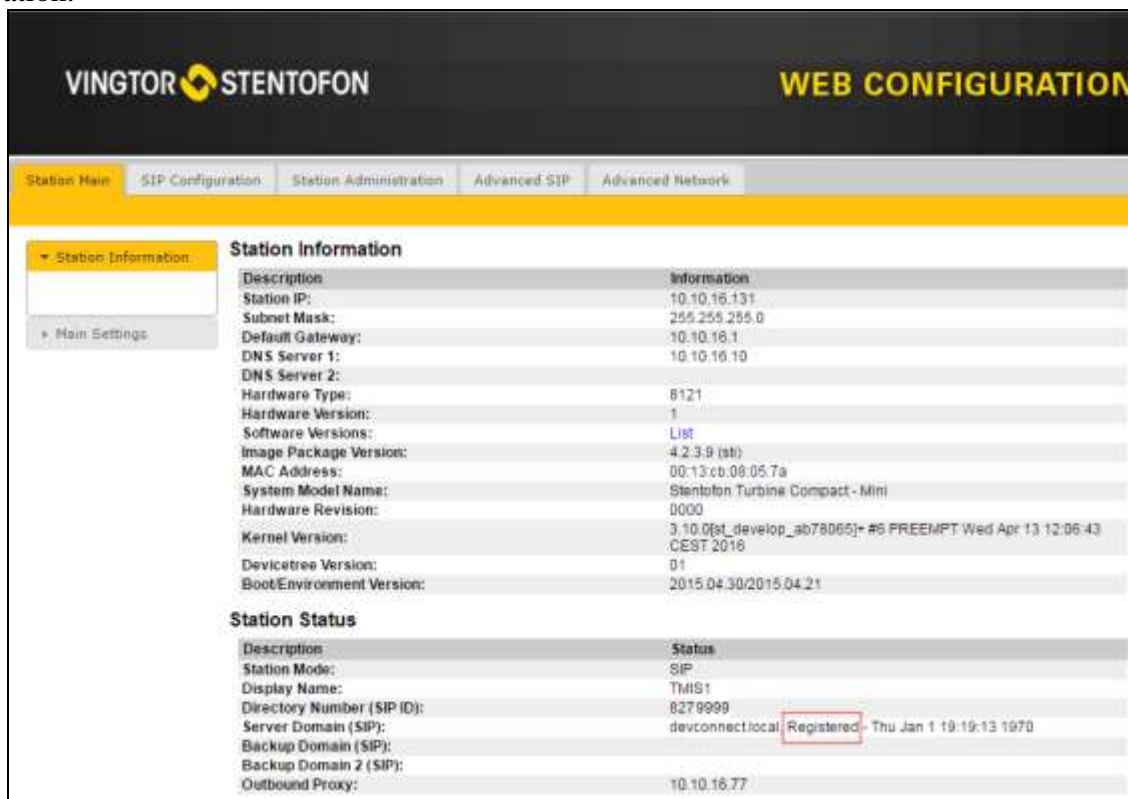
From the System Manager web interface click **Session Manager** → **System Status** → **User Registrations**. Verify that Turbine endpoints are successfully registered as shown below.



Details	Address	First Name	Last Name	Actual Location	IP Address	Remote Office	Shared Control	Simult. Devices	AST Device	Registered
Show	8275000@devconnect.local	961153P	Station	---	10.10.16.131	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/> (AC)
Show	8279999@devconnect.local	Stentofon3	Intercom	---	10.10.16.131	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/> (AC)

7.2. Verify Turbine SIP Registration

From the Stentofon web interface, select **Information** from the left menu. Verify that the **Registration state** shows **Registered**. Place a call to another endpoint to verify basic call operation.



Description	Information
Station IP:	10.10.16.131
Subnet Mask:	255.255.255.0
Default Gateway:	10.10.16.1
DNS Server 1:	10.10.16.10
DNS Server 2:	
Hardware Type:	8121
Hardware Version:	1
Software Versions:	List
Image Package Version:	4.2.3.9 (stb)
MAC Address:	00:13:c0:08:05:7a
System Model Name:	Stentofon Turbine Compact - Mini
Hardware Revision:	0000
Kernel Version:	3.10.0[st_develop_ab78065]-#6 FREEMPT Wed Apr 13 12:06:43 CEST 2016
Devicetree Version:	01
Boot/Environment Version:	2015.04.30/2015.04.21

Description	Status
Station Mode:	SIP
Display Name:	THIS1
Directory Number (SIP ID):	8279999
Server Domain (SIP):	devconnect.local Registered Thu Jan 1 19:19:13 1970
Backup Domain (SIP):	
Backup Domain 2 (SIP):	
Outbound Proxy:	10.10.16.77

7.3. Verify Successful Calls

Place a call to and from the Turbine endpoint. Verify 2-way audio is heard and validate call terminates successfully.

8. Conclusion

These Application Notes describe the configuration steps required for configuring Zenitel Turbine to interoperate with Avaya Aura® Session Manager. All feature and serviceability tests were completed successfully with observations made in **Section 2.2**.

9. Additional References

This section references the Avaya and Zenitel product documentation that are relevant to these Application Notes.

The following Avaya product documentation can be found at <http://support.avaya.com>.

[1] *Administering Avaya Aura® Session Manager Release 7.0, August 2015*

The Zenitel Turbine documentation can be found at <http://www.zenitel.com>.

[2] *A100K11013-Pulse-Getting-Started.pdf*.

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