



Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya Aura® Session Manager R7.1 and Avaya Aura® Communication Manager R7.1 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 R7.1 and Avaya Aura® Communication Manager R7.1. 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 Turbine 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.

Avaya recommends our customers implement Avaya solutions using appropriate security and encryption capabilities enabled by our products. The testing referenced in these DevConnect Application Notes included the enablement of supported encryption capabilities in the Avaya products. Readers should consult the appropriate Avaya product documentation for further information regarding security and encryption capabilities supported by those Avaya products.

Support for these security and encryption capabilities in any non-Avaya solution component is the responsibility of each individual vendor. Readers should consult the appropriate vendor-supplied product documentation for more information regarding those products.

For the testing associated with these Application Notes, the interface between Avaya systems and the Zenitel Turbine IP Intercoms utilized enabled capabilities of TLS/SRTP.

2.1. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability testing. TCIS 1-3, TCIS 4-5, TCIV-3/TCIV6, TFIE 1-2 and TMIS-1 models were tested. The feature testing was to verify that:

- Turbine successfully registers with Session Manager using IP address and FQDN using UDP, TCP and TLS
- Turbine successfully establishes audio calls with good quality RTP and SRTP audio to 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 (H.323) deskphones. Session Manager has SIP registrations with, Turbine and 96x1 IP (SIP) deskphones. An ISDN-PRI trunk connects Communication Manager to the PSTN.

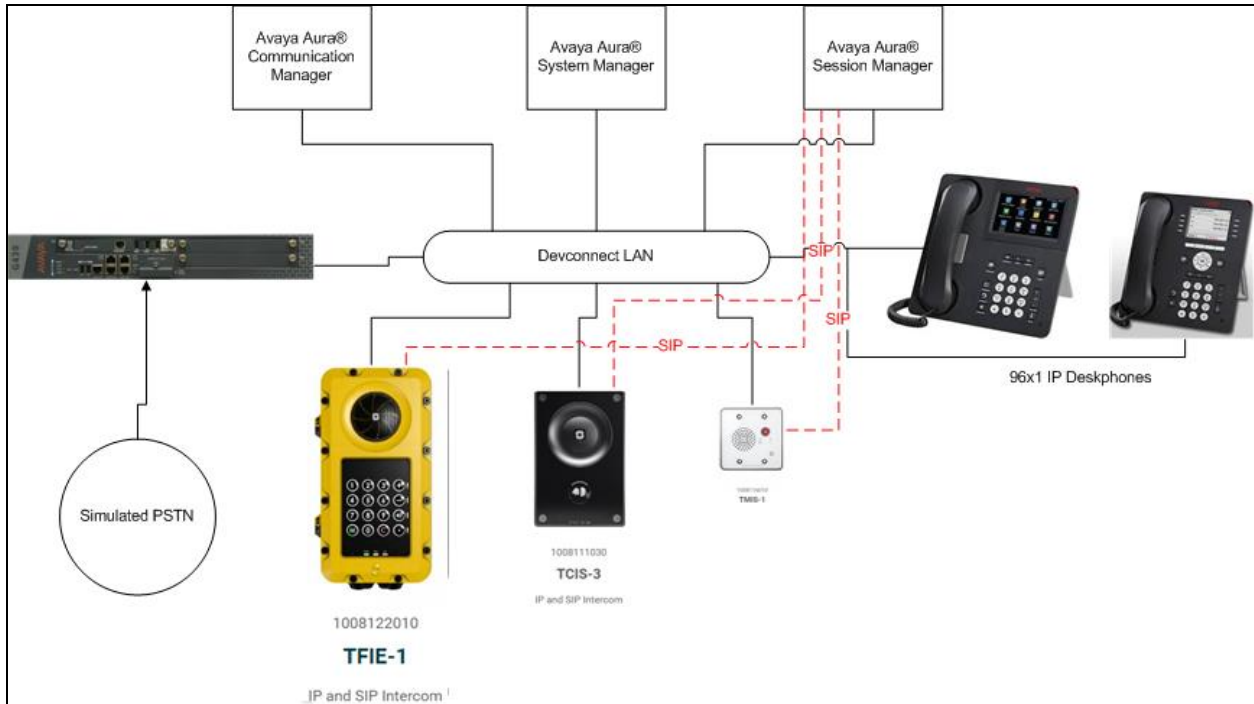


Figure 1: Avaya Aura® Session Manager and Avaya Aura® Communication Manager with Zenitel Turbine Configuration

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	R7.1.1.0 Build 7.1.1.0.1125193 Software Update Revision 7.1.1.0.046931 Feature Pack 1
Avaya Aura® Communication Manager running on VMware Virtual Machine	R017x.01.0.532.0 Build 7.1.1.0.0.532.23955 Software Update Revision RHEL7.2-SSP001 KERNEL-3.10.0
Avaya G430 Media Gateway <ul style="list-style-type: none">• MGP	38.20.1
Avaya Aura® Media Server	7.8.0.309
Avaya Aura® Session Manager running on VMware Virtual Machine	R7.0 SP1 7.1.1.0.711008
Avaya 9611G IP Telephone Avaya 9641G IP Telephone	Release 6.6229 H323 Release 6.6229 H323 Release 7.1.0.1.1 SIP Release 7.1.0.11 SIP
Zenitel Turbine	4.7.3

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

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.711MU** and **G.711A** are configured. Media Encryption (**1-srtp-aescm128-hmac80**) was set for this test with **Encrypted SRTP: best-effort**.

```
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.711MU      n          2         20
2: G.711A      n          2         20
3:
4:
5:
6:
7:

Media Encryption                               Encrypted SRTP: best-effort
1: 1-srtp-aescm128-hmac80
2:
3:
```

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://<IP Address>/SMGR> and log in with the relevant credentials.

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.

User ID:

Password:

[Change Password](#)

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**

Home / Users / User Management / Manage Users

Search [Help ?](#)

User Management

[Advanced Search](#)

26 Items 15

<input type="checkbox"/>	Last Name	First Name	Display Name	Login Name	SIP Handle	Last Login
--------------------------	-----------	------------	--------------	------------	------------	------------

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.

AVAYA
Aura System Manager 7.0

Last Logged on at May 13, 2017 10:00 AM
GO... Log

Home User Management x

Home / Users / User Management / Manage Users

New User Profile Commit & Continue Commit

Identity * Communication Profile Membership Contacts

User Provisioning Rule

User Provisioning Rule: []

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: []

Confirm Password: []

Localized Display Name: []

Endpoint Display Name: []

Title: []

Click on the **Communication Profile** tab and enter and confirm a **Communication Profile Password**, this is used when logging in the SIP endpoint. 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.

The screenshot shows the 'New User Profile' configuration page in the Avaya Aura System Manager 7.0. The page is divided into several sections:

- Navigation:** A sidebar on the left contains 'User Management' with sub-items: 'Manage Users', 'Public Contacts', 'Shared Addresses', 'System Presence', 'ACLs', 'Communication', 'Profile Password', and 'Policy'. The main breadcrumb is 'Home / Users / User Management / Manage Users'.
- Page Header:** 'AVAYA Aura System Manager 7.0' and 'Last Logged on at May 13, 2016 3:10 PM'. There are 'GO...' and 'Log off' buttons.
- Form Tabs:** 'Identity', 'Communication Profile', 'Membership', and 'Contacts'. The 'Communication Profile' tab is active.
- Communication Profile Section:** Contains fields for 'Communication Profile Password' and 'Confirm Password', both masked with dots. Below is a 'Name' section with a 'New' button, a table with one row 'Primary', and a 'Default' checkbox checked.
- Communication Address Section:** Contains a 'New' button, an 'Edit' button, and a 'Delete' button. Below is a table with columns 'Type', 'Handle', and 'Domain'. The table is currently empty with the text 'No Records found'. Below the table, there are fields for 'Type' (set to 'Avaya SIP'), 'Fully Qualified Address' (set to '8279999'), and 'Domain' (set to 'devconnect.local'). There are 'Add' and 'Cancel' buttons at the bottom right.

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

Primary	Secondary	Maximum
22	0	22

Secondary Session Manager

Survivability Server

Max. Simultaneous Devices ▼

Block New Registration When Maximum Registrations Active?

Application Sequences

Origination Sequence ▼

Termination Sequence ▼

Call Routing Settings

* Home Location ▼

Conference Factory Set ▼

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, 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 ▼

* Profile Type ▼

Use Existing Endpoints

* Extension

* Template ▼

Set Type

Security Code

Port

Voice Mail Number

Preferred Handle ▼

Calculate Route Pattern

Sip Trunk

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
- Add Root Certificate
- 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 IP-StationWeb web interface. At the top, there is a navigation bar with tabs for 'Station Main', 'SIP Configuration', 'Station Administration', 'Advanced SIP', and 'Advanced Network'. The 'Station Main' tab is active. On the left side, there is a sidebar with a 'Station Information' dropdown menu and a 'Main Settings' button. The main content area is titled 'TFIE-1 Information' and contains two tables.

Description	Information
Station IP:	10.10.16.102
Subnet Mask:	255.255.255.0
Default Gateway:	10.10.16.1
DNS Server 1:	10.10.16.10
DNS Server 2:	
Hardware Type:	8124
Hardware Version:	1
Software Versions:	List
Image Package Version:	4.7.3.0 (sti)
MAC Address:	00:13:cb:0d:10:1f
System Model Name:	Vingtor-Stentofon Turbine Extended - Industrial
Hardware Revision:	0004
Kernel Version:	3.10.0[release/intercom4.7_27e5eb5]+ #1 PREEMPT Tue Oct 24 16:15:51 CEST 2017
Devicetree Version:	06
Boot/Environment Version:	2016.02.05/2017.05.19

Description	Status
Station Mode:	SIP
Display Name:	TFIE-1
Directory Number (SIP ID):	8279999
Server Domain (SIP):	devconnect.local, Registered - Thu Jan 1 18:27:00 1970
Backup Domain (SIP):	
Backup Domain 2 (SIP):	
Outbound Proxy:	10.10.16.77

6.2. Add Session Manager root Certificate

Select **SIP Configuration** tab and from the left hand menu select **Certificates**. The Turbine certificates are listed. Click on the **Choose file** and browse to the location of the root certificate .pem file. When selected, click on the **Upload** button.

The screenshot shows the SIP Configuration interface. The top navigation bar includes 'Station Main', 'SIP Configuration' (selected), 'Station Administration', 'Advanced SIP', and 'Advanced Network'. On the left, a menu lists various settings, with 'Certificates' selected. The main content area is titled 'Certificates' and contains a table with the following data:

	Name	
Certificate 1	turbine_server_sha256.key	Delete
Certificate 3	turbine_server_sha1.key	Delete

Below the table is the 'Upload Certificate' section, which includes a 'Choose File' button, the text 'SystemManagerCA.pem', and an 'Upload' button.

The root certificate is uploaded and is shown in the list.

The screenshot shows the SIP Configuration interface after the root certificate has been uploaded. The top navigation bar and left menu are the same as in the previous screenshot. The 'Certificates' table now includes an additional entry:

	Name	
Certificate 1	turbine_server_sha256.key	Delete
Certificate 2	SystemManagerCA.pem	Delete
Certificate 3	turbine_server_sha1.key	Delete

The 'Upload Certificate' section remains visible below the table.

6.3. 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, TFIE 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 shows a web interface for configuring SIP settings. The top navigation bar includes 'Station Main', 'SIP Configuration', 'Station Administration', 'Advanced SIP', and 'Advanced Network'. The left sidebar has 'Station Information' and 'Main Settings'. The main content area is titled 'Station Mode' and includes radio buttons for 'Use Alphacom', 'Use Exigo', 'Use SIP' (selected), 'Use Pulse', and 'Use Pulse Server'. Below this is the 'Product Model And Accessory' section with a 'Model:' dropdown set to 'TFIE-1' and an 'Accessory:' dropdown set to 'No accessory'. The 'IP Settings' section has radio buttons for 'DHCP' and 'Static IP' (selected). It contains a table for IP configuration with fields for IP-address, Subnet-mask, Gateway, DNS Server 1, and DNS Server 2, each with four input boxes. The 'Hostname:' field is set to 'zenitel0d101f'. There are three checkboxes: 'Disable Reset to Factory default settings using frontboard and I/O:' (unchecked), 'Read IP Address:' (checked), and 'Enable RSTP:' (unchecked). A 'Save' button is at the bottom.

Field	Input 1	Input 2	Input 3	Input 4
IP-address:	10	5	11	185
Subnet-mask:	255	255	0	0
Gateway:	169	254	1	1
DNS Server 1:	0	0	0	0
DNS Server 2:	0	0	0	0

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** for UDP/TCP.

Station Main		SIP Configuration		Station Administration		Advanced SIP		Advanced Network																																																										
<div style="display: flex;"> <div style="width: 20%; border: 1px solid #ccc; padding: 5px;"> <ul style="list-style-type: none"> ▼ SIP Settings ▶ Audio Settings ▶ Direct Access Key Settings ▶ Relay Settings ▶ Time Settings ▶ I/O Settings ▶ Keyboard Settings ▶ Script Configuration ▶ Script Events ▶ Script Upload ▶ Audio Messages ▶ Certificates </div> <div style="width: 80%; padding: 5px;"> <h3>Account Settings</h3> <table border="1"> <thead> <tr> <th>Description</th> <th colspan="2">Configuration</th> </tr> </thead> <tbody> <tr> <td>Display Name:</td> <td colspan="2">TFIE-1</td> </tr> <tr> <td>Directory Number (SIP ID):</td> <td colspan="2">8279999</td> </tr> <tr> <td>Server Domain (SIP):</td> <td colspan="2">devconnect.local</td> </tr> <tr> <td>Backup Domain (SIP):</td> <td colspan="2"></td> </tr> <tr> <td>Backup Domain 2 (SIP):</td> <td colspan="2"></td> </tr> <tr> <td>Registration Method:</td> <td colspan="2">Parallell ▼</td> </tr> <tr> <td>Authentication User Name:</td> <td colspan="2">8279999</td> </tr> <tr> <td>Authentication Password:</td> <td colspan="2">*****</td> </tr> <tr> <td>Register Interval:</td> <td>600</td> <td>(min. 60 seconds)</td> </tr> <tr> <td>Outbound Proxy [optional]:</td> <td>10.10.16.35</td> <td>Port: 5060</td> </tr> <tr> <td>Outbound Backup Proxy [optional]:</td> <td></td> <td>Port: 5060</td> </tr> <tr> <td>Outbound Backup Proxy 2 [optional]:</td> <td></td> <td>Port: 1</td> </tr> <tr> <td>Outbound Transport:</td> <td colspan="2">UDP ▼</td> </tr> <tr> <td>SIP Scheme:</td> <td colspan="2">sip ▼ Using sips forces all proxies to also use TLS</td> </tr> <tr> <td>RTP Encryption:</td> <td colspan="2">disabled ▼</td> </tr> <tr> <td>SRTP Crypto Type:</td> <td colspan="2">AES_CM_128_HMAC_SHA1_80 ▼</td> </tr> <tr> <td>Use Unencrypted SRTPC:</td> <td colspan="2"><input type="checkbox"/></td> </tr> <tr> <td>TLS Private Key:</td> <td colspan="2">turbine_server_sha256.key ▼</td> </tr> </tbody> </table> </div> </div>										Description	Configuration		Display Name:	TFIE-1		Directory Number (SIP ID):	8279999		Server Domain (SIP):	devconnect.local		Backup Domain (SIP):			Backup Domain 2 (SIP):			Registration Method:	Parallell ▼		Authentication User Name:	8279999		Authentication Password:	*****		Register Interval:	600	(min. 60 seconds)	Outbound Proxy [optional]:	10.10.16.35	Port: 5060	Outbound Backup Proxy [optional]:		Port: 5060	Outbound Backup Proxy 2 [optional]:		Port: 1	Outbound Transport:	UDP ▼		SIP Scheme:	sip ▼ Using sips forces all proxies to also use TLS		RTP Encryption:	disabled ▼		SRTP Crypto Type:	AES_CM_128_HMAC_SHA1_80 ▼		Use Unencrypted SRTPC:	<input type="checkbox"/>		TLS Private Key:	turbine_server_sha256.key ▼	
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- **Outbound Proxy (optional):** Enter the IP address of Session Manager and **5061** as the **Port** for TLS
- **SIP Scheme:** Choose **sips** from the drop down.
- **RTP Encryption:** Select **srtp_encryption** from the drop down.

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

Description	Configuration
Remote Digit For Relay On:	- ▾
Remote Digit For Relay Off:	- ▾
Remote Digit For Relay Slow Flash :	- ▾
Remote Digit For Relay Fast Flash:	- ▾
Remote Digit For Relay Toggle:	- ▾
Remote Digit For Timed Relay On:	6 ▾
Timed Relay Duration:	3 seconds.
Outgoing Ringing:	- ▾
Incoming Ringing:	- ▾
Outgoing Call:	- ▾
Incoming Call:	- ▾
Group Call (Pulse mode only):	- ▾
Idle:	- ▾
Error:	- ▾

6.4. Configure Direct Access Key

Select **SIP Configuration** → **Direct Access Key Settings** from the left menu and select **DAK 1** to configure it. In the **Idle** field select **Call To** from the drop down and enter the extension to be called when the **DAK 1** key is pushed. In the **Call** field select **Answer/End Call** and **On Key Press**.

Station Main				SIP Configuration	Station Administration	Advanced SIP	Advanced Network																																																																																				
<div style="display: flex;"> <div style="width: 20%; border: 1px solid #ccc; padding: 5px;"> <ul style="list-style-type: none"> ▶ SIP Settings ▶ Audio Settings <li style="background-color: #f0f0f0;">▶ Direct Access Key Settings ▶ Relay Settings ▶ Time Settings ▶ I/O Settings ▶ Keyboard Settings ▶ Script Configuration ▶ Script Events ▶ Script Upload ▶ Audio Messages ▶ Certificates </div> <div style="width: 80%; padding: 10px;"> <h3>Direct Access Key Settings</h3> <table border="1"> <thead> <tr> <th></th> <th colspan="3">Function</th> </tr> </thead> <tbody> <tr> <td>DAK 1</td> <td>Idle: Call To</td> <td>8270001</td> <td>Ringlist 1</td> </tr> <tr> <td></td> <td>Call: Answer/End Call</td> <td>On Key Press</td> <td><input type="checkbox"/> Answer Group Call</td> </tr> <tr> <td>Input 1</td> <td>Idle: Call To</td> <td>8270002</td> <td>Ringlist 1</td> </tr> <tr> <td></td> <td>Call: Answer/End Call</td> <td>On Key Press</td> <td><input type="checkbox"/> Answer Group Call</td> </tr> <tr> <td>Input 2</td> <td>Idle: Call To</td> <td></td> <td>No Ringlist</td> </tr> <tr> <td></td> <td>Call: Do Nothing</td> <td></td> <td></td> </tr> <tr> <td>Input 3</td> <td>Idle: Call To</td> <td></td> <td>No Ringlist</td> </tr> <tr> <td></td> <td>Call: Do Nothing</td> <td></td> <td></td> </tr> <tr> <td>Input 4</td> <td>Idle: Call To</td> <td></td> <td>No Ringlist</td> </tr> <tr> <td></td> <td>Call: Do Nothing</td> <td></td> <td></td> </tr> <tr> <td>Input 5</td> <td>Idle: Call To</td> <td></td> <td>No Ringlist</td> </tr> <tr> <td></td> <td>Call: Do Nothing</td> <td></td> <td></td> </tr> <tr> <td>Input 6</td> <td>Idle: Call To</td> <td></td> <td>No Ringlist</td> </tr> <tr> <td></td> <td>Call: Do Nothing</td> <td></td> <td></td> </tr> <tr> <td>PTT / M-key</td> <td>Idle: Call To</td> <td></td> <td>No Ringlist</td> </tr> <tr> <td></td> <td>Call: Push To Talk</td> <td></td> <td></td> </tr> <tr> <td>Offhook</td> <td>Idle: Call To</td> <td></td> <td>No Ringlist</td> </tr> <tr> <td></td> <td>Call: Answer Call</td> <td></td> <td><input type="checkbox"/> Answer Group Call</td> </tr> <tr> <td>Onhook</td> <td>Idle: Call To</td> <td></td> <td>No Ringlist</td> </tr> <tr> <td></td> <td>Call: End Call</td> <td>On Key Press</td> <td></td> </tr> </tbody> </table> </div> </div>									Function			DAK 1	Idle: Call To	8270001	Ringlist 1		Call: Answer/End Call	On Key Press	<input type="checkbox"/> Answer Group Call	Input 1	Idle: Call To	8270002	Ringlist 1		Call: Answer/End Call	On Key Press	<input type="checkbox"/> Answer Group Call	Input 2	Idle: Call To		No Ringlist		Call: Do Nothing			Input 3	Idle: Call To		No Ringlist		Call: Do Nothing			Input 4	Idle: Call To		No Ringlist		Call: Do Nothing			Input 5	Idle: Call To		No Ringlist		Call: Do Nothing			Input 6	Idle: Call To		No Ringlist		Call: Do Nothing			PTT / M-key	Idle: Call To		No Ringlist		Call: Push To Talk			Offhook	Idle: Call To		No Ringlist		Call: Answer Call		<input type="checkbox"/> Answer Group Call	Onhook	Idle: Call To		No Ringlist		Call: End Call	On Key Press	
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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.

User Registrations
Select rows to send notifications to devices. Click on Details column for complete registration status.

View: **Default** Force Unregister AST Device Notifications: Reboot Reload Fallback As of 9:52 AM Advanced Search

21 Items Show 15 Filter: Enable

	Details	Address	First Name	Last Name	Actual Location	IP Address	Remote Office	Shared Control	Simult. Devices	AST Device	Registered		
											Prim	Sec	Surv
<input type="checkbox"/>	Show	8275000@devconnect.local	9611SIP	Station	---	10.10.16.121	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/> (AC)	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	8279999@devconnect.local	Stentofon3	Intercom	---	10.10.16.102	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/> (AC)	<input type="checkbox"/>	<input type="checkbox"/>

Select : All, None Page 1 of 2

7.2. Verify Turbine SIP Registration

From the Turbine 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.

Station Main SIP Configuration Station Administration Advanced SIP Advanced Network

Station Information Main Settings

TFIE-1 Information

Description	Information
Station IP:	10.10.16.102
Subnet Mask:	255.255.255.0
Default Gateway:	10.10.16.1
DNS Server 1:	10.10.16.10
DNS Server 2:	
Hardware Type:	8124
Hardware Version:	1
Software Versions:	List
Image Package Version:	4.7.3.0 (sti)
MAC Address:	00:13:cb:0d:10:1f
System Model Name:	Vingtor-Stentofon Turbine Extended - Industrial
Hardware Revision:	0004
Kernel Version:	3.10.0[release/intercom4.7_27e5eb5]+ #1 PREEMPT Tue Oct 24 16:15:51 CEST 2017
Devicetree Version:	06
Boot/Environment Version:	2016.02.05/2017.05.19

Station Status

Description	Status
Station Mode:	SIP
Display Name:	TFIE-1
Directory Number (SIP ID):	8279999
Server Domain (SIP):	devconnect.local, Registered - Thu Jan 1 18:27:00 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 IP Intercom Substation to interoperate with Avaya Aura® Session Manager and Avaya Aura® Communication 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 7.1.1, August 2017*
- [2] *Administering Avaya Aura® Communication Manager Release 7.1, 8 May 2017*
Document ID 03-300509
- [3] *Avaya Aura® Communication Manager Feature Description and Implementation Release 7.1 8 May 2017, Document ID 555-245-205*
- [4] *Administering Avaya Aura® System Manager 7.1.1, October 2017*

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

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

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