



Avaya Solution & Interoperability Test Lab

Application Notes for Zenitel Turbine with Avaya Aura® Communication Manager and Avaya Aura® Session Manager using TCP – Issue 1.0

Abstract

These Application Notes describe the configuration steps for provisioning Zenitel Turbine IP Intercom Station Series v6.1.1 to interoperate with Avaya Aura® Communication Manager R8.1 and Avaya Aura® Session Manager R8.1. Zenitel Turbine is an IP Intercom that supports voice transmission using Session Initiation Protocol (SIP) and Transmission Control Protocol (TCP).

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as any 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 for provisioning Zenitel Turbine IP Intercom Station Series v6.1.1 to interoperate with Avaya Aura® Communication Manager R8.1 and Avaya Aura® Session Manager R8.1. Zenitel Turbine is an IP Intercom that supports voice transmission using Session Initiation Protocol (SIP) and Transmission Control Protocol (TCP).

The Zenitel Turbine IP Intercom Stations (Turbine Stations) are designed for intelligent communications as part of the Zenitel Intelligent Communication suite: SIP phones. Intelligent Communication is required for enterprise business intelligence and for critical communications.

The Turbine Stations are made for tough environments at entrance and egress points to office buildings and gate and warehouse doors where clear communication is an issue. Also, in sectors like Building Security and Public Safety Oil & Gas, Heavy Industry, Transportation and even Marine. All intercom stations in the Zenitel's Turbine series utilize the latest technology and some of the features include, HD voice quality, Open Duplex, Active Noise Cancellation, MEMS microphone, a 10W Class D amplifier.

During compliance testing, each Zenitel Turbine IP Intercom Station was set up as a SIP user on Session Manager and underwent testing of various call scenarios with other Avaya telephones and Zenitel Turbine IP Intercom Stations.

The following models in the Zenitel Turbine family were tested: TCIS-3, TCIS-6, TMIS-1, TFIE-1, ECPiR-3P. TCIV-3+ was tested but failed as there was no video displayed and so is not supported at this time. Other models in the Turbine family are not covered by this compliance test.

Note: The Zenitel Turbine phones may be referred to as Zenitel Turbine, Turbine Stations, Zenitel Turbine IP Intercom Station, Turbine Intercom, Turbine or Zenitel Turbine IP Intercoms throughout this document, but they all refer to the same phones that were tested.

2. General Test Approach and Test Results

The general test approach was to place calls to and from the Turbine Intercom phones and exercise basic telephone operations. For serviceability testing, failures such as LAN 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's formal testing and Declaration of Conformity is provided only on the headsets/Smartphones that carry the Avaya brand or logo. Avaya may conduct testing of non-Avaya headset/Smartphone to determine interoperability with Avaya phones. However, Avaya does not conduct the testing of non-Avaya headsets/Smartphones for: Acoustic Pressure, Safety, Hearing Aid Compliance, EMC regulations, or any other tests to ensure conformity with safety, audio quality, long-term reliability or any regulation requirements. As a result, Avaya makes no representations whether a particular non-Avaya headset will work with Avaya's telephones or with a different generation of the same Avaya telephone.

Since there is no industry standard for Smartphone interfaces, different manufacturers utilize different Smartphone/headset interfaces with their telephones. Therefore, any claim made by a headset vendor that its product is compatible with Avaya telephones does not equate to a guarantee that the headset will provide adequate safety protection or audio quality

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 did not include use of any specific encryption features as requested by Zenitel.

2.1. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability testing. TCIS-3, TCIS-6, TMIS-1, TFIE-1 and ECPIR-3P models were tested. The feature testing was to verify that:

- Turbine successfully registers with Session Manager using the TCP protocol.
- Turbine successfully establishes audio calls with RTP audio to Avaya H.323, SIP and digital endpoints.
- Turbine successfully establishes audio calls with a simulated PSTN.
- Turbine successfully negotiates the appropriate audio codec.
- DTMF tones could be passed successfully to energize relay on Turbine unit to allow a door to be opened from the phone sending the DTMF tone or perhaps to switch audio direction.
- Turbine successfully calls multiple Avaya destinations in a hunt group.
- Turbine successfully calls a variety of endpoints in its call list set on the Turbine phone.
- Testing of video calls to/from the Turbine TCIV-3+ video phone.

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

Note: Compliance testing was carried out with the Turbine phones set to use TCP/RTP. Testing was also carried out with Turbine phones set to use TLS/SRTP and these Application Notes are labelled, *Application Notes for Zenitel Turbine with Avaya Aura® Communication Manager and Avaya Aura® Session Manager using TLS*.

2.2. Test Results

All test cases passed successfully with the following observation noted.

1. Call Park has a different meaning on the Turbine functionality than that of the Call Park feature on Communication Manager. When the Call Park function is used on Turbine it places multiple calls on hold. For every Direct Access Key (DAK) with Call Park configured, there can be only one active or resumed call.
2. The Turbine TCIV-3+ video phone was included in the compliance testing but failed to produce a video signal to the Avaya phones and so is not supported as part of the Turbine phones that were included in these Application Notes.

2.3. Support

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

- **Phone:** +1 816 231 7200 (Americas) +47 4000 2700 (Global)
- **Email:** cs@zenitel.com
- **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 H.323 and Digital phones registering directly to Communication Manager and SIP phones registering to Session Manager using the features on Communication Manager. A SIP trunk connects Communication Manager to a simulated PSTN.

Note: The Zenitel Turbine phones register to Session Manager the same as the Avaya SIP phones.

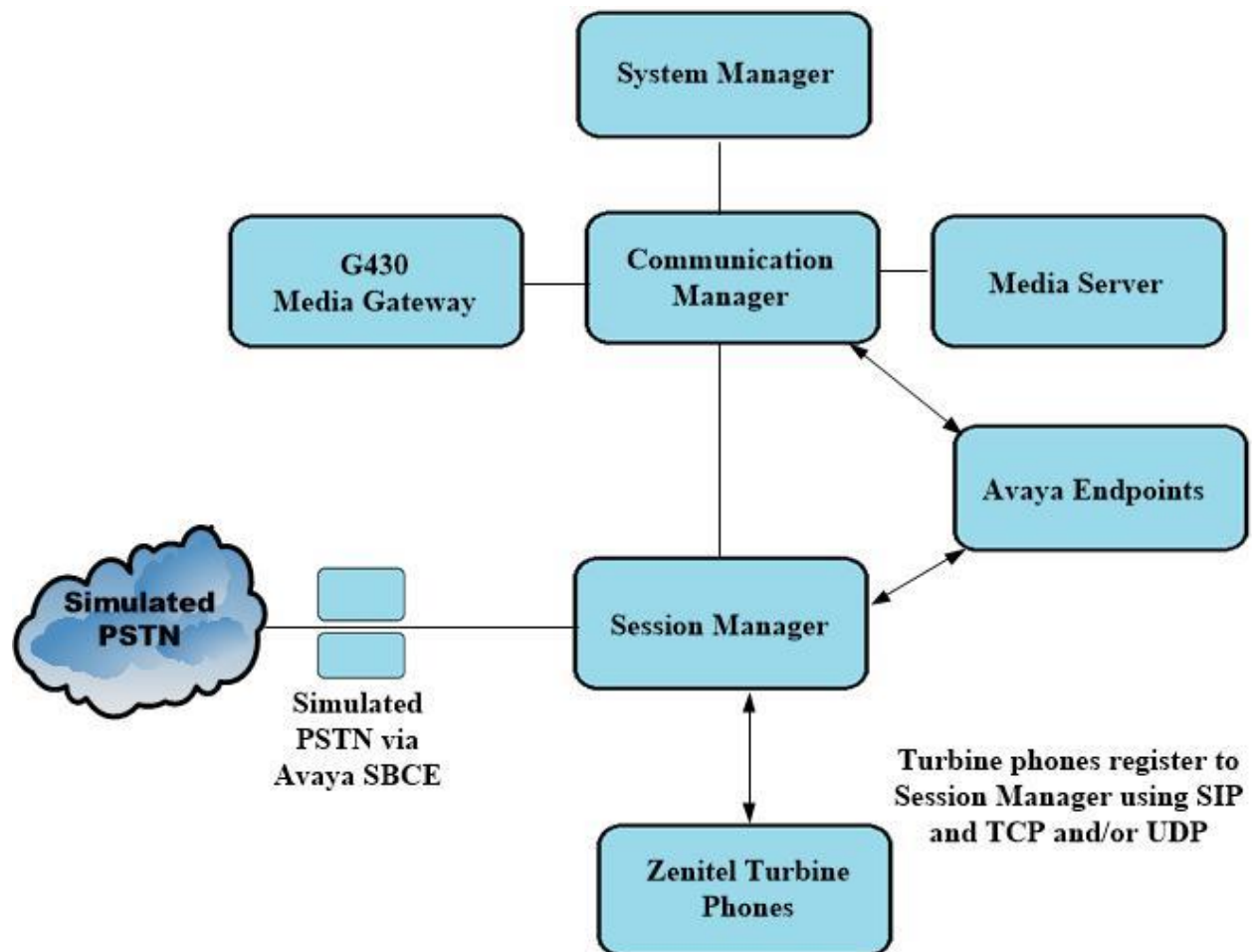


Figure 1: Configuration of Avaya Aura® Communication Manager and Avaya Aura® Session Manager with Zenitel Turbine

4. Equipment and Software Validated

The following equipment and software were used for the compliance test.

Avaya Equipment	Software / Firmware Version
Avaya Aura® System Manager running on a virtual server	8.1.3.0 Build No. – 8.1.0.0.733078 Software Update Revision No: 8.1.3.0.1011784 Feature Pack 3
Avaya Aura® Session Manager running on a virtual server	8.1.3 Build No. – 8.1.3.0.813014
Avaya Aura® Communication Manager running on a virtual server	8.1.3 – FP3 R018x.01.0.890.0 Update ID 01.0.890.0-26568
Avaya Session Border Controller for Enterprise	8.1.1.0-26-19214
Avaya Aura® Media Server	8.0.2.138
Avaya G430 Media Gateway	41.16.0/1
Avaya J179 H.323 Deskphone	6.8304
Avaya Vantage K175 SIP Deskphone	4.0.7.1.5
Avaya 9408 Digital Phone	2.00
Zenitel Equipment	Software / Firmware Version
Zenitel Turbine IP Intercom - TCIS-3 - TCIS-6 - TMIS-1 - TFIE-1 - ECPIR-3P	6.1.1.0 6.1.1.0 6.1.1.0 6.1.1.0 6.1.1.0

5. Configure Avaya Aura® Communication Manager

It is assumed that a fully functioning Communication Manager is in place with the necessary licensing with SIP trunks in place to Session Manager. For further information on the configuration of Communication Manager please see **Section 10** of these Application Notes.

Note: A printout of the Signalling and Trunk groups that were used during compliance testing can be found in the **Appendix** of these Application Notes.

The following sections go through the following.

- System Parameters
- Dial Plan Analysis
- Network Region
- IP Codec

5.1. Configure System Parameters

Ensure that the SIP endpoints license is valid as shown below by using the command **display system-parameters customer-options**.

display system-parameters customer-options		Page	1 of 12
OPTIONAL FEATURES			
G3 Version: V17	Software Package: Enterprise		
Location: 2	System ID (SID): 1		
Platform: 28	Module ID (MID): 1		
		USED	
Platform Maximum Ports:		48000	168
Maximum Stations:		36000	44
Maximum XMOBILE Stations:		36000	0
Maximum Off-PBX Telephones - EC500:		41000	2
Maximum Off-PBX Telephones - OPS:		41000	20
Maximum Off-PBX Telephones - PBFMC:		41000	0
Maximum Off-PBX Telephones - PVFMC:		41000	0
Maximum Off-PBX Telephones - SCCAN:		0	0
Maximum Survivable Processors:		313	1

5.2. Configure Dial Plan Analysis

Use the **change dialplan analysis** command to configure the dial plan using the parameters shown below. Extension numbers (**ext**) are those beginning with **1**. Feature Access Codes (**fac**) use digits **8** and **9** and use characters ***** or **#**.

change dialplan analysis						Page 1 of 12			
DIAL PLAN ANALYSIS TABLE									
Location: all						Percent Full: 5			
Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type	
1	4	ext							
3	4	udp							
6	4	ext							
8	1	fac							
9	1	fac							
*8	4	dac							
*	3	fac							
#	3	fac							

5.3. Configure Network Region

Use **change ip-network-region x** (where x is the network region to be configured) to assign an appropriate domain name to be used by Communication Manager, in the example below **devconnect.local** is used. Note that this domain is also configured in **Section 6.1.1**.

change ip-network-region 1

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IP NETWORK REGION

Region: 1NR Group: 1

Location: 1Authoritative Domain: devconnect.local

Name: PG DefaultStub Network Region: n

MEDIA PARAMETERSIntra-region IP-IP Direct Audio: yes

Codec Set: 1Inter-region IP-IP Direct Audio: yes

UDP Port Min: 2048IP Audio Hairpinning? n

UDP Port Max: 3329

DIFFSERV/TOS PARAMETERS

Call Control PHB Value: 46

Audio PHB Value: 46

Video PHB Value: 26

802.1P/Q PARAMETERS

Call Control 802.1p Priority: 6

Audio 802.1p Priority: 6

Video 802.1p Priority: 5

AUDIO RESOURCE RESERVATION PARAMETERS

H.323 IP ENDPOINTSRSVP Enabled? n

H.323 Link Bounce Recovery? y

Idle Traffic Interval (sec): 20

Keep-Alive Interval (sec): 5

5.4. Configure IP-Codec

Use the **change ip-codec-set x** (where x is the ip-codec set used) command to designate a codec set compatible with the Turbine doorphone. During compliance testing the codecs **G.711A**, **G.729A** and **G.722** were tested.

For compliance testing the Avaya phones are set to use Media Encryption and the Turbine phones have no encryption set, so **none** must be present in the **Media Encryption**.

change ip-codec-set 1

Page 1 of 2

IP MEDIA PARAMETERS

Codec Set: 1

	Audio	Silence	Frames	Packet
	Codec	Suppression	Per Pkt	Size (ms)
1:	G.711A	n	2	20
2:	G.729A	n	2	20
3:	G.722.2	n	1	20
4:	G.722-64K		2	20
5:				
6:				
7:				

Media Encryption	Encrypted SRTCP: enforce-unenc-srtcp
1: 1-srtp-aescm128-hmac80	
2: none	
3:	
4:	

6. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager. Session Manager is configured via System Manager. The procedures include the following areas:

- Domains and Locations
- Adding Zenitel Turbine SIP Users

To make changes on Session Manager a web session is established to System Manager. Log into System Manager by opening a web browser and navigating to <https://<System Manager FQDN>/SMGR>. Enter the appropriate credentials for the **User ID** and **Password** and click on **Log On**.

The screenshot displays the login interface for Avaya Aura Session Manager. The browser address bar shows the URL: smgr81xvmpg.devconnect.local/securityserver/UI/Login?org=dc=nortel,dc=com&goto=https://smgr81xvmpg.devconnect.local:443. The login form includes a 'User ID' field containing 'admin' and a 'Password' field with masked characters. 'Log On' and 'Reset' buttons are positioned below the password field. A disclaimer box on the left contains the following text: '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. Unauthorized users are subject to company disciplinary procedures and or criminal and civil penalties under state, federal, or other applicable domestic and foreign laws. The use of this system may be monitored and recorded for administrative and security reasons. Anyone accessing this system expressly consents to such monitoring and recording, and is advised that if it reveals possible evidence of criminal activity, the evidence of such activity may be provided to law enforcement officials. All users must comply with all corporate instructions regarding the protection of information assets.' A blue box at the bottom right specifies 'Supported Browsers: Internet Explorer 11.x or Firefox 65.0, 66.0 or 67.0.'

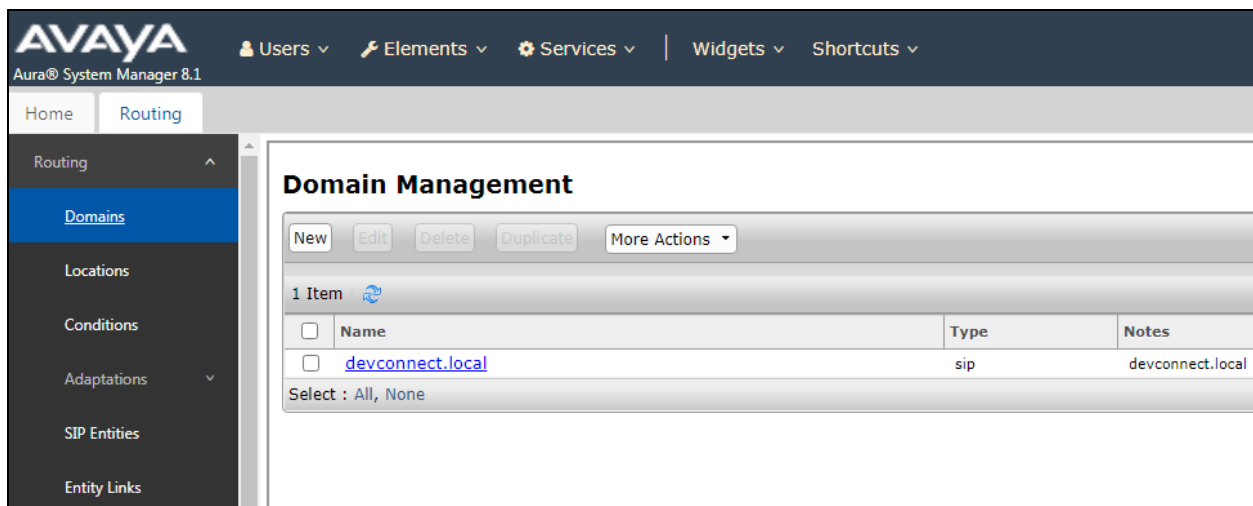
Once logged in navigate to **Elements** and click on **Routing** (not shown).

6.1. Domains and Locations

Note: It is assumed that a domain and a location have already been configured, therefore a quick overview of the domain and location that was used in compliance testing is provided here.

6.1.1. Display the Domain

Select **Domains** from the left window. This will display the domain configured on Session Manager. For compliance testing this domain was **devconnect.local** as shown below. If a domain is not already in place, click on **New**. This will open a new window (not shown) where the domain can be added.

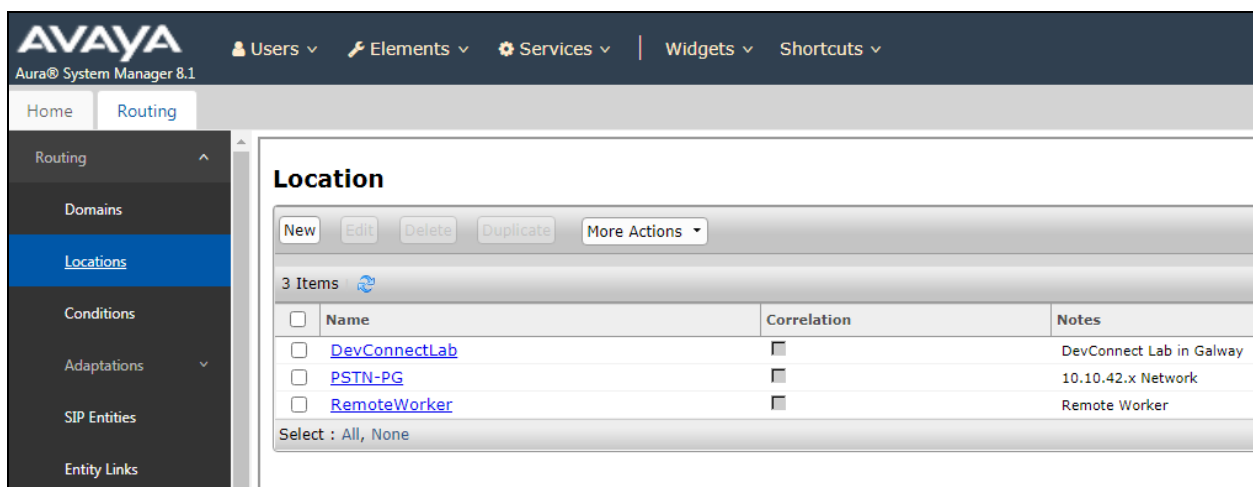


The screenshot shows the Avaya Aura System Manager 8.1 interface. The top navigation bar includes 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts'. The left sidebar has 'Routing' selected, and 'Domains' is highlighted. The main content area is titled 'Domain Management' and shows a table with one item: 'devconnect.local' of type 'sip'.

Name	Type	Notes
devconnect.local	sip	devconnect.local

6.1.2. Display the Location

Select **Locations** from the left window and this will display the location setup. The example below shows the location **DevConnectLab_PG** which was used for compliance testing. If a location is not already in place, then one must be added to include the IP address range of the Avaya solution. Click on **New** to add a new location.

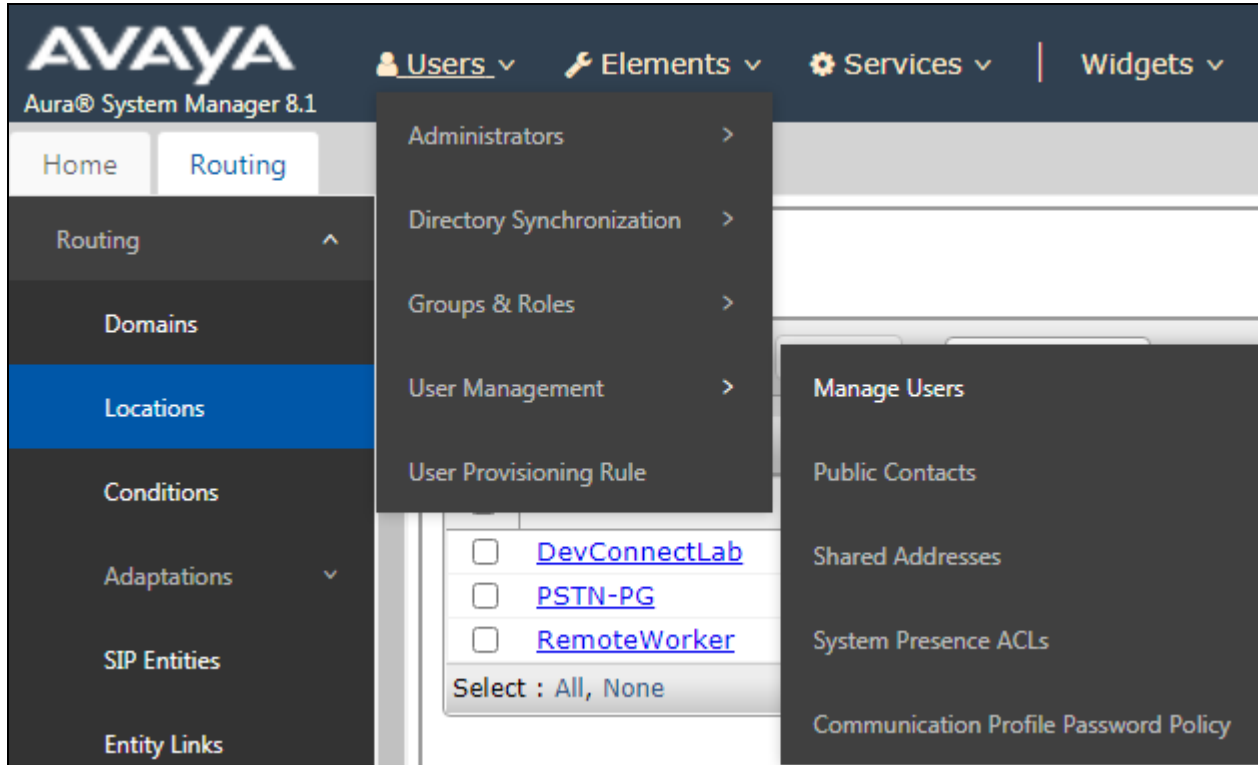


The screenshot shows the Avaya Aura System Manager 8.1 interface. The top navigation bar includes 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts'. The left sidebar has 'Routing' selected, and 'Locations' is highlighted. The main content area is titled 'Location' and shows a table with three items: 'DevConnectLab', 'PSTN-PG', and 'RemoteWorker'.

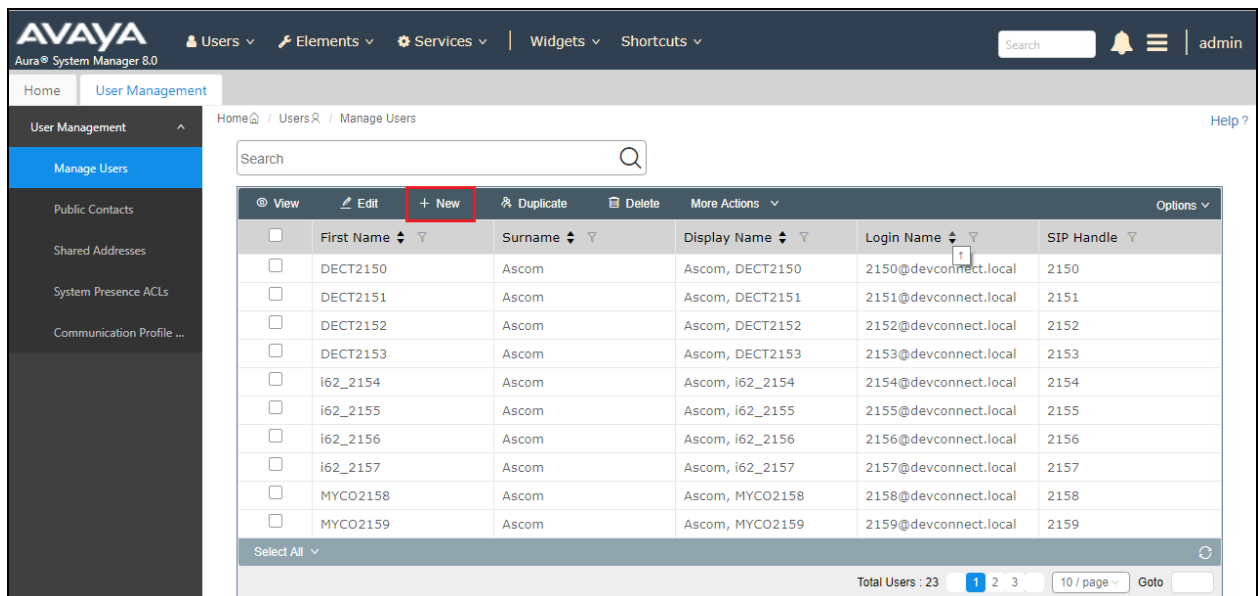
Name	Correlation	Notes
DevConnectLab		DevConnect Lab in Galway
PSTN-PG		10.10.42.x Network
RemoteWorker		Remote Worker

6.2. Adding Zenitel Turbine SIP Users

From the top of the home page click on **Users** → **User Management** → **Manager Users** as shown below.



From **Manager Users** section, click on **New** to add a new SIP user.



Under the **Identity** tab fill in the user's **Last Name** and **First Name** as shown below. Enter the **Login Name**, following the format of "user id@domain". The remaining fields can be left as default.

The screenshot shows the 'User Profile | Edit | 1153@devconnect.local' form with the 'Identity' tab selected. The form contains the following fields:

- User Provisioning Rule:** A dropdown menu.
- * Last Name:** Text input with 'Ext 1153'.
- Last Name (in Latin alphabet characters):** Text input with 'Ext 1153'.
- * First Name:** Text input with 'SIP'.
- First Name (in Latin alphabet characters):** Text input with 'SIP'.
- * Login Name:** Text input with '1153@devconnect.local'.
- Middle Name:** Text input with 'Middle Name Of User'.
- Description:** Text input with 'Description Of User'.
- Email Address:** Text input with 'Email Address Of User'.
- Password:** Text input.
- User Type:** Dropdown menu with 'Basic' selected.
- Confirm Password:** Text input.
- Localized Display Name:** Text input with 'Ext 1153, SIP'.
- Endpoint Display Name:** Text input with 'Ext 1153, SIP'.
- Title Of User:** Text input with 'Title Of User'.

Under the **Communication Profile** tab enter **Communication Profile Password** and **Re-enter Comm-Profile Password**, note that his password is required when configuring the Turbine phone in **Section 7.2**.

The screenshot shows the 'User Profile | Edit | 1153@devconnect.local' form with the 'Communication Profile' tab selected. A modal dialog titled 'Comm-Profile Password' is open, containing the following fields:

- Comm-Profile Password:** Text input with masked characters '....'.
- * Re-enter Comm-Profile Password:** Text input with masked characters '....' and a green checkmark icon.
- Generate Comm-Profile Password:** A blue link.
- Buttons:** 'Cancel' and 'OK'.

The background form shows the 'Communication Profile' tab with a 'Communication Profile Password' section and a list of profiles (Session Manager Profile, Avaya Breeze3 Profile, CM Endpoint Profile, Presence Profile) with toggle switches.

Staying on the **Communication Profile** tab, click on **New** to add a new **Communication Address**.

The screenshot shows the 'User Profile | Edit | 1153@devconnect.local' page with the 'Communication Profile' tab selected. On the left, there is a sidebar with 'Communication Profile Password', 'PROFILE SET : Primary', 'Communication Address' (highlighted), and 'PROFILES' including 'Session Manager Profile' (checked). The main area has a table with columns 'Type' and 'Select All'. A '+ New' button is highlighted in the top bar.

Enter the extension number and the domain for the **Fully Qualified Address** and click on **OK** once finished.

The screenshot shows the same page as before, but with a 'Communication Address Add/Edit' dialog box open. The dialog has a 'Type' dropdown set to 'Avaya SIP' and a '*Fully Qualified Address' field with '1153' in the extension box and 'devconnect.local' in the domain dropdown. 'Cancel' and 'OK' buttons are at the bottom right.

Ensure **Session Manager Profile** is checked and enter the **Primary Session Manager** details, enter the **Origination Sequence** and the **Termination Sequence**. Scroll down to complete the profile.

User Profile | Edit | 1153@devconnect.local

Identity

Communication Profile

Membership

Contacts

Communication Profile Password

PROFILE SET : Primary

Communication Address

PROFILES

Session Manager Profile

Avaya Breeze® Profile

CM Endpoint Profile

Presence Profile

SIP Registration

* Primary Session Manager :

SM81vmpg

Secondary Session Manager :

Start typing...

Survivability Server :

Start typing...

Max. Simultaneous Devices :

1

Block New Registration When Maximum Registrations Active? :

Application Sequences

Origination Sequence :

CMAPPSEQ

Termination Sequence :

CMAPPSEQ

Emergency Calling Application Sequences

Emergency Calling Origination Sequence :

Select

Emergency Calling Termination Sequence :

Select

PG; Reviewed:
SPOC 9/4/2021

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Enter the **Home Location**, this should be the location configured in **Section 6.1.2**. Click on **Commit** at the top of the page (not shown).

Application Sequences

Origination Sequence :

CMAPPSEQ

Termination Sequence :

CMAPPSEQ

Emergency Calling Application Sequences

Emergency Calling Origination Sequence :

Select

Emergency Calling Termination Sequence :

Select

Call Routing Settings

* Home Location :

DevConnectLab_PG

Conference Factory Set :

Select

Call History Settings

Enable Centralized Call History? : ☐

Ensure that **CM Endpoint Profile** is selected in the left window. Select the Communication Manager that is configured for the **System** and choose the **9620SIP_DEFAULT_CM_8_1** as the **Template**. **Sip Trunk** should be set to **aar**, providing that the routing is setup correctly on Communication Manager. The **Profile Type** should be set to **Endpoint** and the **Extension** is the number assigned to the Turbine phone. Click on **Endpoint Editor** to configure the buttons and features for that handset on Communication Manager.

The screenshot displays the 'User Profile | Edit | 1153@devconnect.local' interface. The 'Communication Profile' tab is active. On the left, a sidebar lists profile types: 'Session Manager Profile' (disabled), 'Avaya BreezeS Profile' (disabled), 'CM Endpoint Profile' (selected and enabled), and 'Presence Profile' (disabled). The main area contains the following fields and settings:

- System:** cm8bxvmpg
- Profile Type:** Endpoint
- Extension:** 1153
- Set Type:** 9620SIP
- Port:** JP
- Preferred Handle:** Select
- Sip Trunk:** aar
- Use Existing Endpoints:** ☐
- Template:** 9620SIP_DEFAULT_CM_8_1
- Security Code:** Enter Security Code
- Voice Mail Number:** (empty field)
- Calculate Route Pattern:** ☒
- SIP URI:** Select
- Delete on Unassign from User or on Delete User:** ☒
- Override Endpoint Name and Localized Name:** ☒
- Allow H.323 and SIP Endpoint Dual Registration:** ☐

At the top right, there are buttons for 'Commit & Continue', 'Commit', and 'Cancel'.

Under the **Feature Options** tab (not shown), ensure that **IP Video** is ticked, if the Turbine phone is capable of video. Other tabs can be checked but for compliance testing the values were left as default. Click on **Done** (not shown) to complete.

Note: For compliance testing the default value of three call appearance buttons were used. This can be changed under the **Button Assignment** tab.

Active Station Ringing	single	Auto Answer	none
MWI Served User Type	None	Coverage After Forwarding	system
Per Station CPN - Send Calling Number	None	Display Language	english
AUDIX Name	None	Hunt-to Station	
Remote Soft Phone Emergency Calls	as-on-local	Loss Group	19
LWC Reception	spe	Survivable COR	internal
IP Phone Group ID		Time of Day Lock Table	None
Speakerphone	2-way	Voice Mail Number	
Short/Prefixed Registration Allowed	default	Music Source	
EC500 State	enabled		
Bridging Tone for This Extension	None		

Features

<input type="checkbox"/> Always Use	<input type="checkbox"/> Idle Appearance Preference
<input type="checkbox"/> IP Audio Hairpinning	<input type="checkbox"/> IP SoftPhone
<input type="checkbox"/> Bridged Call Alerting	<input checked="" type="checkbox"/> LWC Activation
<input type="checkbox"/> Bridged Idle Line Preference	<input type="checkbox"/> CDR Privacy
<input checked="" type="checkbox"/> Coverage Message Retrieval	<input checked="" type="checkbox"/> Precedence Call Waiting
<input type="checkbox"/> Data Restriction	<input checked="" type="checkbox"/> Direct IP-IP Audio Connections
<input checked="" type="checkbox"/> Survivable Trunk Dest	<input type="checkbox"/> H.320 Conversion
<input type="checkbox"/> Bridged Appearance Origination Restriction	<input checked="" type="checkbox"/> IP Video
<input checked="" type="checkbox"/> Restrict Last Appearance	<input type="checkbox"/> Per Button Ring Control
<input type="checkbox"/> Turn on mute for remote off-hook attempt	
<input type="checkbox"/> IP Hoteling	

Once the **CM Endpoint Profile** is completed correctly, click on **Commit** to save the new user.

User Profile | Edit | 1153@devconnect.local
Commit & Continue
Commit
Cancel

Identity
Communication Profile
Membership
Contacts

Communication Profile Password

PROFILE SET : Primary

Communication Address

PROFILES

Session Manager Profile
Avaya Breeze® Profile
CM Endpoint Profile
Presence Profile

* System : cm81xvmpg

* Profile Type : Endpoint

Use Existing Endpoints :

* Extension : 1153

Template : 9620SIP_DEFAULT_CM_8_1

* Set Type : 9620SIP

Security Code : Enter Security Code

Port : IP

Voice Mail Number :

Preferred Handle : Select

Calculate Route Pattern :

Sip Trunk : aar

SIP URI : Select

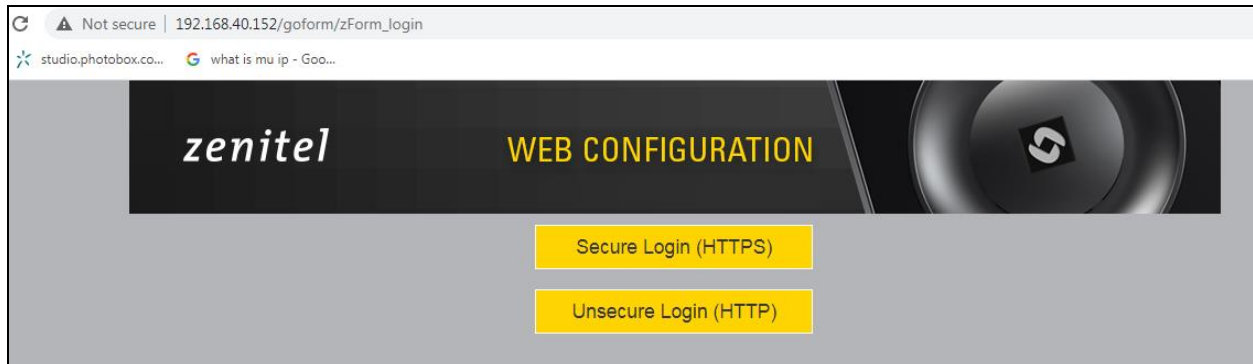
Delete on Unassign from User or on Delete User :

Override Endpoint Name and Localized Name :

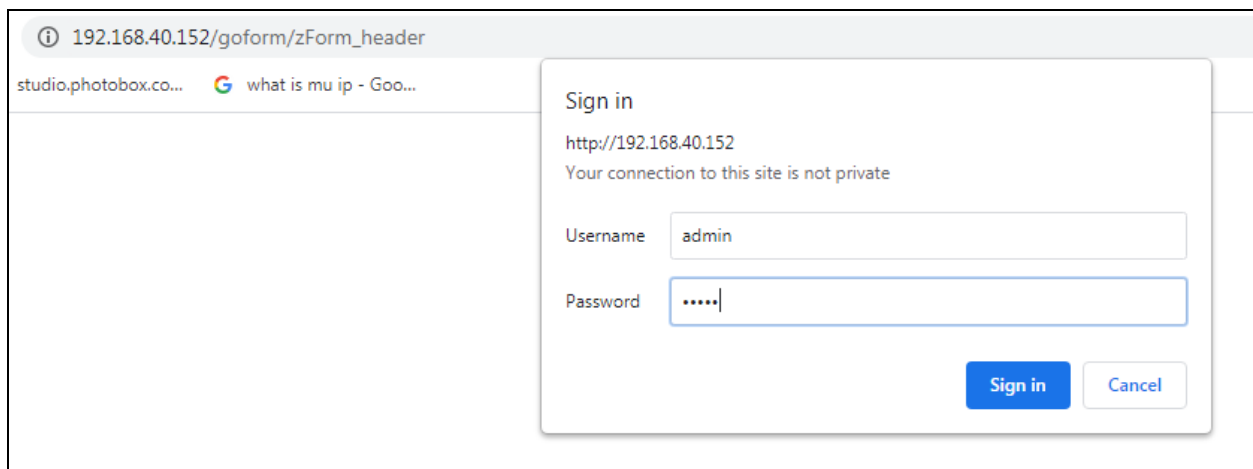
Allow H.323 and SIP Endpoint Dual Registration :

7. Configure Zenitel Turbine

The following steps detail the configuration for Turbine using the web interface. Access the Turbine web interface, enter **http://<ipaddress>** in an Internet browser window, where **<ipaddress>** is the IP address of Turbine. For compliance testing **Unsecure Login (HTTP)** was chosen.



Log in with the appropriate credentials.



Upon logging in, information on that Turbine station is displayed. The following settings should be checked.

- Configure Advanced Configuration Mode
- SIP Configuration
- Direct Access Keys
- Audio

zenitel **WEB CONFIGURATION**

Main **SIP Configuration** **Station Administration**

▼ Information

▶ Main Settings

▶ Recovery

▶ Legal Information

TCIV-2+/TCIV-3+ Information

Description	Information
IP Address:	192.168.40.152
Subnet Mask:	255.255.255.0
Default Gateway:	192.168.40.1
IPv6 Address	
DNS Server 1:	192.168.40.1
DNS Server 2:	
DNS Server 3:	
MAC Address:	00:13:cb:28:05:ad
Software Version:	6.1.1.0
More Information:	Show/Hide

Status

Description	Status
Mode:	SIP
Uptime:	up 41 minutes
Name:	
Number (SIP ID):	1153
Server Domain (SIP):	devconnect.local, Registered - Tue Apr 20 14:23:05 2021
Backup Domain (SIP):	
Backup Domain 2 (SIP):	
Outbound Proxy:	10.10.40.32

Select **Recovery** from the left window and under **Preferences** enter the password for **Advanced configuration mode**, this password can be obtained from a Zenitel engineer as per **Section 2.3**.

Once the password is entered the check box will appear as shown and a tick can be placed and click on **Save** to confirm.

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7.2. SIP Configuration

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

- **Name:** Enter the desired name.
- **Number (SIP ID):** Enter a user extension administered from **Section 6.2**.
- **Server Domain (SIP):** Enter the Domain as per **Section 6.1.1**.
- **Authentication User Name:** Enter a user extension administered from **Section 6.2**.
- **Authentication Password:** Enter the **Login Code** from **Section 6.2**.
- **Outbound Proxy (optional):** Enter the IP address of Session Manager and **5060** as the **Port**.
- **Outbound Transport:** Set this to **TCP**.
- **SIP Scheme:** Set this to **sip**.
- **TLS Private Key:** This does not apply and can be left at default.

Main	SIP Configuration	Station Administration	Advanced SIP	Advanced Network																																				
Account / Call																																								
Account Settings																																								
<table><thead><tr><th>Description</th><th>Configuration</th></tr></thead><tbody><tr><td>Name:</td><td><input type="text"/></td></tr><tr><td>Number (SIP ID):</td><td><input type="text" value="1153"/></td></tr><tr><td>Server Domain (SIP):</td><td><input type="text" value="devconnect.local"/></td></tr><tr><td>Backup Domain (SIP):</td><td><input type="text"/></td></tr><tr><td>Backup Domain 2 (SIP):</td><td><input type="text"/></td></tr><tr><td>Registration Method:</td><td><input type="text" value="Parallel"/></td></tr><tr><td>Authentication User Name:</td><td><input type="text" value="1153"/></td></tr><tr><td>Authentication Password:</td><td><input type="password" value="...."/></td></tr><tr><td>Register Interval:</td><td><input type="text" value="100"/> (min. 30 seconds)</td></tr><tr><td>Register Failure Interval:</td><td><input type="text" value="60"/> (min. 5 seconds)</td></tr><tr><td>Outbound Proxy [optional]:</td><td><input type="text" value="10.10.40.32"/> Port: <input type="text" value="5060"/></td></tr><tr><td>Outbound Backup Proxy [optional]:</td><td><input type="text"/> Port: <input type="text" value="5060"/></td></tr><tr><td>Outbound Backup Proxy 2 [optional]:</td><td><input type="text"/> Port: <input type="text" value="5060"/></td></tr><tr><td>Outbound Transport:</td><td><input type="text" value="TCP"/></td></tr><tr><td>SIP Scheme:</td><td><input type="text" value="sip"/> Using sips forces all proxies to also use TLS</td></tr><tr><td>Verify TLS hostname:</td><td><input type="checkbox"/></td></tr><tr><td>TLS Private Key:</td><td><input type="text" value="turbine_server_sha256.key"/></td></tr></tbody></table>					Description	Configuration	Name:	<input type="text"/>	Number (SIP ID):	<input type="text" value="1153"/>	Server Domain (SIP):	<input type="text" value="devconnect.local"/>	Backup Domain (SIP):	<input type="text"/>	Backup Domain 2 (SIP):	<input type="text"/>	Registration Method:	<input type="text" value="Parallel"/>	Authentication User Name:	<input type="text" value="1153"/>	Authentication Password:	<input type="password" value="...."/>	Register Interval:	<input type="text" value="100"/> (min. 30 seconds)	Register Failure Interval:	<input type="text" value="60"/> (min. 5 seconds)	Outbound Proxy [optional]:	<input type="text" value="10.10.40.32"/> Port: <input type="text" value="5060"/>	Outbound Backup Proxy [optional]:	<input type="text"/> Port: <input type="text" value="5060"/>	Outbound Backup Proxy 2 [optional]:	<input type="text"/> Port: <input type="text" value="5060"/>	Outbound Transport:	<input type="text" value="TCP"/>	SIP Scheme:	<input type="text" value="sip"/> Using sips forces all proxies to also use TLS	Verify TLS hostname:	<input type="checkbox"/>	TLS Private Key:	<input type="text" value="turbine_server_sha256.key"/>
Description	Configuration																																							
Name:	<input type="text"/>																																							
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Verify TLS hostname:	<input type="checkbox"/>																																							
TLS Private Key:	<input type="text" value="turbine_server_sha256.key"/>																																							
Call Settings																																								

In the **Call Settings** section, configure as required the **DTMF method** as **RFC 2833** or whatever is set on Communication Manager. Configure other options as required. Click **SAVE** when done and a screen will appear (shown on the next page) to confirm the setting. The **Codec** is also set here, with **g711a** being set with the highest priority, as shown in the example below.

Call Settings	
Description	Configuration
Enable Auto Answer:	<input type="checkbox"/>
Auto Answer Delay:	0 seconds. Max 30 seconds.
Press and Hold Time:	0 seconds. Max 60 seconds. Defines how long a DAK key/Input must be pressed before the call is established.
Max Trying Time:	15 How long to wait on response before hanging up.
Max Ringing Time:	120 How long a call can be ringing before hanging up.
Max Conversation Time:	3600 How long a call can be in conversation before hanging up.
Max MP114 Speech Time:	0 How long between MP114 speech start/end before hanging up.
Max Queued Time:	20 How long a call can be queued before hanging up.
Max Queued Calls:	4 How many incoming calls can be queued. Max 5.
Use NAT Keep Alive:	<input type="checkbox"/>
Dialing Method:	Enbloc Dialing ▾
Enbloc Dialing Timeout:	No Timeout ▾
DTMF method:	RFC 2833 ▾
Conversation Mode:	Duplex ▾
PTT Mode:	Mic and speaker is controlled by PTT button ▾
Resume Call Automatically:	<input type="checkbox"/> Resume Call On-Hold Automatically After Emergency Priority Ends
Remote Controlled Audio Direction:	<input type="checkbox"/> (Received DTMF * to listen, DTMF # to talk, DTMF 0 for open duplex)
SIP Message Controlled Audio Direction:	<input type="checkbox"/> (SIP MESSAGE controls audio direction)
Boost Volume on Push To Talk:	<input type="checkbox"/>
Override Remote Push To Talk:	<input type="checkbox"/>
Force Open Duplex Using DTMF:	- ▾
Send DTMF */# with M key:	<input type="checkbox"/>
RTP Timeout value:	0 seconds. 0 = RTP Timeout Disabled.
SIP OPTIONS Timeout value:	0 seconds. 0 = SIP OPTIONS Timeout Disabled.
Codec g729:	Low Priority ▾
Codec g722:	Low Priority ▾
Codec g711a:	High Priority ▾
Codec g711u:	Low Priority ▾

SAVE

At this point the phone needs to be rebooted in order to save the SIP configuration, however this can be rebooted at a later stage should one wish to proceed with the configuration.

▶ DAVC	SIP Backup Domain:
▶ Direct Access Keys	SIP Backup Domain 2:
▶ Relays / Outputs	Registration Method: Parallel
▶ Time	SIP Authentication Username: 1153
▶ I/O	SIP Registration Interval updated: 100
▶ Frontboard Mapping	SIP Registration Fail Interval updated: 60
▶ Video	SIP Outbound Proxy Address: 10.10.40.32
▶ Advanced Video	SIP Outbound Proxy Port: 5060
▶ Script Upload	SIP Outbound Proxy Backup Address:
▶ Script Configuration	SIP Outbound Proxy Port: 5060
▶ Script Events	SIP Outbound Proxy Backup Address 2:
▶ Audio Messages	SIP Outbound Proxy Port 2: 5060
▶ Multicast Paging	Outbound Transport: TCP
▶ Certificates	SIP Scheme: sip
	TLS Private Key: turbine_server_sha256.key
	Not using Unencrypted SRTCP
	Not using Verify TLS hostname
	RTP timeout value: 0
	SIP OPTIONS timeout value: 0
	Auto answer mode: OFF
	Delay Call Setup: 0
	Max Trying Time: 15
	Max Ringing Time: 120
	Max Conversation Time: 3600
	Max Queued Time: 20
	Max Queued Calls: 4
	Max MP114 Speech Time: 0
	Use NAT keepalive: OFF
	Enbloc Dialing: ON
	Enbloc Dialing Timeout: 0 seconds
	DTMF method: RFC2833
	Default speaking mode: Open Duplex
	Resume Call Automatically: OFF
	Remote Controlled Volume Override Mode: OFF
	Message Controlled Volume Override Mode: OFF
	Not overriding remote Push To Talk
	Not boosting Volume On Push To Talk
	Send DTMF */# using M key: FALSE
	Configuration Saved!
	These changes require a reboot
	REBOOT

7.3. Configure Direct Access Keys

Click on the **Direct Access Keys** in the left window, this will bring up the functions as shown below where an extension to call can be assigned to the call button of the Turbine Intercom. This extension was an Avaya telephone, so when the button is pressed this telephone is called. Select **Button 1** to configure it. In the **Idle** field, select **Call To** from the drop down and enter the extension to be called when the button key is pushed. In the **Call** field, select **Answer/End Call** and **On Key Press**. This can be changed to use Hold or Transfer and other call features should they be required.

Main	SIP Configuration	Station Administration	Advanced SIP	Advanced Network
Account Settings				
Function				
Button 1				
Idle: Call To <input type="text" value="5123"/> No Ringlist <input type="text"/>				
Call: Answer/End Call <input type="text" value="Filter Dir. No."/> On Key Press <input type="text"/> <input type="checkbox"/> Answer Group Call				
Input 1				
Idle: Do Nothing <input type="text"/>				
Call: Answer/End Call <input type="text" value="Filter Dir. No."/> On Key Press <input type="text"/> <input type="checkbox"/> Answer Group Call				
Input 2				
Idle: Do Nothing <input type="text"/>				
Call: Answer/End Call <input type="text" value="Filter Dir. No."/> On Key Press <input type="text"/> <input type="checkbox"/> Answer Group Call				
Input 3				
Idle: Call To <input type="text"/> No Ringlist <input type="text"/>				
Call: Do Nothing <input type="text"/>				

7.4. Configure Audio

Click on **Audio** in the left window, the volume of the speaker can be changed here.

SIP	Audio Settings
Audio	Description Configuration
DAVC	Speaker Volume: <input type="text" value="3"/>
Direct Access Keys	Volume Override Level: <input type="text" value="5"/> Sets the volume during volume override. Volume and handset override happens during Emergency Group calls.
Relays / Outputs	Microphone Sensitivity: <input type="text" value="5"/> Default value 5. 0 = very low sensitivity
Time	Volume Control Ch2: <input type="text" value="0"/> Line Out Gain Shouldn't be used with accessories Valid range: [-62..+24] dB
I/O	Audio Profile: <input type="text" value="Normal"/>
Keyboard	Noise Reduction Level: <input type="text" value="0"/> 0 = disabled.
RTSP	Tone Volume: <input type="text" value="0"/> (-1)=disabled, 0=default, [1..4]=[-22..-1]dB
Script	Audio Out Source: <input type="text" value="Voice Audio"/> Main Audio Out (Speaker) Sources
Script Events	Audio Input Source: <input type="text" value="Normal Microphone"/> Audio source can be either line in or normal microphone
Script Upload	Line Out Source: <input type="text" value="Audio Ch2"/> Line out can play audio either from VoIP signal or direct from microphone
Audio Messages	Automatic Gain Control (AGC): <input type="checkbox"/> Automatic Gain Control. If speech level and environmental noise are very unstable it may be turned on.
Multicast Paging	Hardware AGC: <input type="text" value="Disabled"/> Hardware Automatic Gain Control. Select Area Profile or Manual Control to enter own values. Doesn't work if AGC is enabled. Not recommend to use in Duplex Conversation Modes!
Certificates	Automatic Volume Control (AVC): <input type="checkbox"/> Volume depends on noise level
	AVC Debug: <input type="checkbox"/> Shows current volume level on OLED display
	AVC Advanced: <input type="checkbox"/> Check to open advanced settings

If the phone was not rebooted earlier during the SIP configuration then click the **Main** tab and then click on **Recovery** as shown below. The telephone can be rebooted from this page.

Main

SIP Configuration

Station Administration

Advanced SIP

Advanced Network

Information

Main Settings

Recovery

Commands

Description	Action
Full reboot	REBOOT
Partial reboot	REBOOT
Factory reset	FACTORY RESET
Factory reset with DHCP	FACTORY RESET

Preferences

8. Verification Steps

The following steps can be taken to ensure that connections between Zenitel Turbine phones and Session Manager are up.

8.1. Session Manager Registration

Log into System Manager as done previously in **Section 6**, select **Session Manager** → **Dashboard**.

The screenshot shows the Avaya Aura System Manager 8.0 interface. The top navigation bar includes 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts'. A dropdown menu is open under 'Elements', listing various components: Avaya Breeze®, Communication Manager, Communication Server 1000, Conferencing, Device Adapter, Device Services, Media Server, Meeting Exchange, Messaging, Presence, Routing, Session Manager, and Web Gateway. The 'Session Manager' option is highlighted, and a sub-menu is visible with the following items: Dashboard, Session Manager Administration, Global Settings, Communication Profile Editor, Network Configuration, Device and Location Configuration, and Application Configuration. The main dashboard area contains several widgets: 'System Resource Utilization' (a bar chart showing utilization for 'opt', 'var', and 'emdata'), 'Alarms' (a circular gauge showing alarm levels), 'Application State' (a table showing license status, deployment type, multi-tenancy, OOBM state, and hardening mode), and 'Information' (a table showing counts and sync status for CM, Session Manager, System Manager, and UCM Applications). The bottom right corner displays usage statistics and a warning for 'STANEOUS ADMINISTRATIVE LOGINS'.

Resource	Utilization
opt	~10
var	~5
emdata	~15

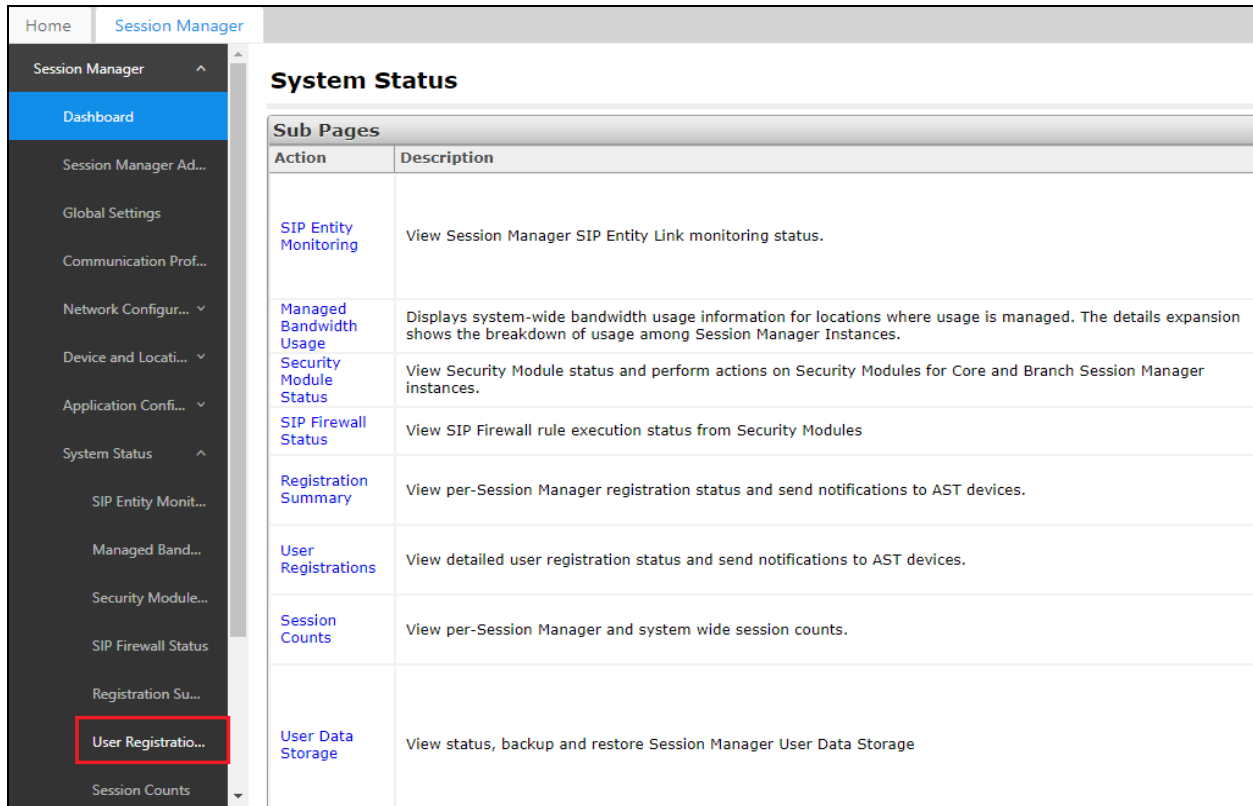
Property	Value
License Status	Active
Deployment Type	VMware
Multi-Tenancy	DISABLED
OOBM State	DISABLED
Hardening Mode	Standard

Elements	Count	Sync Status
CM	1	Green
Session Manager	1	Green
System Manager	1	Green
UCM Applications	8	Green

Alarm Type	Count
Critical	0
Major	0
Indeterminate	0
Minor	0
Warning	0

Category	Value
Usage	0000
STANEOUS ADMINISTRATIVE LOGINS	0

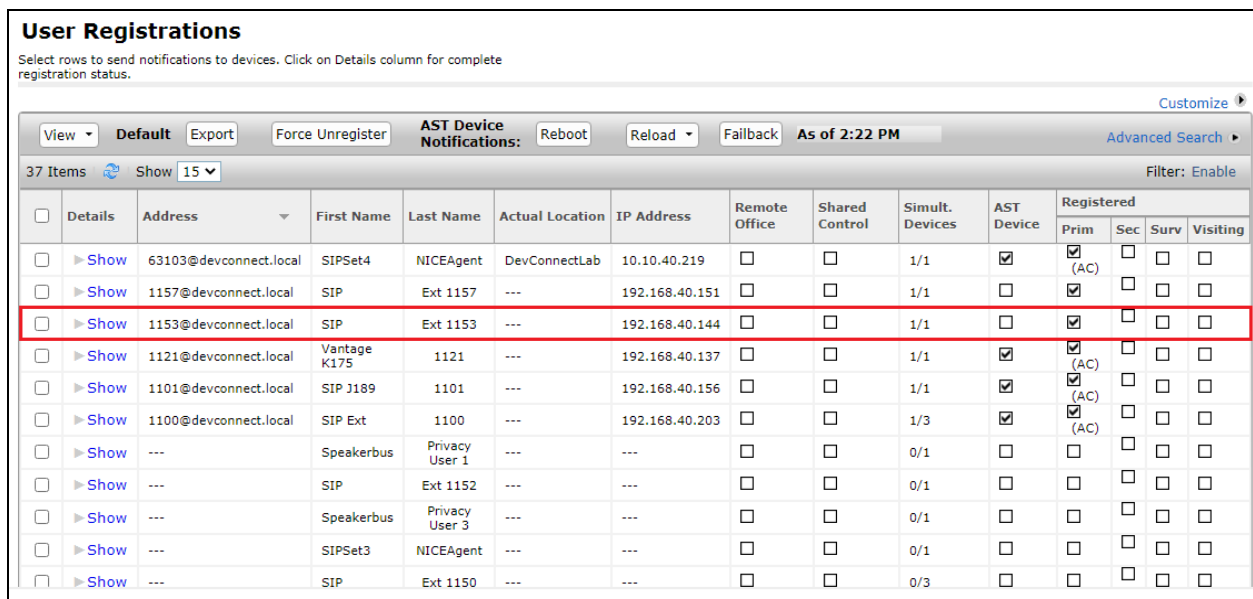
Under **System Status** in the left window, select **User Registrations** to display all the SIP users that are currently registered with Session Manager.



System Status

Sub Pages	
Action	Description
SIP Entity Monitoring	View Session Manager SIP Entity Link monitoring status.
Managed Bandwidth Usage	Displays system-wide bandwidth usage information for locations where usage is managed. The details expansion shows the breakdown of usage among Session Manager Instances.
Security Module Status	View Security Module status and perform actions on Security Modules for Core and Branch Session Manager Instances.
SIP Firewall Status	View SIP Firewall rule execution status from Security Modules
Registration Summary	View per-Session Manager registration status and send notifications to AST devices.
User Registrations	View detailed user registration status and send notifications to AST devices.
Session Counts	View per-Session Manager and system wide session counts.
User Data Storage	View status, backup and restore Session Manager User Data Storage

The user(s) should show as being registered as seen below.



User Registrations

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

View: Default Export Force Unregister AST Device Notifications: Reboot Reload Failback As of 2:22 PM Advanced Search

37 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	Visiting
<input type="checkbox"/>	Show	63103@devconnect.local	SIPSet4	NICEAgent	DevConnectLab	10.10.40.219	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	1157@devconnect.local	SIP	Ext 1157	---	192.168.40.151	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	1153@devconnect.local	SIP	Ext 1153	---	192.168.40.144	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	1121@devconnect.local	Vantage K175	1121	---	192.168.40.137	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	1101@devconnect.local	SIP J189	1101	---	192.168.40.156	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	1100@devconnect.local	SIP Ext	1100	---	192.168.40.203	<input type="checkbox"/>	<input type="checkbox"/>	1/3	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	---	Speakerbus	Privacy User 1	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	---	SIP	Ext 1152	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	---	Speakerbus	Privacy User 3	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	---	SIPSet3	NICEAgent	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	---	SIP	Ext 1150	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/3	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>

8.2. Zenitel Turbine Registration

To verify that the Turbine phone is registered correctly, log into the phone as per **Section 7**, the home page will show the following information where it can be seen if the phone is **Registered**.

TCIV-2+/TCIV-3+ Information	
Description	Information
IP Address:	192.168.40.152
Subnet Mask:	255.255.255.0
Default Gateway:	192.168.40.1
IPv6 Address	
DNS Server 1:	192.168.40.1
DNS Server 2:	
DNS Server 3:	
MAC Address:	00:13:cb:28:05:ad
Software Version:	6.1.1.0
More Information:	Show/Hide
Status	
Description	Status
Mode:	SIP
Uptime:	up 41 minutes
Name:	
Number (SIP ID):	1153
Server Domain (SIP):	devconnect.local, Registered - Tue Apr 20 14:23:05 2021
Backup Domain (SIP):	
Backup Domain 2 (SIP):	
Outbound Proxy:	10.10.40.32

9. Conclusion

These Application Notes describe the configuration steps required for Zenitel Turbine to successfully interoperate with Avaya Aura® Communication Manager R8.1 and Avaya Aura® Session Manager R8.1 over TCP by registering the Turbine phones with Session Manager as third-party SIP phones. Please refer to **Section 2.2** for test results and observations.

10. Additional References

This section references the product documentation relevant to these Application Notes. Product documentation for Avaya products may be found at <http://support.avaya.com>.

1. *Avaya Aura® Communication Manager Feature Description and Implementation*, Release 8.1.x
2. *Administering Avaya Aura® Session Manager*, Release 8.1.x

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

Appendix

Signaling Group

display signaling-group 1	Page 1 of 3
SIGNALING GROUP	
Group Number: 1	Group Type: sip
IMS Enabled? n	Transport Method: tls
Q-SIP? n	
IP Video? y	Enforce SIPS URI for SRTP? n
Peer Detection Enabled? y	Peer Server: SM
Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y	Clustered? n
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n	
Alert Incoming SIP Crisis Calls? n	
Near-end Node Name: procr	Far-end Node Name: SM81vmppg
Near-end Listen Port: 5061	Far-end Listen Port: 5061
	Far-end Network Region: 1
Far-end Domain:	
Incoming Dialog Loopbacks: eliminate	Bypass If IP Threshold Exceeded? n
DTMF over IP: rtp-payload	RFC 3389 Comfort Noise? n
Session Establishment Timer(min): 3	Direct IP-IP Audio Connections? y
Enable Layer 3 Test? y	IP Audio Hairpinning? n
H.323 Station Outgoing Direct Media? n	Initial IP-IP Direct Media? n
	Alternate Route Timer(sec): 6

Trunk Group Page 1

display trunk-group 1	Page 1 of 5
TRUNK GROUP	
Group Number: 1	Group Type: sip
Group Name: SIP PHONES	CDR Reports: y
Direction: two-way	COR: 1
Dial Access? n	TN: 1
Queue Length: 0	TAC: *801
Service Type: tie	Outgoing Display? n
	Night Service:
	Auth Code? n
	Member Assignment Method: auto
	Signaling Group: 1
	Number of Members: 10

Page 2

```
display trunk-group 1                                     Page 2 of 5
  Group Type: sip

TRUNK PARAMETERS

  Unicode Name: auto

                                         Redirect On OPTIM Failure: 5000

  SCCAN? n                                         Digital Loss Group: 18
    Preferred Minimum Session Refresh Interval(sec): 600

Disconnect Supervision - In? y Out? y

  XOIP Treatment: auto    Delay Call Setup When Accessed Via IGAR? n

Caller ID for Service Link Call to H.323 1xC: station-extension
```

Page 3

```
display trunk-group 1                                     Page 3 of 5
TRUNK FEATURES

  ACA Assignment? n          Measured: none          Maintenance Tests? y

Suppress # Outpulsing? n    Numbering Format: private
                               UII Treatment: shared
                               Maximum Size of UII Contents: 128
                               Replace Restricted Numbers? n
                               Replace Unavailable Numbers? n

                               Hold/Unhold Notifications? y
                               Modify Tandem Calling Number: no

  Send UCID? y

Show ANSWERED BY on Display? y

DSN Term? n
```

Page 4

```
display trunk-group 1                                     Page 4 of 5
                                     SHARED UI FEATURE PRIORITIES

                                     ASAI: 1

Universal Call ID (UCID): 2

MULTI SITE ROUTING (MSR)

    In-VDN Time: 3
    VDN Name: 4
    Collected Digits: 5
    Other LAI Information: 6
    Held Call UCID: 7
```

Page 5

```
trunk-group 1                                           Page 5 of 5
                                     PROTOCOL VARIATIONS

Mark Users as Phone? y
Prepend '+' to Calling/Alerting/Diverting/Connected Number? n
    Send Transferring Party Information? y
        Network Call Redirection? y
Build Refer-To URI of REFER From Contact For NCR? n
    Send Diversion Header? n
    Support Request History? y
    Telephone Event Payload Type: 101

    Convert 180 to 183 for Early Media? n
    Always Use re-INVITE for Display Updates? n
        Identity for Calling Party Display: P-Asserted-Identity
    Block Sending Calling Party Location in INVITE? n
    Accept Redirect to Blank User Destination? n
        Enable Q-SIP? n

Interworking of ISDN Clearing with In-Band Tones: keep-channel-active
    Request URI Contents: may-have-extra-digits
```

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