



## Avaya Solution & Interoperability Test Lab

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# Application Notes for Configuring Avaya IP Office Server Edition 10.1 to interoperate with Zenitel Turbine - Issue 1.0

### Abstract

These Application Notes describe the configuration steps required for Zenitel Turbine to interoperate with Avaya IP Office Server Edition 10.1. 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 IP Office. 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 IP Office 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/TCIV-6, TFIE 1-2 and TMIS-1 models were tested. The feature testing was to verify that:

- Turbine successfully registers with IP Office 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 IP office
- 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 IP Office.

## 2.2. Test Results

All test cases passed successfully.

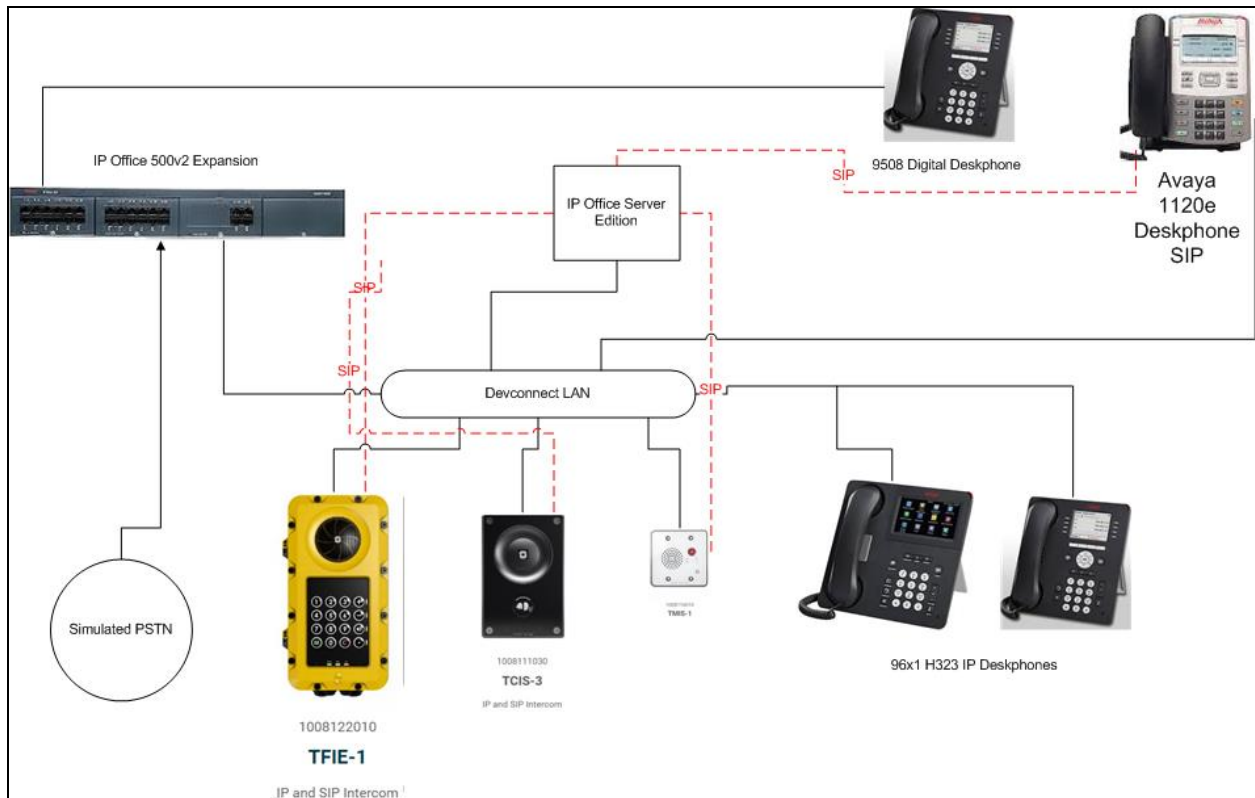
## 2.3. Support

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 IP Office. The configuration consists of IP Office Server Edition and 500v2 Expansion. IP Office has connections to 96x1 IP (H.323) deskphones. IP Office has SIP registrations with Turbine and 96x1 IP (SIP) deskphones. An ISDN-PRI trunk connects IP Office to the PSTN.



**Figure 1: Avaya IP Office 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 IP Office Server Edition	10.1.0.0.0 Build 237
Avaya IP Office 500v2	10.1.0.0.0 Build 237
Avaya 1120 SIP	SIP 1120e.04.04.23.00
Avaya 9508 Digital	NA
Avaya 9608G IP Telephone H323	6.6401
Zenitel Turbine	4.7.3

**Note:** Compliance Testing is applicable when the tested solution is deployed with a standalone IP Office 500 V2 and also when deployed with IP Office Server Edition in all configurations.

## 5. Avaya IP Office Configuration

Configuration and verification operations on the Avaya IP Office illustrated in this section were all performed using Avaya IP Office Manager. The information provided in this section describes the configuration of the Avaya IP Office for this solution. It is implied a working system is already in place. For all other provisioning information such as initial installation and configuration, please refer to the product documentation in **Section 9**. The configuration operations described in this section can be summarized as follows:

- Launch Avaya IP Office Manager
- LAN1 Configuration
- VoIP Configuration
- Create a SIP Extension for the Turbine Intercom
- Create a User for the Turbine Intercom
- Save Configuration

### 5.1. Launch Avaya IP Office Manager

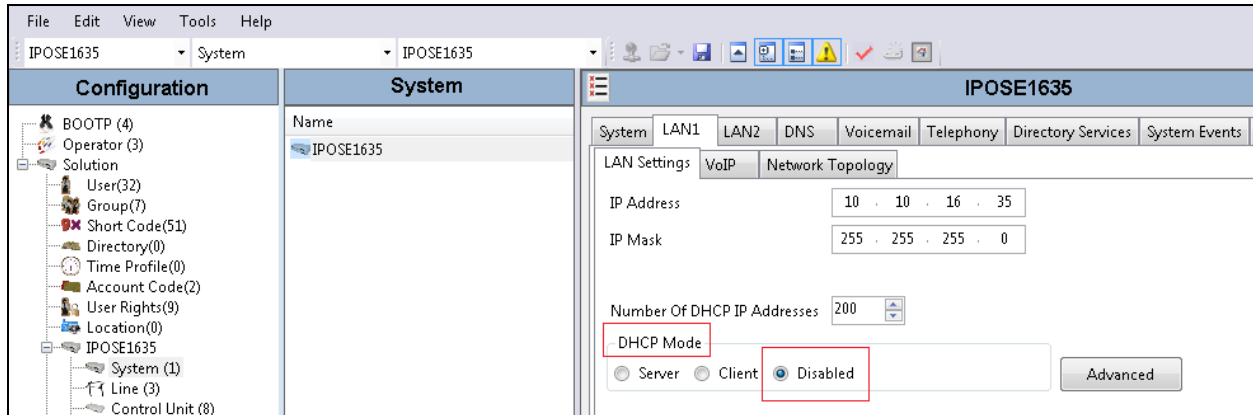
From the Avaya IP Office Manager PC, go to **Start**→**Programs**→**IP Office**→**Manager** to launch the Manager application. Log in to Avaya IP Office using the appropriate credentials to receive its configuration (not shown). In the IP Office window, click on Configuration. During compliance testing the System was called IPOSE1635.

The screenshot displays the Avaya IP Office Manager Configuration window. The left pane shows a tree view of the configuration hierarchy, including Solution, User, Group, Short Code, Directory, Time Profile, Account Code, User Rights, Location, IPOSE1635, System, Line, Control Unit, Extension, User, Group, Short Code, Service, Incoming Call Route, IP Route, Licence, ARS, Location, Authorization Code, and IPOMC. The main pane shows the Summary page for Server Edition Primary, which includes hardware installed and system settings information. The hardware installed section lists the control unit as IPO-Linux-PC, secondary server as NONE, expansion systems as 10.10.16.36, system identification as 21747dd27963f4aa47fe2c26975f294afebcaee0, and serial number as 00505694637d. The system settings section lists IP address as 10.10.16.35, sub-net mask as 255.255.255.0, system locale as United Kingdom (UK English), device ID as NONE, and number of extensions on system as 27. The bottom pane shows a table with the following data:

Description	Name	Address	Primary Link	Users Configured	Extensions Configured
Solution				32	48
Primary Server	IPOSE1635	10.10.16.35		25	27
Expansion System	IPOMC	10.10.16.36	Bothway	7	21

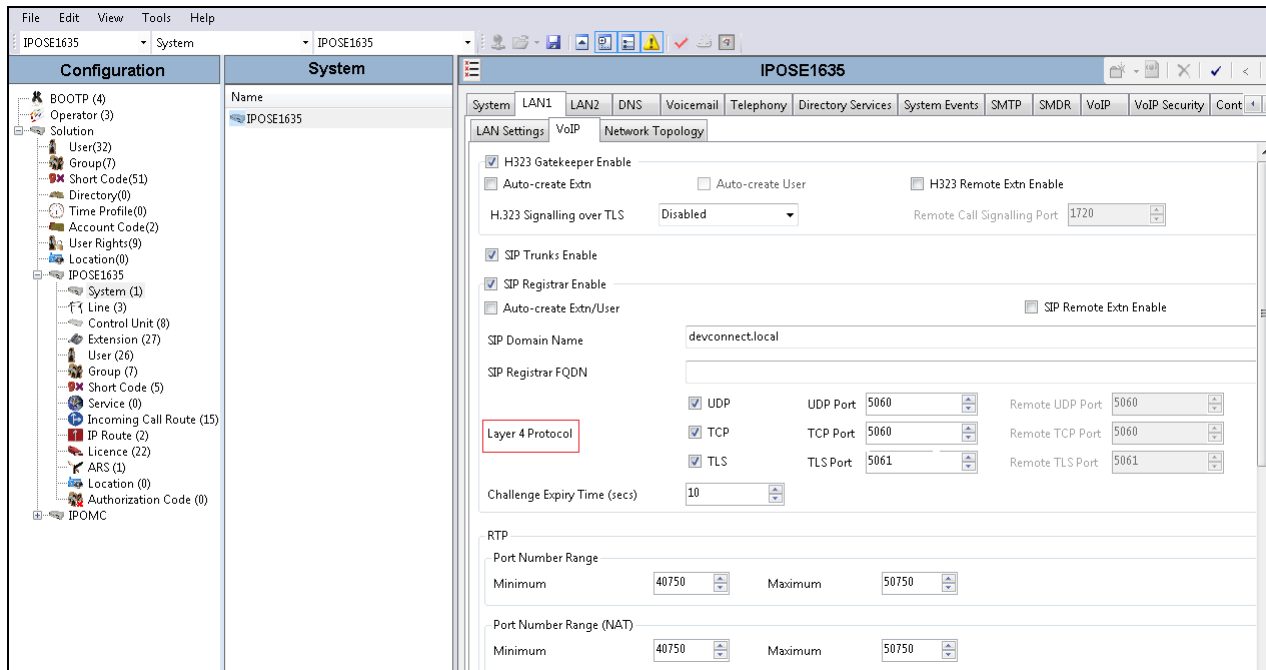
## 5.2. LAN1 configuration

For the Turbine handsets to communicate with the IP Office **DHCP MODE** must be disabled. To disable DHCP, select **IPOSE1635** → **System (1)** then on the **LAN1** tab followed by the **LAN Settings** tab click on the **Disabled** radio button in the **DHCP Mode** section. Click the **OK** button (not shown) to save.



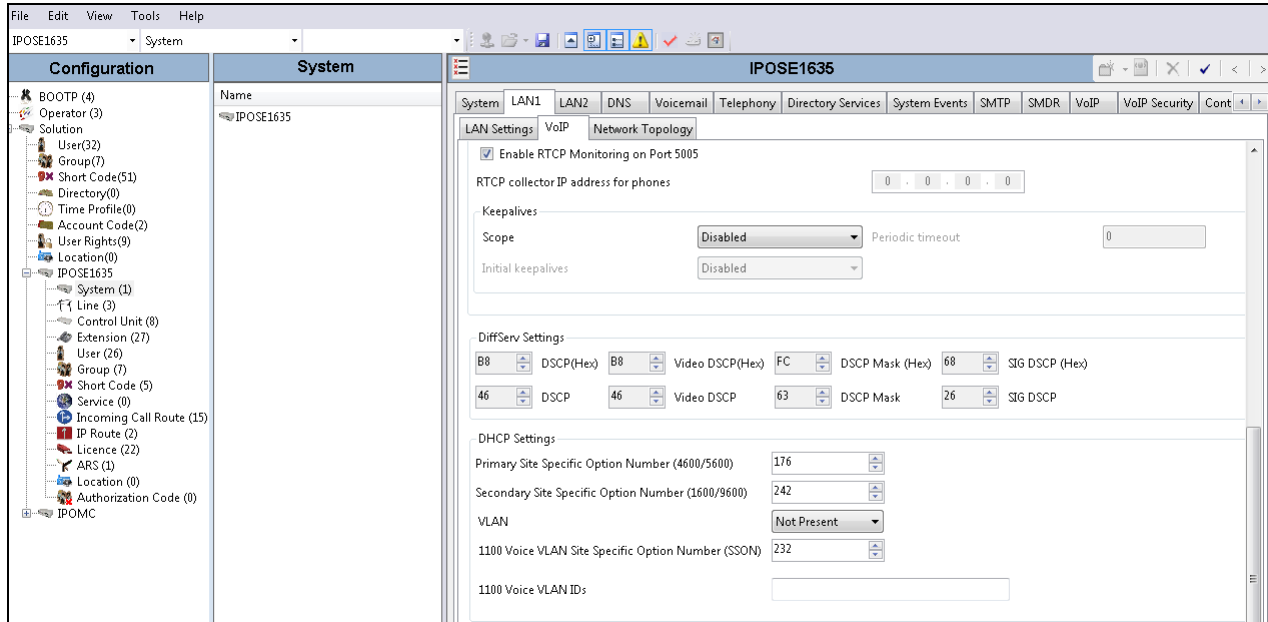
## 5.3. VoIP Configuration

Select the **VoIP** tab and in the **Layer 4 Protocol** section check the **UDP**, **TCP** and **TLS** check boxes and select **Port 5060** and **5061** from the dropdown boxes. Using the scroll bar on the right hand side, scroll down to the **DiffServ Settings** section.



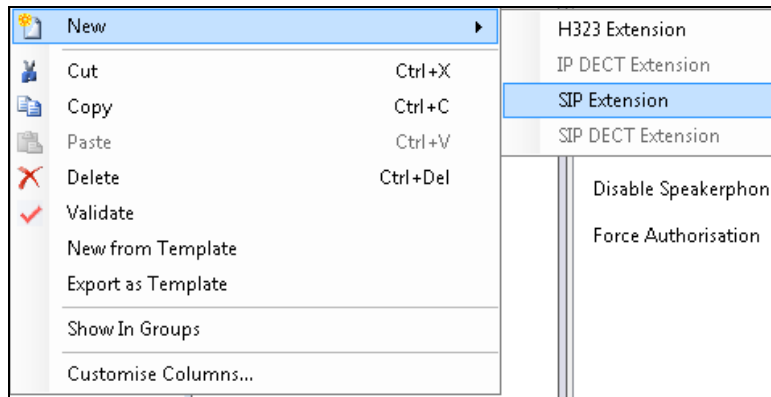


At the **DiffServ Settings** section select **46** from the **DSCP** drop down box and **26** from the **SIG DSCP** dropdown box. Click the **OK** button to save.



#### 5.4. Create a SIP Extension for the Turbine Intercom

The DECT Handsets are configured as SIP Extensions on the IP Office. From the Configuration Tree click on **Extension** then right click and select **New** followed by **SIP Extension**. The example below shows an extension 8352001; repeat these steps for each DECT Handset extension.



When the new window opens enter the **Base Extension**. The Extension ID will be automatically filled in.

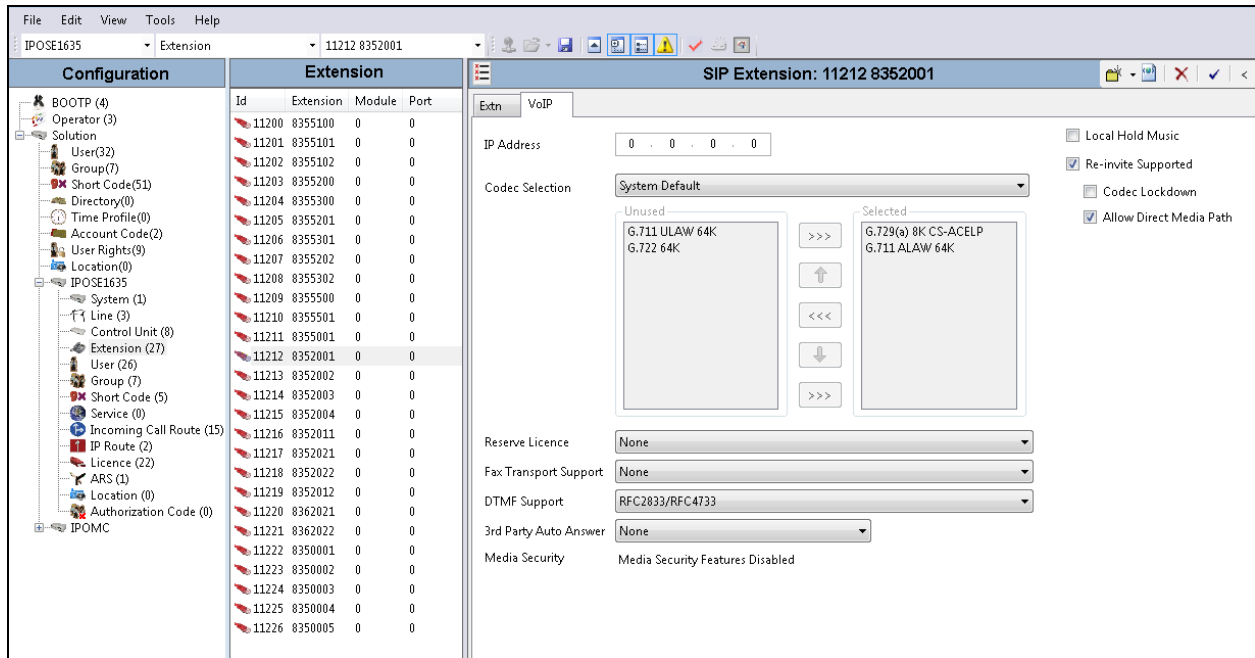
The screenshot displays the Avaya System Manager interface. On the left, a tree view shows the configuration hierarchy, with 'Extension (27)' selected under 'IPOSE1635'. The main area is divided into two panes. The left pane shows a table of extensions, and the right pane shows the configuration details for the selected extension.

Id	Extension	Module	Port
11200	8355100	0	0
11201	8355101	0	0
11202	8355102	0	0
11203	8355200	0	0
11204	8355300	0	0
11205	8355201	0	0
11206	8355301	0	0
11207	8355202	0	0
11208	8355302	0	0
11209	8355500	0	0
11210	8355501	0	0
11211	8355001	0	0
11212	8352001	0	0
11213	8352002	0	0
11214	8352003	0	0
11215	8352004	0	0
11216	8352011	0	0
11217	8352021	0	0
11218	8352022	0	0
11219	8352012	0	0
11220	8362021	0	0
11221	8362022	0	0
11222	8350001	0	0
11223	8350002	0	0
11224	8350003	0	0
11225	8350004	0	0
11226	8350005	0	0

The right pane, titled 'SIP Extension: 11212 8352001', shows the following configuration details:

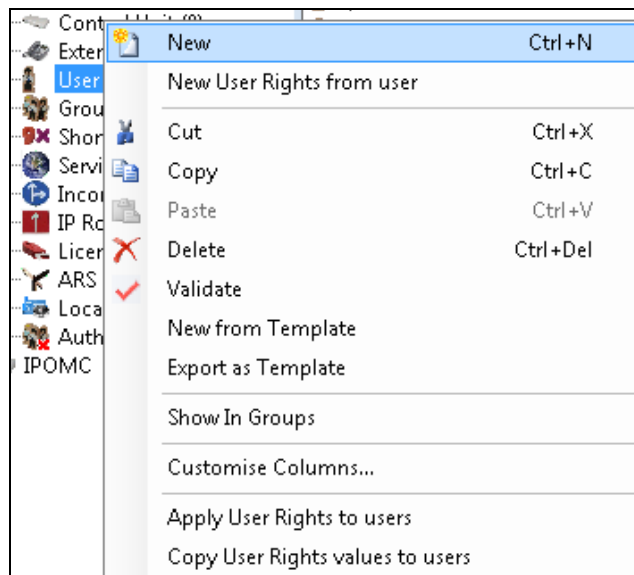
- Extension Id: 11212
- Base Extension: 8352001
- Caller Display Type: On
- Reset Volume After Calls:
- Device Type: Unknown SIP device
- Location: Automatic
- Fallback As Remote Worker: Auto
- Module: 0
- Port: 0
- Disable Speakerphone:
- Force Authorisation:

Click on the **VoIP** tab, and when the **VoIP** tab opens click the **Allow Direct Media Path** check box. Click the **OK** button to save.



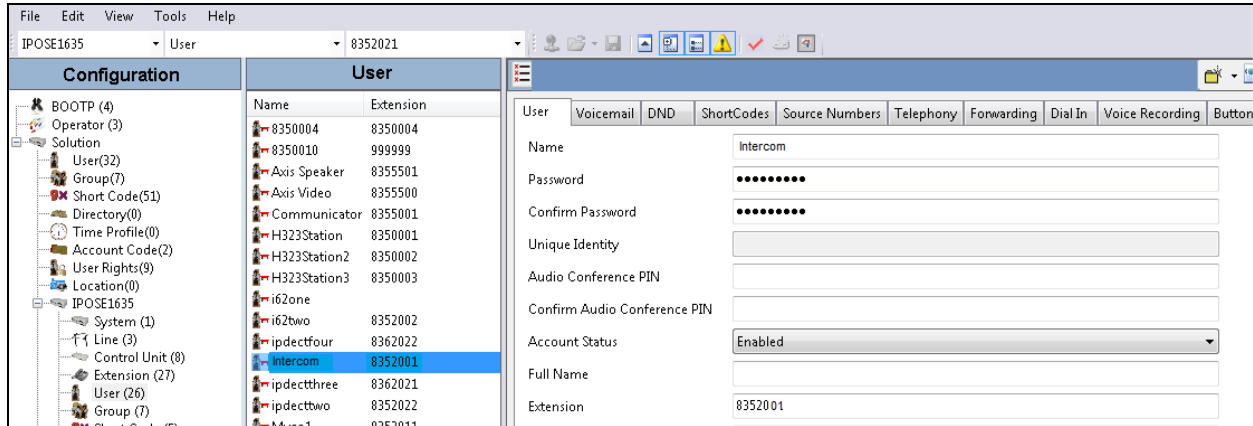
## 5.5. Create a User for the Turbine Intercom

A user must be configured for all Turbine Intercom Extensions. From the Configuration Tree, click on **User** then right click and select **New**.



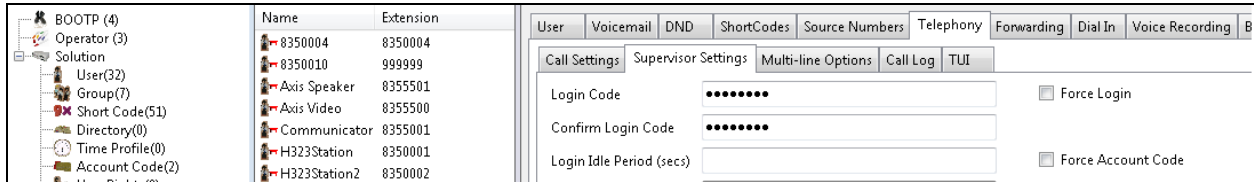
When the **User** window opens, select the **User** tab and enter the follow:

- **Name** Enter an name for this user, i.e. **Intercom**
- **Password** Enter the Password
- **Confirm Password** Confirm the Password
- **Extension** Enter the Extension which was created previously, i.e. **Section 5.4**



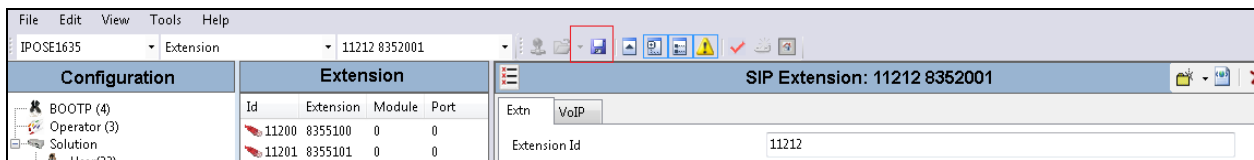
Click on **Telephony** tab followed by the **Supervisor Settings** tab and enter a Login Code in the **Login Code** box. Click the **OK** button (not shown) to save.

**Note:** The Login Code is used by the Funktel f.airnet DECT Handset to log in to the IP Office in **Section 6**. Ensure all DECT Handset Users use the same **Login Code**.

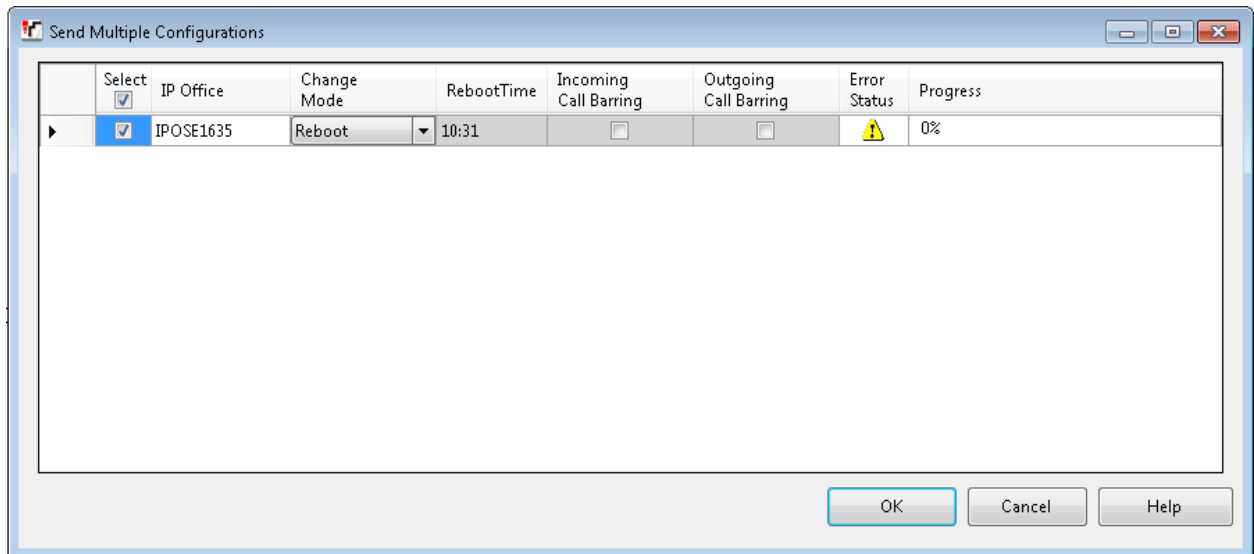


## 5.6. Save Configuration

Once all the configurations have been made it must be sent to the IP Office. Click on the **Save** Icon as shown below.



Once the **Save Configuration** Window opens, click the **OK** button.



## 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 Zenitel Turbine web interface. At the top, there are navigation tabs: Station Main, SIP Configuration, Station Administration, Advanced SIP, and Advanced Network. The main content area is divided into two sections: Station Information and TFIE-1 Information.

**Station 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):	8352001
Server Domain (SIP):	devconnect.local, Registered - Thu Jan 1 18:27:00 1970
Backup Domain (SIP):	
Backup Domain 2 (SIP):	
Outbound Proxy:	10.10.16.35

## 6.2. Add IP Office 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 sidebar menu lists various settings: SIP Settings, Audio Settings, Direct Access Key Settings, Relay Settings, Time Settings, I/O Settings, Keyboard Settings, Script Configuration, Script Events, and Script Upload. The main content area is titled 'Certificates' and contains a table with the following data:

	Name	
Certificate 1	<a href="#">turbine_server_sha1.key</a>	Delete
Certificate 2	<a href="#">turbine_server_sha256.key</a>	Delete

Below the table is the 'Upload Certificate' section, which includes a 'Choose File' button, a text input field containing 'root-ca.pem', and an 'Upload' button.

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

This screenshot shows the same SIP Configuration interface as the previous one, but with an additional certificate. The 'Certificates' table now includes three entries:

	Name	
Certificate 1	<a href="#">root-ca.pem</a>	Delete
Certificate 2	<a href="#">turbine_server_sha256.key</a>	Delete
Certificate 3	<a href="#">turbine_server_sha1.key</a>	Delete

The 'Upload Certificate' section below shows the 'Choose File' button and the text input field now displays 'No file chosen', indicating that the file has been successfully uploaded.

### 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 divided into three sections:

- Station Mode:** Radio buttons for 'Use Alphacom', 'Use Exigo', 'Use SIP' (selected), 'Use Pulse', and 'Use Pulse Server'.
- Product Model And Accessory:** 'Model:' dropdown set to 'TFIE-1' and 'Accessory:' dropdown set to 'No accessory'.
- IP Settings:** 'DHCP' and 'Static IP' (selected) radio buttons. Below are input fields for:
 

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

 'Hostname:' set to 'zenitel0d101f'. Checkboxes for '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.



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 5.4**.
- **Server Domain (SIP):** Enter the Domain of IP Office.
- **Authentication User Name:** Enter a user extension administered from **Section 5.4**.
- **Authentication Password:** Enter the **Communication Profile Password** from **Section 5.2**.
- **Outbound Proxy (optional):** Enter the IP address of IP Office 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;"> <p>▼ SIP Settings</p> <p>▶ Audio Settings</p> <p>▶ Direct Access Key Settings</p> <p>▶ Relay Settings</p> <p>▶ Time Settings</p> <p>▶ I/O Settings</p> <p>▶ Keyboard Settings</p> <p>▶ Script Configuration</p> <p>▶ Script Events</p> <p>▶ Script Upload</p> <p>▶ Audio Messages</p> <p>▶ Certificates</p> </div> <div style="width: 80%; padding: 10px;"> <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">8352001</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">8355001</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 SRTP:</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>   <h3>Call Settings</h3> <table border="1"> <thead> <tr> <th>Description</th> <th colspan="2">Configuration</th> </tr> </thead> <tbody> <tr> <td>Enable Auto Answer:</td> <td colspan="2"><input checked="" type="checkbox"/></td> </tr> <tr> <td>Auto Answer Delay:</td> <td>0</td> <td>seconds. Max 30 seconds.</td> </tr> <tr> <td>Press and Hold Time:</td> <td>0</td> <td>seconds. Max 60 seconds. Defines how long a DAK key/Input must be pressed before the call is established.</td> </tr> <tr> <td>Max Ringing Time:</td> <td>120</td> <td>How long a call can be ringing before hanging up.</td> </tr> <tr> <td>Max Conversation Time:</td> <td>3600</td> <td>How long a call can be in conversation before hanging up.</td> </tr> <tr> <td>Max Queued Time:</td> <td>20</td> <td>How long a call can be queued before hanging up.</td> </tr> <tr> <td>Max Queued Calls:</td> <td>5</td> <td>How many incoming calls can be queued. Max 5.</td> </tr> <tr> <td>Dialing Method:</td> <td colspan="2">Enbloc Dialing ▼</td> </tr> <tr> <td>Enbloc Dialing Timeout:</td> <td colspan="2">No Timeout ▼</td> </tr> <tr> <td>DTMF method:</td> <td colspan="2">SIP INFO ▼</td> </tr> </tbody> </table> </div> </div>										Description	Configuration		Display Name:	TFIE-1		Directory Number (SIP ID):	8352001		Server Domain (SIP):	devconnect.local		Backup Domain (SIP):			Backup Domain 2 (SIP):			Registration Method:	Parallell ▼		Authentication User Name:	8355001		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 SRTP:	<input type="checkbox"/>		TLS Private Key:	turbine_server_sha256.key ▼		Description	Configuration		Enable Auto Answer:	<input checked="" 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 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 Queued Time:	20	How long a call can be queued before hanging up.	Max Queued Calls:	5	How many incoming calls can be queued. Max 5.	Dialing Method:	Enbloc Dialing ▼		Enbloc Dialing Timeout:	No Timeout ▼		DTMF method:	SIP INFO ▼	
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- **Outbound Proxy (optional):** Enter the IP address of IP Office 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																																						
<div style="display: flex;"> <div style="width: 20%; border: 1px solid gray; padding: 5px;"> <p>▼ SIP Settings</p> <p>▶ Audio Settings</p> <p>▶ Direct Access Key Settings</p> <p>▶ Relay Settings</p> <p>▶ Time Settings</p> <p>▶ I/O Settings</p> <p>▶ Keyboard Settings</p> <p>▶ Script Configuration</p> <p>▶ Script Events</p> <p>▶ Script Upload</p> <p>▶ Audio Messages</p> <p>▶ Certificates</p> </div> <div style="width: 80%; padding: 5px;"> <h3>Account Settings</h3> <table border="1"> <thead> <tr> <th>Description</th> <th>Configuration</th> </tr> </thead> <tbody> <tr> <td>Display Name:</td> <td>TFIE-1</td> </tr> <tr> <td>Directory Number (SIP ID):</td> <td>8352001</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 (min. 60 seconds)</td> </tr> <tr> <td>Outbound Proxy [optional]:</td> <td>10.10.16.35 Port: 5061</td> </tr> <tr> <td>Outbound Backup Proxy [optional]:</td> <td>Port: 5060</td> </tr> <tr> <td>Outbound Backup Proxy 2 [optional]:</td> <td>Port: 1</td> </tr> <tr> <td>Outbound Transport:</td> <td>TLS ▼</td> </tr> <tr> <td>SIP Scheme:</td> <td>sips ▼ Using sips forces all proxies to also use TLS</td> </tr> <tr> <td>RTP Encryption:</td> <td>srtp_encryption ▼</td> </tr> <tr> <td>SRTP Crypto Type:</td> <td>AES_CM_128_HMAC_SHA1_80 ▼</td> </tr> <tr> <td>Use Unencrypted SRTP:</td> <td><input type="checkbox"/></td> </tr> <tr> <td>TLS Private Key:</td> <td>turbine_server_sha256.key ▼</td> </tr> </tbody> </table> </div> </div>						Description	Configuration	Display Name:	TFIE-1	Directory Number (SIP ID):	8352001	Server Domain (SIP):	devconnect.local	Backup Domain (SIP):		Backup Domain 2 (SIP):		Registration Method:	Parallel ▼	Authentication User Name:	8279999	Authentication Password:	*****	Register Interval:	600 (min. 60 seconds)	Outbound Proxy [optional]:	10.10.16.35 Port: 5061	Outbound Backup Proxy [optional]:	Port: 5060	Outbound Backup Proxy 2 [optional]:	Port: 1	Outbound Transport:	TLS ▼	SIP Scheme:	sips ▼ Using sips forces all proxies to also use TLS	RTP Encryption:	srtp_encryption ▼	SRTP Crypto Type:	AES_CM_128_HMAC_SHA1_80 ▼	Use Unencrypted SRTP:	<input type="checkbox"/>	TLS Private Key:	turbine_server_sha256.key ▼
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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:	- ▾
<b>Remote Digit For Timed Relay On:</b>	<b>6 ▾</b>
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**.

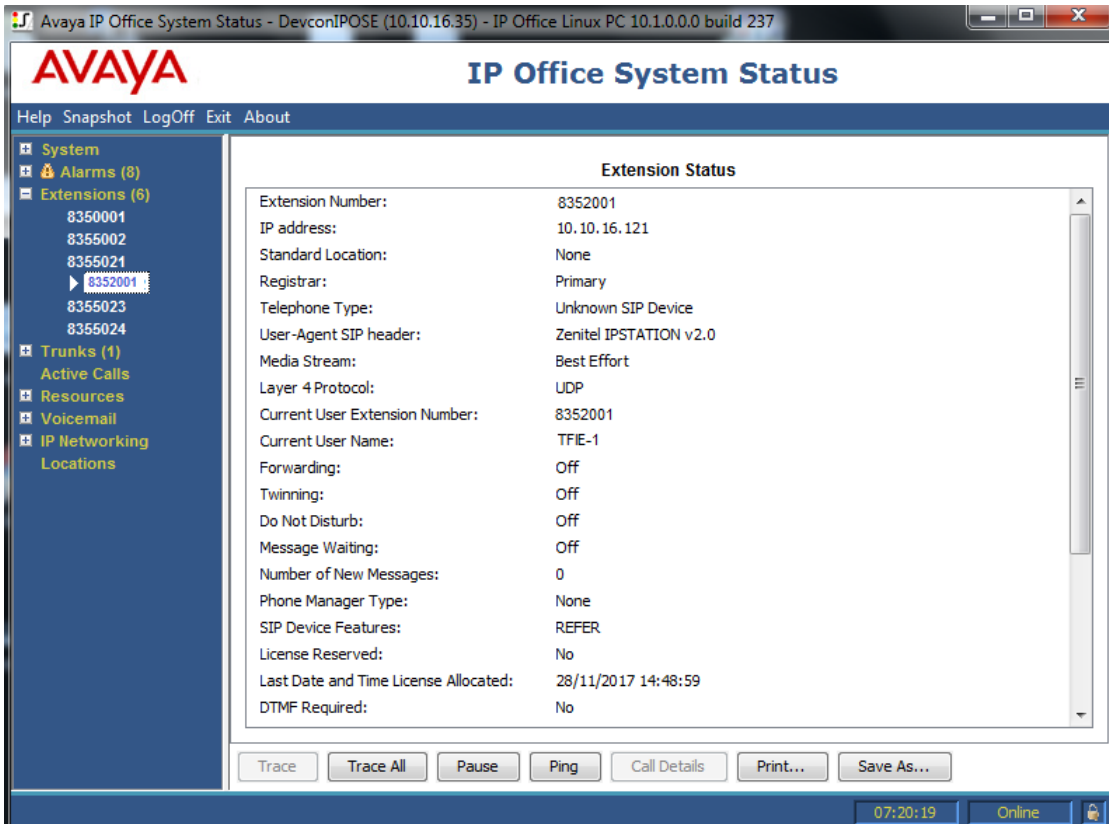
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## 7. Verification Steps

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

### 7.1. Verify Avaya IP Office SIP Endpoint Registration

Open the IP Office System Status application and click on **Extensions**. Verify that Turbine endpoints are successfully registered as shown below.



The screenshot displays the Avaya IP Office System Status application interface. The title bar indicates the application is running on a Linux PC with IP address 10.10.16.35 and build 237. The main window features the Avaya logo and the title "IP Office System Status". A navigation menu on the left includes options like System, Alarms (8), Extensions (6), Trunks (1), Active Calls, Resources, Voicemail, IP Networking, and Locations. The "Extensions (6)" menu item is expanded, showing a list of extension numbers: 8350001, 8355002, 8355024, 8352001 (highlighted), 8355023, and 8355024. The main content area displays the "Extension Status" for extension 8352001, with the following details:

Extension Status	
Extension Number:	8352001
IP address:	10.10.16.121
Standard Location:	None
Registrar:	Primary
Telephone Type:	Unknown SIP Device
User-Agent SIP header:	Zenitel IPSTATION v2.0
Media Stream:	Best Effort
Layer 4 Protocol:	UDP
Current User Extension Number:	8352001
Current User Name:	TFIE-1
Forwarding:	Off
Twinning:	Off
Do Not Disturb:	Off
Message Waiting:	Off
Number of New Messages:	0
Phone Manager Type:	None
SIP Device Features:	REFER
License Reserved:	No
Last Date and Time License Allocated:	28/11/2017 14:48:59
DTMF Required:	No

At the bottom of the application, there are several control buttons: Trace, Trace All, Pause, Ping, Call Details, Print..., and Save As... The status bar at the bottom right shows the time as 07:20:19 and the extension status as Online.

## 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.

The screenshot displays the Turbine web interface with the following structure:

- Navigation tabs: Station Main, SIP Configuration, Station Administration, Advanced SIP, Advanced Network.
- Left sidebar: Station Information (selected), Main Settings.
- Main content area:
  - TFIE-1 Information** table:

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:	<a href="#">List</a>
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** table:

Description	Status
Station Mode:	SIP
Display Name:	TFIE-1
Directory Number (SIP ID):	8352001
Server Domain (SIP):	devconnect.local, Registered - Thu Jan 22 17:08:53 1970
Backup Domain (SIP):	
Backup Domain 2 (SIP):	
Outbound Proxy:	10.10.16.35

## 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 IP Office. 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.

These documents form part of the Avaya official technical reference documentation suite. Further information may be obtained from <http://support.avaya.com> or from your Avaya representative.

[1] *Avaya IP Office Manager 10.1, Document 15-601011, Issue 1, June 2017*

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

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

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