



Avaya Solution & Interoperability Test Lab

Application Notes for Avaya IP Office Release 10 with AT&T IP Toll Free Service – Issue 1.0

Abstract

These Application Notes describe the steps for configuring Avaya IP Office Release 10 with the AT&T IP Toll Free service using AVPN or MIS/PNT transport connections.

The AT&T IP Toll Free service is a managed Voice over IP (VoIP) communications solution providing toll-free services over SIP trunks for business customers.

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.

AT&T is a member of the Avaya DevConnect Service Provider program. 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 steps for configuring Avaya IP Office R10 with the AT&T IP Toll Free service using **AVPN** or **MIS/PNT** transport connections.

Avaya IP Office is a versatile communications solution that combines the reliability and ease of a traditional telephony system with the applications and advantages of an IP telephony solution. This converged communications solution can help businesses reduce costs, increase productivity, and improve customer service

The AT&T IP Toll Free service is a managed Voice over IP (VoIP) communications solution providing toll-free services over SIP trunks for business customers. The AT&T Toll Free service utilizes AVPN¹ or MIS/PNT² transport services.

Note – The AT&T IP Toll Free service will be referred to as IPTF in the remainder of this document.

Note – The solution described in these application notes also applies to the AT&T Business in a Box service.

2. General Test Approach and Test Results

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.

The interoperability compliance testing focused on verifying inbound and outbound call flows between IPTF and the Customer Premises Equipment (CPE) containing the Avaya IP Office Release 10 (see **Section 3.2** for call flow examples).

The test environment described in these Application Notes consisted of:

- A simulated enterprise with Avaya IP Office 10, Avaya SIP, H.323 and Analog telephones, as well as a fax machine emulator (Ventafax).
- Laboratory versions of the IPTF service, to which the simulated enterprise was connected via AVPN/MIS transport.

¹ AVPN uses compressed RTP (cRTP).

² MIS/PNT does not support cRTP.

2.1. Interoperability Compliance Testing

The compliance testing was based on a test plan provided by AT&T, for the functionality required for certification as a solution supported on the IPTF network. Calls were made from the PSTN across the IPTF test network, to the CPE.

The interoperability compliance testing focused on verifying inbound call flows (see **Section 3.2**) between Avaya IP Office and the IPTF service.

The compliance testing was based on a test plan provided by AT&T, for the functionality required for certification as a solution supported on the AT&T network.

The following SIP trunking VoIP features were tested with the IPTF service:

- Incoming calls from PSTN, routed by the IPTF service, to Avaya IP Office. These calls are via the Avaya IP Office SIP Line and may be generated/answered by Avaya SIP telephones/Softphones, H.323 telephones, Analog telephones, Analog fax machines or via Hunt Groups. Coverage to Voicemail Pro, and Voicemail Pro auto-attendant applications, were also used.
- Inbound fax using T.38 or G.711, and G3 or SG3 endpoints.
- Proper disconnect when the caller abandons a call before answer, and when the Avaya IP Office party or the PSTN party terminates an active call.
- Proper busy tone heard when an Avaya IP Office user calls a busy PSTN user, or a PSTN user calls a busy Avaya IP Office user (i.e., if no redirection was configured for user busy conditions).
- SIP OPTIONS monitoring of the health of the SIP trunk. In the reference configuration Avaya IP Office sent OPTIONS to the IPTF service Border Element and AT&T responded with *405 Method Not Allowed* (which is the expected response). That response is sufficient for Avaya IP Office to consider the connection up.
- Incoming calls using the G.729A and G.711 ULAW codecs.
- Long duration calls.
- DTMF transmission (RFC 2833) for successful voice mail navigation, including navigation of a simple auto-attendant application configured on Voicemail Pro, as well as IPTF DTMF generated features.
- Telephony features such as call waiting, hold, transfer, and conference.
- Verify reception of IPTF SIP Multipart/NSS headers, including SDP and XML content.
- AT&T IP Toll Free features such as Legacy Transfer Connect and Alternate Destination Routing.

2.2. Test Results

The test objectives stated in **Section 2.1**, with limitations as noted below, were verified.

1. **Avaya IP Office only supports a packet size (ptime) of 20 msecs, and therefore does not specify a ptime value in the SIP SDP (in either requests or responses).**
 - Although no issues were found during testing, AT&T recommends that for maximum customer bandwidth utilization, a ptime value of 30 should be specified.
2. **IP Toll Free ADR Call Redirection feature based on SIP error code response.**

Upon receiving an error response, IPTF service can be configured to invoke ADR Call Redirection. The following error codes were producible by the reference configuration and tested successfully; 408 Request Timeout, 480 Temporarily Unavailable, 486 Busy Here, and 503 Service Unavailable. The following error codes are also supported by IPTF service, but were not producible by the reference configuration, and thus not tested; 500 Server Internal Error, 504 Server Timeout, and 600 Busy Everywhere.
3. **Enhanced CID – NSS feature.** The inbound calls to Avaya IP Office are not exercising the Enhanced CID feature. Although Avaya IP Office is accepting SIP Multipart/NSS headers, it is neither passing nor acting upon it. It is simply being ignored.

2.3. Support

AT&T customers may obtain support for the AT&T IP Toll Free service by calling (800) 325-5555.

Avaya customers may obtain documentation and support for Avaya products by visiting: <http://support.avaya.com>. In the United States, (866) GO-AVAYA (866-462-8292) provides access to overall sales and service support menus. Customers may also use specific numbers (provided on <http://support.avaya.com>) to directly access specific support and consultation services based upon their Avaya support agreements.

3. Reference Configuration

Note – Documents used to provision the test environment are listed in **Section 9**. References to these documents are indicated by the notation [x], where x is the document reference number.

The reference configuration used in these Application Notes is shown in **Figure 1** on the next page and consists of the following components:

- An Avaya Server Edition Primary server with an IP500 V2 Expansion System for fax and digital endpoint support. The single Server Edition Primary server provides IP Office Server Edition, Voicemail Pro, and Avaya one-X® Portal for IP Office.

- Avaya IP Office Server Edition provides the voice communications services for a particular enterprise site.
- Voicemail Pro provides the voice messaging capabilities in the reference configuration. This solution is extensible to the standalone IP Office 500V2 Embedded Voicemail as well.
- Avaya “desk” telephones are represented with an Avaya 1616 H.323 set, an Avaya 9611 H.323 set, an Avaya 9508 Digital set, an Avaya 1140E SIP set, as well as Avaya Communicator for Windows (SIP). Fax endpoints are represented by PCs running Ventafax emulation software connected by modem to an Avaya IP Office 500 V2 analog port.
- In the reference configuration, Avaya IP Office interface “LAN 1” is connected to the private CPE, and interface “LAN 2” is connected to the public network and AT&T.
- The AT&T IPTF service requires the following SIP trunk network settings between the Avaya IP Office LAN 2 (SIP Trunk) interface and the IPTF Border Element:
 - UDP transport using port 5060
 - RTP port ranges 16384-32767
- AT&T provided the inbound and outbound access numbers (DID and DNIS) used in the reference configuration. Note that the IPTF service may deliver various digit lengths in the SIP Invite R-URI depending on the circuit order provisioning. In the reference configuration, the IPTF service delivered 15 digits.

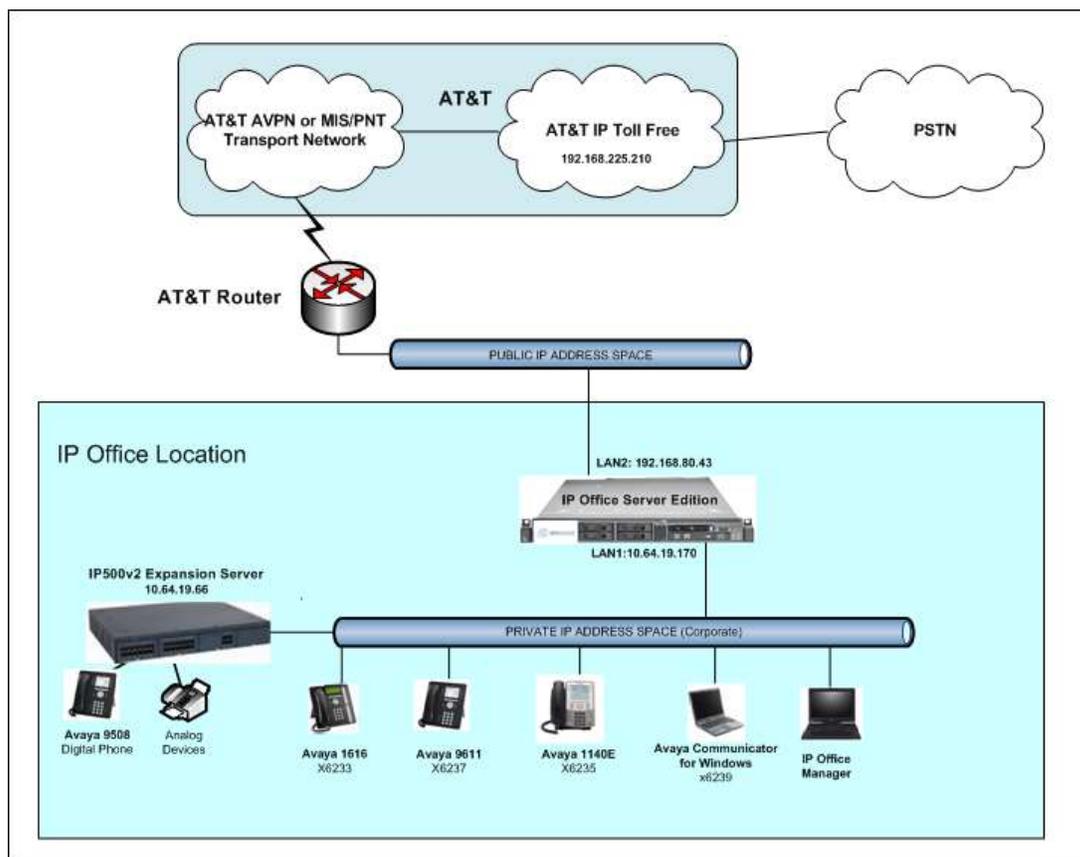


Figure 1: Reference Configuration

3.1. Illustrative Configuration Information

The specific values listed in **Table 1** below and in subsequent sections are used in the reference configuration described in these Application Notes, and are for illustrative purposes only. Customers must obtain and use the values based on their own specific configurations.

Note – The Avaya IP Office Server Edition LAN 2 interface is defined as the SIP trunk (see **Section 5.3.3**) and communicates with AT&T Border Elements (BEs) located in the AT&T IPTF network. For security reasons, the IP addresses of the AT&T BEs are not included in this document. However as placeholders in the following configuration sections, the IP addresses **192.168.80.42** (Avaya IP Office Server Edition LAN 2 address), and **192.168.225.210** (AT&T BE IP address), are specified. In addition, AT&T DID/DNIS numbers shown in this document are examples as well. AT&T Customer Care will provide the actual Border Element IP addresses and DID/DNIS numbers as part of the IPTF provisioning process.

Component	Illustrative Value in these Application Notes
Avaya IP Office Server Edition	
Private network LAN1 interface	10.64.19.170
Public network LAN2 interface	192.168.80.42
Avaya IP Office Expansion System (IP500 V2)	
Private network LAN1 interface	10.64.19.66
AT&T IPTF Service	
Border Element IP Address	192.168.225.210

Table 1: Illustrative Values Used in these Application Notes

3.2. Call Flows

To understand how inbound AT&T IPTF service calls are handled by Avaya IP Office, two basic call flows are described in this section.

3.2.1. Inbound

The first call scenario illustrated in the figure below is an inbound AT&T IPTF service call that arrives on Avaya IP Office, which in turn routes the call to a hunt group, phone or a fax endpoint.

1. A PSTN phone originates a call to an IPTF service number.
2. The PSTN routes the call to the AT&T IPTF service network.
3. The AT&T IPTF service routes the call to Avaya IP Office.
4. Avaya IP Office applies any necessary digit manipulations based upon the DID and routes the call to a hunt group, phone or a fax endpoint.

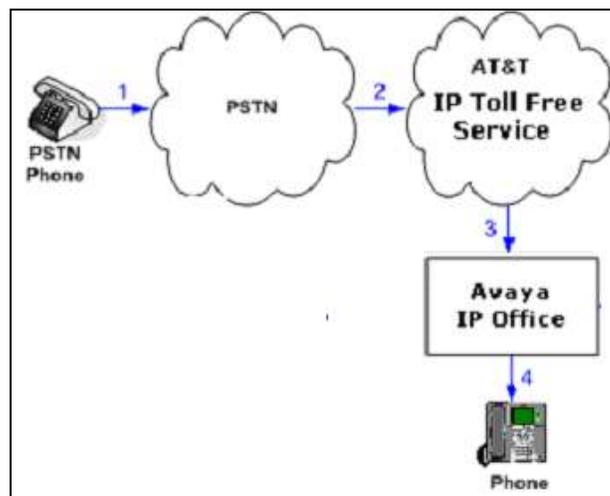


Figure 2: Inbound AT&T IPTF Call

3.2.2. Coverage to Voicemail

The call scenario illustrated in the figure below is an inbound call that is covered to Voicemail. In the reference configuration, the Voicemail system used is Voicemail Pro, running on Avaya Server Edition Primary server.

1. Same as the first call scenario in **Section 3.2.1**.
2. The Avaya IP Office phone does not answer the call, and the call covers to Voicemail Pro.

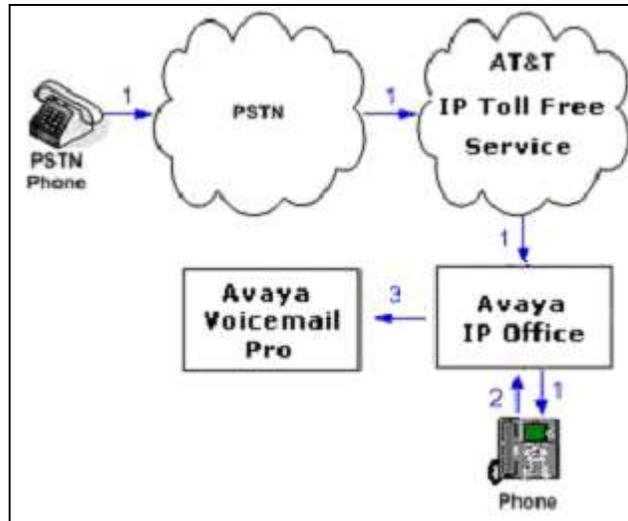


Figure 3: Coverage to Voicemail (Voicemail Pro)

4. Equipment and Software Validated

The following equipment and software was used for the reference configuration described in these Application Notes.

Equipment/Software	Release/Version
Avaya IP Office Server Edition <ul style="list-style-type: none">▪ IP Office▪ Voicemail Pro▪ Avaya WebRTC Gateway▪ Avaya one-X® Portal for IP Office	10.0.0.0 build 550 10.0.0.0 build 469 10.0.0.0 build 140 10.0.0.0 build 980
Avaya IP Office 500 V2 <ul style="list-style-type: none">▪ Avaya IP Office TCM 8▪ Avaya IP Office COMBO6210/ATM4	10.0.0.0 build 550 10.0.0.0 build 550
Avaya IP Office Server Edition Manager	10.0.0.0 build 550
Avaya Communicator for Windows (SIP)	2.1.3.237
Avaya 9641G (H.323) IP Deskphone	6.6229
Avaya 1616 (H.323) Telephone	Ha1616ua1_390A.bin
Avaya 1140E (SIP) Telephone	04.04.23
Avaya 9508 Digital Telephone	0.55
Analog Fax device	Ventafax 6.3

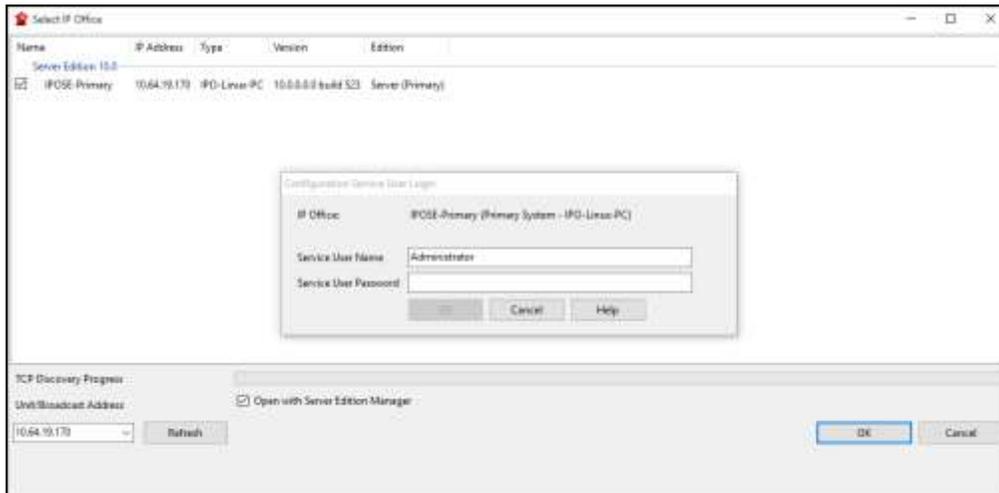
Table 2: Equipment and Software Versions

Note – Compliance Testing is applicable when the tested solution is deployed with a standalone IP Office 500 V2, and also when deployed with all configurations of IP Office Server Edition. IP Office Server Edition requires an Expansion IP Office 500 V2 to support analog or digital endpoints or trunks.

5. Avaya IP Office Primary Configuration

Note – This section describes attributes of the reference configuration, but is not meant to be prescriptive. In the following sections, only the parameters that are highlighted in **bold** text are applicable to the reference configuration. Other parameter values may or may not match based on local configurations. Many forms contain multiple tabs. Only those tabs with provisioning related to the reference configuration are discussed. Any other tab/form should be considered default values. Additionally, the screen shots referenced in these sections may not be the complete form.

IP Office is configured via the IP Office Manager program. For more information on IP Office Manager, consult reference [3]. From the IP Office Manager PC, select **Start** → **All Apps** → **IP Office** → **Manager** to launch the Manager application. Navigate to **File** → **Open Configuration** (not shown), select the proper Avaya IP Office system from the pop-up window, and log in using the appropriate credentials.

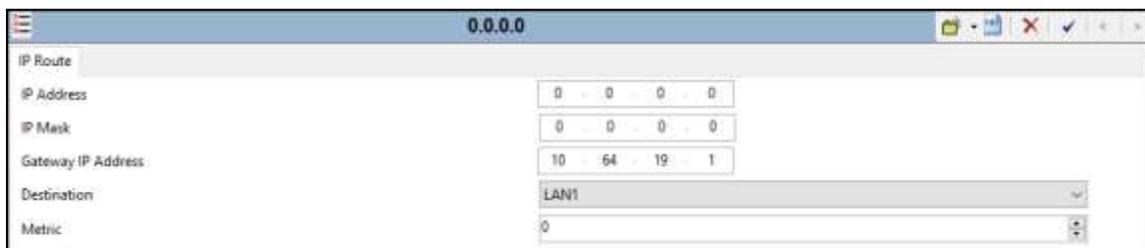


On Server Edition systems, the Solution View screen will appear, similar to the one shown on next page. If the left navigation pane does not immediately appear, click on the **Configuration** link as highlighted below. In the reference configuration, IP users registered to the Primary server and failover to the Secondary server. Digital and Analog users are configured on the Expansion System. A SIP trunk to the AT&T IPTF service is configured on the Primary server. Clicking the “plus” sign next to the Primary server system name, e.g., **IPOSE-Primary**, on the left navigation pane will expand the menu on this server.

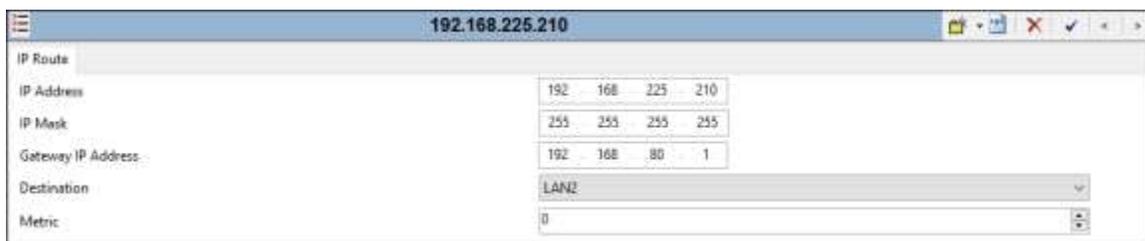


5.1. IP Route

In the sample configuration, the Primary server LAN1 port is physically connected to the CPE network. The default gateway for this network is **10.64.19.1**.



The Primary server LAN2 port is physically connected to the AT&T network and has a default gateway of **192.168.80.1**. To add an IP Route in IP Office, right-click **IP Route** from the Navigation pane, and select **New** (not shown). To view or edit an existing route, select **IP Route** from the Navigation pane, and select the appropriate route from the Group pane. The following screen shows the Details pane with the relevant route using **Destination LAN2** (to AT&T).



5.2. Licensing

In the sample configuration, **IPOSE-Primary** was used as the system name of the Primary Server, **IPOSE-Secondary** was used as the system name of the Secondary Server and **IP500 Expansion** was used as the system name of the Expansion System. All navigation described in the following sections (e.g., **License**) appears as submenus underneath the system name in the Navigation Pane.

The configuration and features described in these Application Notes require the IP Office system to be licensed appropriately. If a desired feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative.

To verify that there is a SIP Trunk Channels License with sufficient capacity, click **License** in the Navigation pane. Confirm a valid **SIP Trunk Channels** license with sufficient **Instances** (trunk channels).

The screenshot shows the 'License' tab for a 'Remote Server'. The 'License Mode' is 'WebLM Normal' and the 'Licensed Version' is '10.0'. A table lists various features and their instances:

Feature	Instances	Status	Expiration Date	Source
Additional Voicemail Pro Ports	152	Valid	Never	WebLM
VMPro TTS Professional	1	Valid	Never	WebLM
Power User	6	Valid	Never	WebLM
Avaya IP endpoints	0	Valid	Never	WebLM
SIP Trunk Channels	50	Valid	Never	WebLM
CTI Link Pro	1	Valid	Never	WebLM
Server Edition R10	1	Valid	Never	WebLM
Web Collaboration	5	Valid	Never	WebLM
UMS Web Services	1	Valid	Never	WebLM
Basic User	5	Valid	Never	WebLM

On the **Remote Server** tab, set the **SIP Trunk Sessions** to a value sufficient for the number of simultaneous SIP trunk calls needed.

The screenshot shows the 'License' tab for a 'Remote Server' with the 'Reserved Licenses' section highlighted. The 'SIP Trunk Sessions' value is set to 50. Other license settings are also visible:

License Feature	Value	Server Edition	Value
SIP Trunk Sessions	50	Server Edition	1
SM Trunk Sessions	0	Avaya IP Endpoints	0
Voicemail Pro Ports	152	3rd Party IP Endpoints	0
VMPro Recordings Administrators	0	Receptionist	0
VMPro TTS Professional	1	Basic User	3
CTI Link Pro	1	Office Worker	0
UMS Web Services	1	Power User	0
Mac Softphones	0	Avaya Softphone	0
Avaya Contact Center Select	0	Web Collaboration	3
Third Party Recorder	0		

5.3. System Settings

This section illustrates the configuration of system settings. Select **System** in the Navigation pane to configure these settings. The subsection order corresponds to a left to right navigation of the tabs in the Details pane for System settings. For all of the following configuration sections, the **OK** button (not shown) must be selected in order for any changes to be saved.

5.3.1. LAN 1 Tab

In the reference configuration, LAN1 was used to connect the Avaya IP Office to the CPE network (see **Section 3**).

5.3.1.1 LAN 1 – LAN Settings Tab

To view or configure the LAN 1 IP address, select the **LAN 1 → LAN Settings** tab, and enter the following:

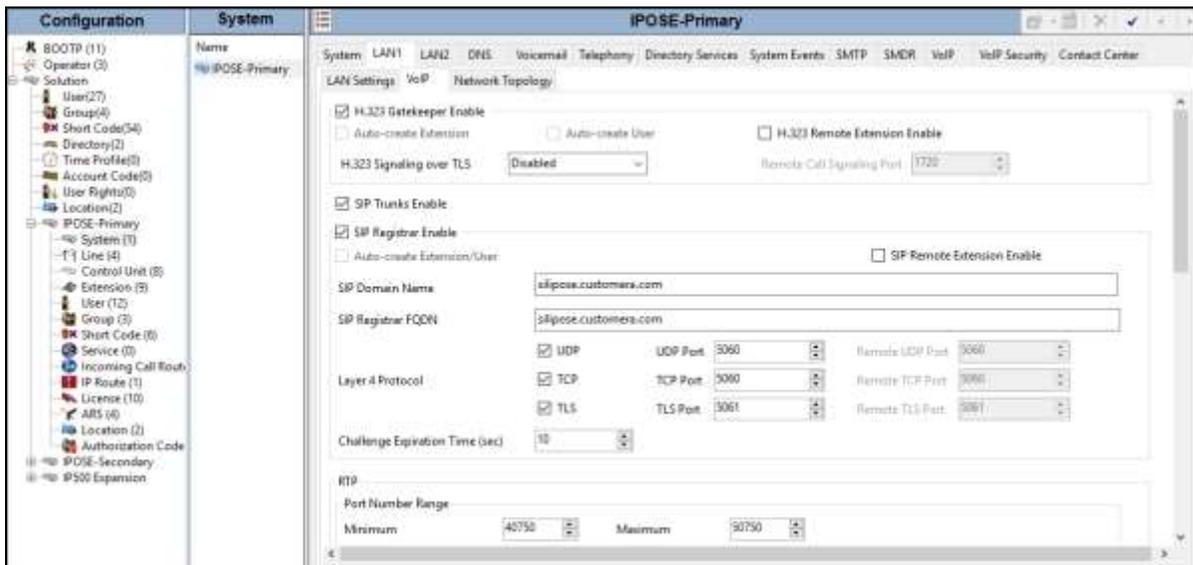
- **IP Address:** Set to **10.64.19.170** as specified in the reference configuration.



5.3.1.2 LAN 1 - VoIP Tab

Select the **LAN1 → VoIP** tab as shown in the following screen. The following settings were used in the reference configuration:

- The **H323 Gatekeeper Enable** box is checked to allow the use of Avaya IP Telephones using the H.323 protocol, such as the Avaya 1600-Series Telephones used in the reference configuration.
- The **SIP Registrar Enable** box is checked to allow Avaya 11xx (SIP) and Avaya IP Office Softphone (SIP) usage.
- The **SIP Domain Name** used in the reference configuration is **silipose.customera.com**.
- The **SIP Registrar FQDN** used in the reference configuration is **silipose.customera.com**.
- In the **Layer 4 Protocol** section, select **UDP/5060, TCP/5060, TLS/5061**.
- Let all other values default.

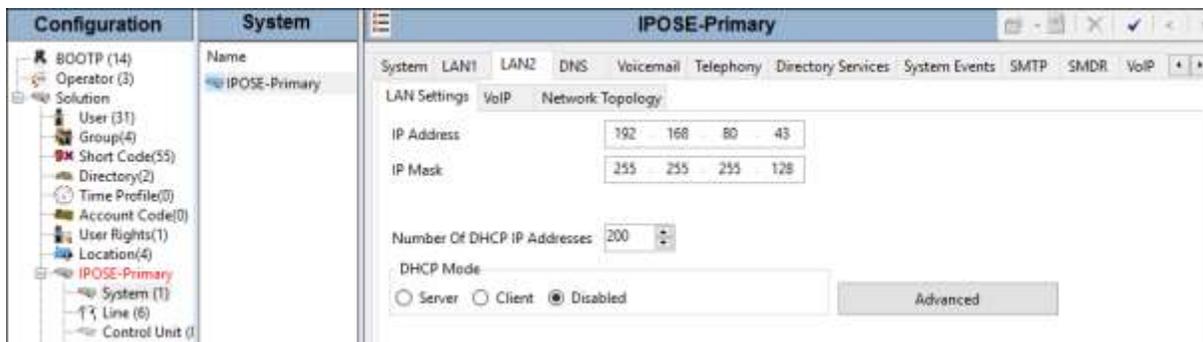


5.3.2. LAN 2 Tab

The LAN 2 interface is used for the SIP trunk connection to AT&T. In the sample configuration, LAN2 is used to connect the IP Office to the AT&T network. To view or configure the **IP Address** of LAN2, select the **LAN2** tab followed by the **LAN Settings** tab. As shown in **Figure 1**, the IP Address of the IP Office, known to AT&T, is **192.168.80.43**. **DHCP Mode** is set to **Disabled** since DHCP is unnecessary towards AT&T. Other parameters on this screen may be set according to customer requirements

5.3.2.1 LAN 2 - LAN Settings Tab

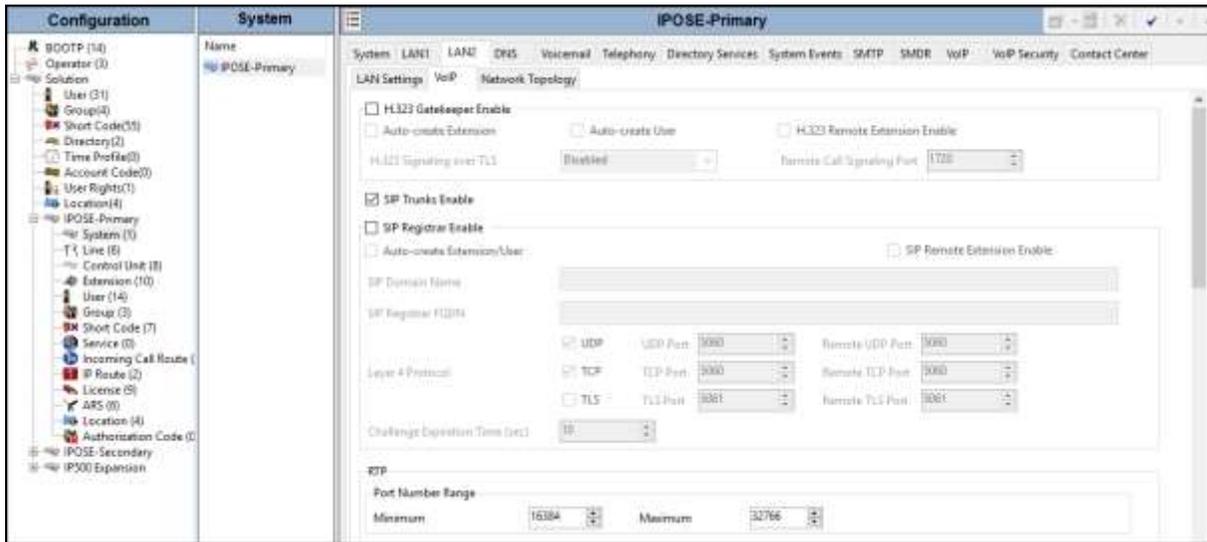
- **IP Address:** In the reference configuration the IP Office public address is **192.168.80.43**.
- Other parameters on this screen are set to their defaults.



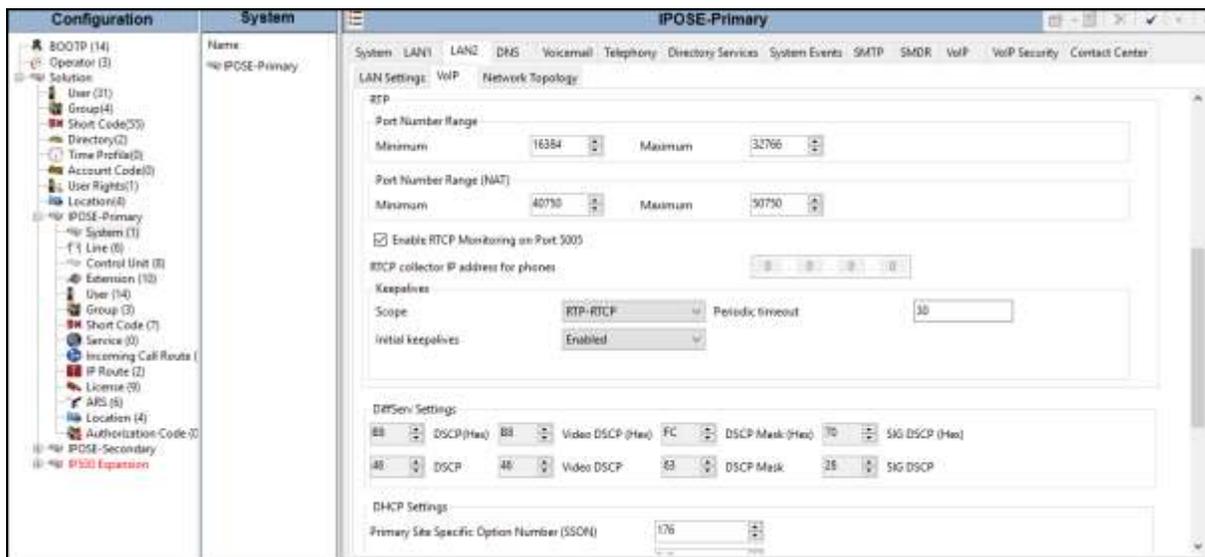
5.3.2.2 LAN 2 - VoIP Tab

Select the **LAN2 → VoIP** tab as shown in the following screen. The following settings were used in the reference configuration

- Select the **SIP Trunks Enabled** option.
- **RTP Port Number Range:** The AT&T IPTF service requires that the RTP use the port range 16384 to 32767.
 - **16384** entered in the **Port Range (Minimum)** field.
 - **32766** entered in the **Port Range (Maximum)** field, as this field requires even numbers.



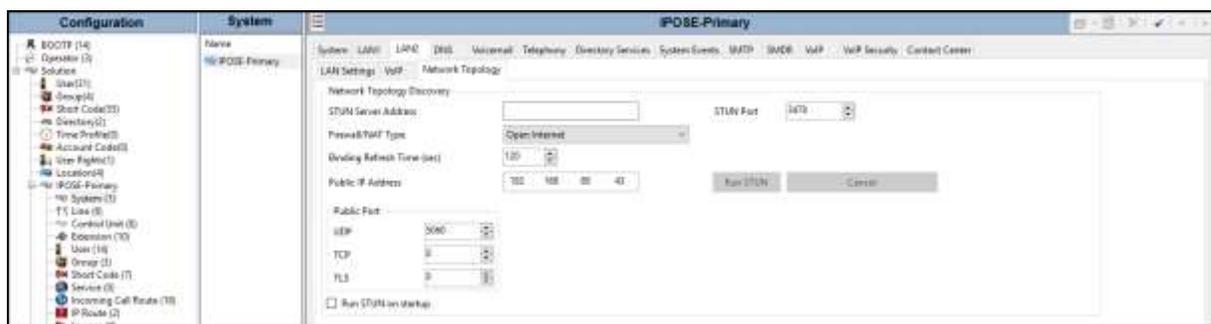
- To prevent possible issues with network firewalls closing idle RTP channels, it is recommended that **RTP Keepalives** are enabled. Scrolling down to the bottom of the form, enter the following:
 - **Scope:** Select **RTP-RTCP**
 - **Periodic Timeout:** Enter **30**
 - **Initial keepalives:** Select **Enabled**
- Other parameters on this screen are set to the defaults.



5.3.2.3 LAN 2 - Network Topology Tab

- The **Firewall/NAT Type** is set to **Open Internet** in the reference configuration.

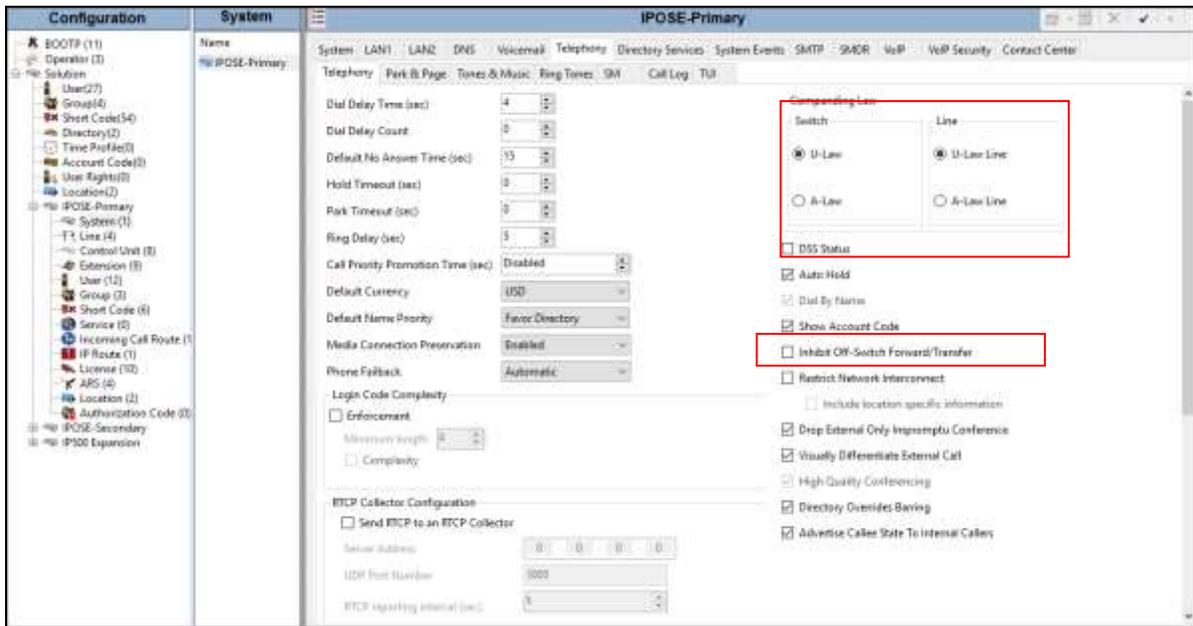
Note – the **Firewall/NAT Type** parameter may need to be set differently, depending if firewall and/or Network Address Translation (NAT) devices are used at the customer premise.
- **Binding Refresh Time:** This field specifies how often IP Office will issue a SIP OPTIONS message to check the SIP trunk connection status to AT&T. In the reference configuration, **120** secs is specified.
- **Public IP Address:** In the reference configuration the IP Office public address is **192.168.80.43**.
- Set the **Public Port** to **UDP/5060**.



5.3.3. Telephony Tab

To view or change telephony settings, select the **Telephony** tab and **Telephony** sub-tab as shown below. The settings presented here simply illustrate the values used in the reference configuration and are not intended to be prescriptive.

- Uncheck the **Inhibit Off-Switch Forward/Transfer** box. This is so that call forwarding and call transfer to PSTN destinations via the AT&T IPTF service can be tested.
- Set the **Companding Law** parameters to **U-LAW** as is typical in North America.
- Default values are used in the other fields.



5.3.4. VoIP Tab

On the left, observe the list of **Available Codecs**. By selecting codecs in this column, they will appear in the **Default Codec Selection** → **Unused** column. Codecs may be selected from the **Unused** list and moved to the **Selected** column by use of the >>> button, thereby making the selected codecs available in other screens where codec configuration may be performed (e.g., SIP Lines and Extensions).

The up and down arrow buttons are used to order the selected codecs. By default, all IP (SIP and H.323) lines and extensions will assume the system default **Selected** codec list, unless configured otherwise for the specific SIP Line or extension (see the note below).

- Populate the **Selected** column with **G.711 ULAW 64K** as the first codec and **G.729(a) 8K CS-ACELP** as the second codec.
- In the **RFC2833 Default Payload** setting field, specify **100**, which is the recommended value for AT&T interoperability.

Note – In the reference configuration, the Extension codec lists (see **Section 5.6.2**) also specify *G.711ULAW* and *G.729(a)* (in that order), and the SIP Line (see **Section 5.4.6**) offers *G.729(a)* and *G.711ULAW* (in that order). In this manner, local Avaya IP Office calls will offer G.711mu first, and SIP trunk calls will offer G.729A first.



5.4. SIP Line

The following sections describe the configuration of a SIP Line. The SIP Line terminates the CPE end of the SIP trunk to the AT&T IPTF service.

The recommended method for creating/configuring a SIP Line is to use the template associated with the provisioning described in these Application Notes. The template is an .xml file that can be used by Avaya IP Office Manager to create a new SIP Line for SIP trunking with the AT&T IPTF service. Follow the steps in **Section 5.4.2** to create a SIP Trunk from the template.

Some items relevant to a specific customer environment are not included in the template or may need to be updated after the SIP Line is created. Examples include the following:

- IP addresses
- SIP Credentials (if applicable)
- SIP URI entries
- Setting of the **Use Network Topology Info** field on the Transport tab.

Therefore, it is important that the SIP Line configuration be reviewed and updated if necessary, after the SIP Line is created via the template. The resulting SIP Line data can be verified against the manual configuration as shown in **Sections 5.4.3 – 5.4.8**.

In addition, the following SIP Line settings are not supported on Basic Edition:

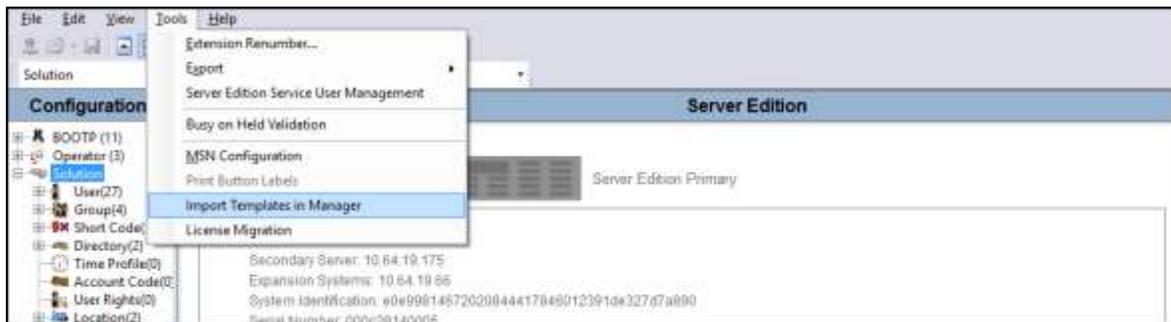
- SIL Line – Originator number for forwarded and twinning calls
- Transport – Second Explicit DNS Server
- SIP Credentials – Registration Requirements
- SIP Advanced Engineering

Alternatively, a SIP Line can be created manually. To do so, right-click **Line** in the Navigation Pane and select **New → SIP Line**. Then, follow the steps outlined in **Sections 5.4.3 – 5.4.8**.

5.4.1. Importing a SIP Line Template

Note – DevConnect generated SIP Line templates are always exported in an XML format. These XML templates do not include sensitive customer specific information and are therefore suitable for distribution. The XML format templates can be used to create SIP trunks on both IP Office Standard Edition (IP500 V2) and IP Office Server Edition systems. Alternatively, binary templates may be generated. However, binary templates include all the configuration parameters of the Trunk, including sensitive customer specific information. Therefore, binary templates should only be used for cloning trunks within a specific customer’s environment.

1. Copy a previously created template file to a location (e.g., *\temp*) on the same computer where IP Office Manager is installed.
2. Import the template into IP Office Manager. From IP Office Manager, select **Tools → Import Templates in Manager**.

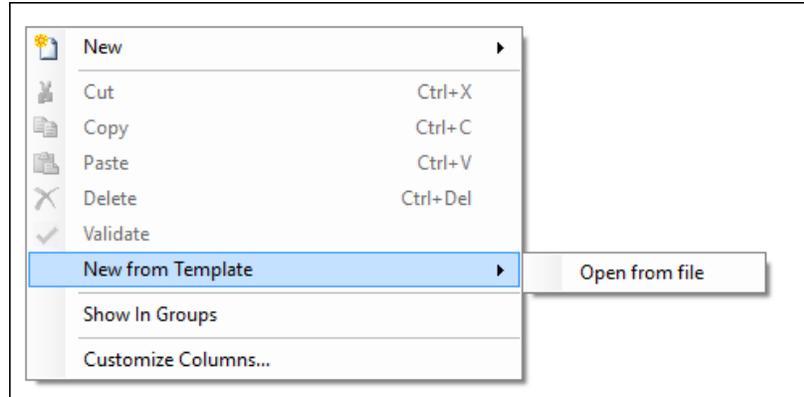


3. A folder browser will open (not shown). Select the directory used in **step 1** to store the template(s) (e.g., *\temp*). In the reference configuration, template file **IPO10TF.xml** was imported. The template files are automatically copied into the IP Office default template location, **C:\Program Files\Avaya\IP Office\Manager\Templates**.
4. After the import is complete, a final import status pop-up window will open stating success or failure.

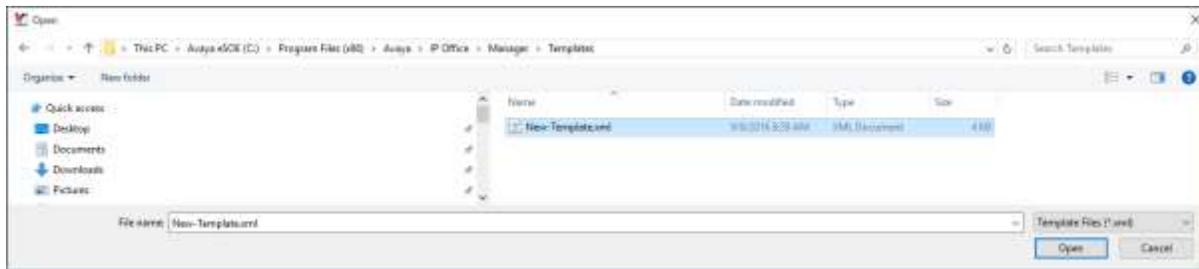


5.4.2. Creating a SIP Trunk from an XML Template

1. To create the SIP Trunk from a template, right-click on **Line** in the Navigation Pane, and hover over **New from Template**, and select **Open from file**.



Navigate to **C:\Program Files\Avaya\IP Office\Manager\Templates**. Select ***.xml** as the file type, find the template, and click **Open**.



The newly created SIP Line will appear in the Navigation pane (e.g., SIP Line 2).

Line Number	Line Type	Line SubType
1	IP Office Line	WebSocket Server SCN
3	IP Office Line	WebSocket Server SCN
2	SIP Line	

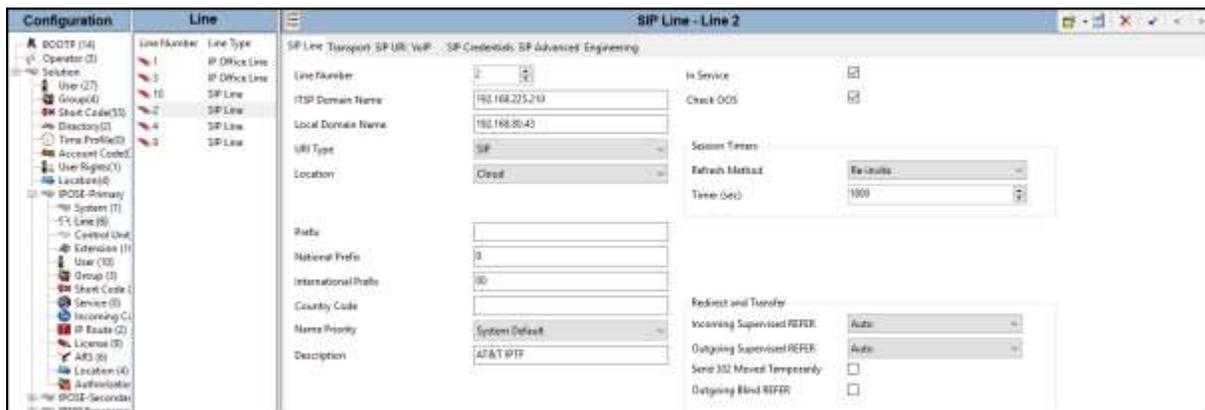
Once the SIP Line is created, verify the configuration of the SIP Line with the configuration shown in **Sections 5.4.3 – 5.4.8**.

5.4.3. SIP Line – SIP Line tab

The **SIP Line** tab is shown below for **Line Number 2**, used for the SIP Trunk to AT&T. Note, if no SIP Line exists, right click on the **Line** item in the **Navigation** pane and select **New → SIP Line** (not shown). In the reference configuration, SIP Line 2 was created. The SIP Line form is completed as follows:

- **ITSP Domain Name:** Set to the IP address of the AT&T Border Element IP address (e.g., **192.168.225.210**).

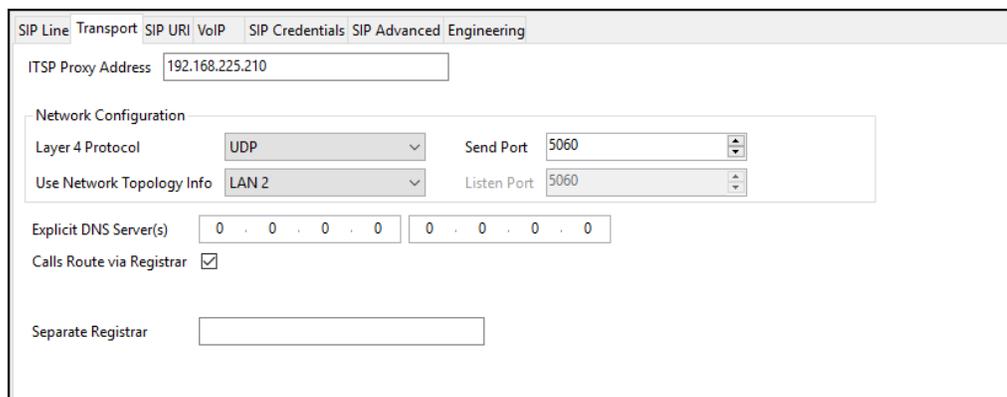
- **Local Domain Name:** Set to the IP address of the Avaya IP Office LAN2 SIP trunking interface (e.g., **192.168.80.43**).
- **In Service** and **Check OOS:** These boxes are checked (default).
- **Refresh Method:** Set to **Re-Invite**, as AT&T does not support UPDATE.
- **Incoming Supervised Refer:** Set this field to **Auto** (default).
- **Outgoing Supervised Refer:** Set this field to **Auto** (default).
- **Send 302 Moved Temporarily:** Verify this field is unchecked (default).
- **Outgoing Blind REFER:** Verify this field is unchecked (default).
- Use the default values for the other fields.
- Click **OK** (not shown).



5.4.4. SIP Line - Transport tab

Select the **SIP Line** → **Transport** tab and configure the following:

- **ITSP Proxy Address:** Set to the AT&T Border Element IP address (e.g., **192.168.225.210**).
- **Network Configuration** → **Layer 4 Protocol:** Set to **UDP**.
- **Network Configuration** → **Send Port:** Set to **5060** (default).
- **Network Configuration** → **Use Network Topology Info:** Set to **LAN 2**.
- **Calls Route via Registrar:** Verify this field is checked (default).
- Click **OK** (not shown).



5.4.5. SIP Line - SIP URI tab

Select the **SIP Line** → **SIP URI** tab. To add a new SIP URI, click the **Add...** button. At the bottom of the screen, a **New Channel** area will be opened. Configure the following:

- **Local URI, Contact, and Display Name** fields: Set these fields to **Auto**. This setting replaces the wildcard “*” used in previous releases.
- Verify **Identity, Send Caller ID, and Diversion Header**: Set to the default **None**.
- Verify **Registration**: Set to the default **0: <None>**.
- **Incoming Group**: Set to an unused group number, e.g., **3**. This value references the table created with **Incoming Call Routes** in **Section 5.7**.
- **Outbound Group**: Set to an unused group number, e.g., **3**.
- **Max Sessions**: In the reference configuration this was set to **10**. This sets the maximum number of simultaneous calls that can use the URI before Avaya IP Office returns busy to any further calls.
- Click **OK**.

URI	Groups	Local URI	Contact	Display Name	Identity	Header	Originator Number	Send Caller ID	Diversion Header	Credent
1	3 3	Auto	Auto	Auto	None	PAI		None	None	0: <None>

Edit URI

Local URI: Auto

Contact: Auto

Display Name: Auto

Identity: None

Header: P Asserted ID

Forwarding And Twinning

Originator Number:

Send Caller ID: None

Diversion Header: None

Registration: 0: <None>

Incoming Group: 3

Outgoing Group: 3

Max Sessions: 10

OK

Cancel

- To edit an existing entry, click an entry in the list and click the **Edit** button.
- When all SIP URI entries have been added or edited, click **OK** at the bottom of the screen (not shown).

5.4.6. SIP Line - VoIP tab

Select the **SIP Line** → **VoIP** tab and enter the following:

- The **Codec Selection** drop-down box → **System Default** will list all available codecs. In the reference configuration, **Custom** was selected and **G729(a) 8K CS-ACELP**, and **G.711 ULAW 64K** were specified. This causes Avaya IP Office to include these codecs in the Session Description Protocol (SDP) offer, and in the order specified. Note that in the reference configuration G.729A is set as the preferred codec on the SIP trunk to the AT&T IPTF network (see the note below regarding IPTF and Silence Suppression).
- T.38 fax was used in the reference configuration. Set the **Fax Transport Support** drop-down menu to **T38**. G.711 fax also worked in the reference configuration (T.38 option disabled); however, T.38 is the preferred method.
- The **Re-invite Supported** parameter can be checked to allow for codec re-negotiation in cases where the target of an incoming call or transfer does not support the codec originally negotiated on the trunk.
- The **DTMF Support** parameter can remain set to the default value **RFC2833/RFC4733**.
- Click **OK** (not shown).

The screenshot shows the configuration interface for a SIP Line on the VoIP tab. The 'Codec Selection' dropdown is set to 'Custom'. Below it, there are two lists: 'Unused' and 'Selected'. The 'Unused' list contains 'G.711 ALAW 64K' and 'G.722 64K'. The 'Selected' list contains 'G.729(a) 8K CS-ACELP' and 'G.711 ULAW 64K'. There are navigation buttons between the lists: '>>>', '↑', '<<<', '↓', and '>>>'. Below the lists, the 'Fax Transport Support' dropdown is set to 'T38', 'DTMF Support' is 'RFC2833/RFC4733', and 'Media Security' is 'Disabled'. On the right side, there are several checkboxes: 'Local Hold Music' (unchecked), 'Re-invite Supported' (checked), 'Codec Lockdown' (unchecked), 'Allow Direct Media Path' (unchecked), 'Force direct media with phones' (unchecked), and 'PRACK/100rel Supported' (unchecked).

5.4.7. SIP Line - T38 Fax Tab

Note – This tab is only available when configuring a SIP line on IP Office 500 V2, and the settings on this tab are only accessible if **Re-invite Supported** and a **Fax Transport Support** option (**T38**) are selected on the **VoIP** tab (**Section 5.4.6**). See **Section 6.4** for T.38 fax settings.

5.4.8. SIP Line – SIP Advanced Tab

IP Office can be configured to signal when a call is placed on hold by sending an INVITE with media attribute “sendonly”. AT&T in turn will respond with media attribute “recvonly”, and will stop sending RTP media for the duration the call is on hold. When the call is taken off of hold, IP Office will send another INVITE with media attribute “sendrecv” indicating to AT&T to start sending RTP again.

To have Avaya IP Office signal to AT&T when a call is placed on/off hold, select the **SIP Line** → **SIP Advanced** tab and enter the following:

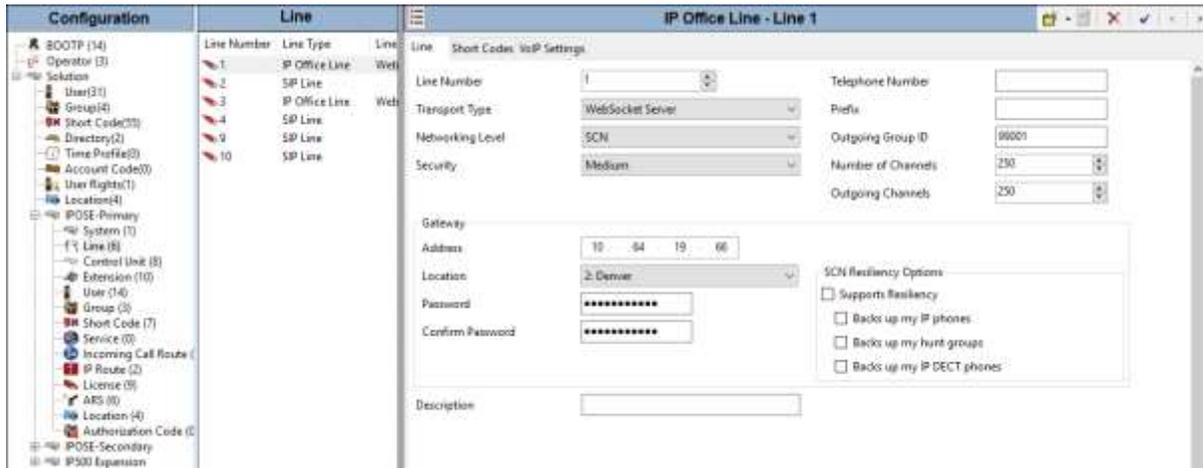
- Select **Indicate HOLD**.

The screenshot shows the 'SIP Line: Transport: SIP URI: VoIP' configuration window, specifically the 'SIP Advanced' tab. The window is divided into several sections:

- Addressing:** Association Method is set to 'By Source IP address', Call Routing Method is 'Request URI', and Suppress DNS SRV Lookups is unchecked.
- Identity:** A list of checkboxes for various identity-related settings. 'Cache Auth Credentials' is checked, while others are unchecked.
- Media:** A section containing several checkboxes and dropdown menus. The 'Indicate HOLD' checkbox is checked and highlighted with a red box. Other settings include 'Allow Empty INVITE', 'Send Empty re-INVITE', 'Allow To Tag Change', 'P-Early-Media Support' (set to 'None'), 'Send SilenceSupp=Off', 'Force Early Direct Media', and 'Media Connection Preservation' (set to 'Disabled').
- Call Control:** A section with numerical input fields for 'Call Initiation Timeout (s)' (4) and 'Call Queuing Timeout (mins)' (5). It also includes dropdown menus for 'Service Busy Response' (503 - Service Unavailable), 'on No User Responding Send' (408-Request Timeout), and 'Action on CAC Location Limit' (Allow Voicemail). There are also checkboxes for 'Suppress Q.850 Reason Header', 'Emulate NOTIFY for REFER', and 'No REFER if using Diversion'.

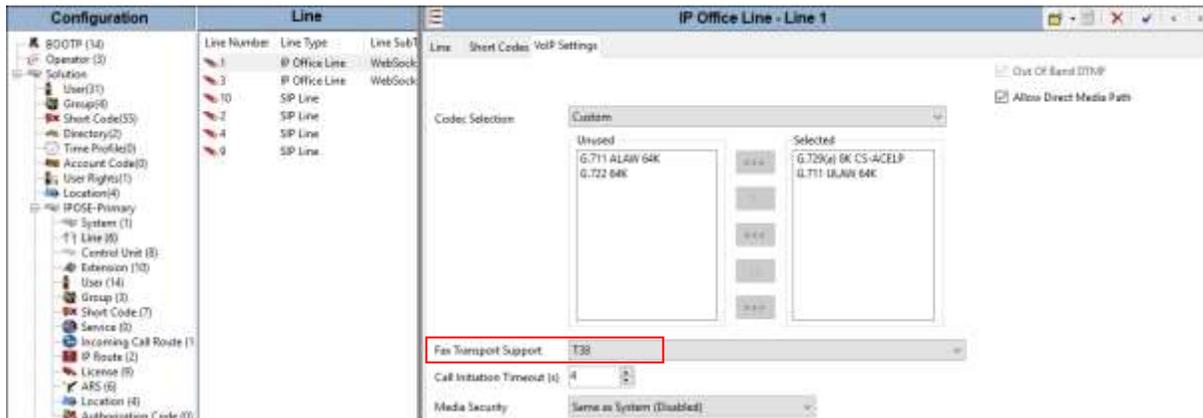
5.5. IP Office Line

IP Office Lines are automatically created on each server when a Secondary server or Expansion System is added to the solution. To edit an existing IP Office Line, select **Line** in the Navigation pane, and select the appropriate Line to be configured in the Group pane. Below is the IP Office Line to the Expansion System.



In the reference configuration, a fax machine is connected to one of the analog ports on the Expansion System. To accommodate T.38 fax, select the **VoIP Settings** tab and configure the following:

- **Fax Transport Support: T38**



5.6. Users, Extensions, and Hunt Groups

In this section, examples of Avaya IP Office Users, Extensions, and Hunt Groups are illustrated. Note that the following examples do not discuss all available options, and the screen shots may not display all available parameters. Parameters/options not discussed, should assume to be default.

5.6.1. Analog User 6300

The following screen shows the **User** tab for analog phone User **6300**. This user corresponds to a fax machine.

1. To add a User, right click on **User** in the Navigation pane, and select **New** (not shown). To edit an existing User, select **User** in the Navigation pane, and select the appropriate user to be configured.

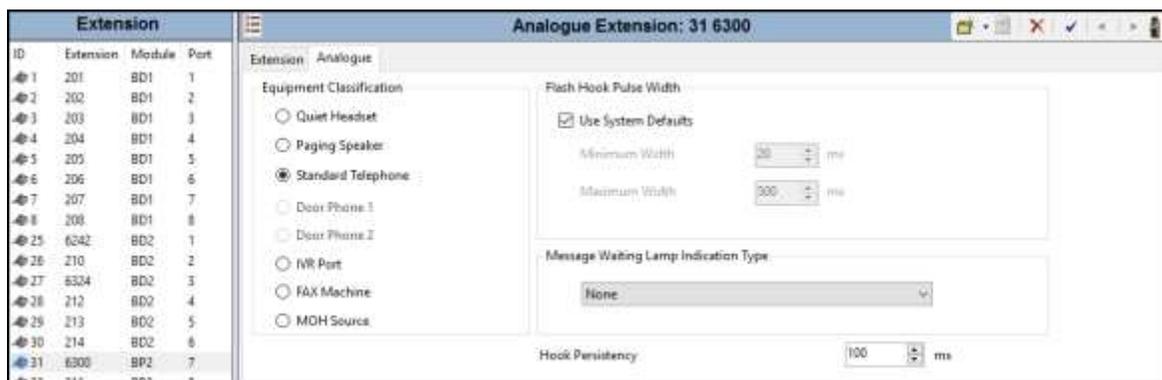
The screenshot shows the Avaya IP Office configuration interface. On the left is a navigation pane titled 'User' with a list of users. The user 'Fax' with extension 6300 is selected. The main window is titled 'IP500 Expansion : Fax: 6300' and contains a configuration form for this user. The form includes fields for Name, Password, Confirm Password, Unique Identity, Conference PIN, Confirm Audio: Conference PIN, Account Status (set to 'Enabled'), Full Name, Extension (6300), Email Address, Locale, Priority (5), System Phone Rights (None), and Profile (Basic User). There are also checkboxes for 'Receptionist', 'Enable Softphone', and 'Enable one-X Portal Services'. The 'OK', 'Cancel', and 'Help' buttons are at the bottom right.

IP Office	Name	Extension
IP500 Expansion	Extn6302	6302
IP500 Expansion	ATT-Avaya 9508	6324
IPOSE-Primary	ATT-softphone	6323
IPOSE-Primary	ATT-Avaya 9611	6322
IPOSE-Primary	ATT-Avaya 1140E	6321
IPOSE-Primary	ATT-Avaya 1616	6320
IP500 Expansion	Fax	6300
IP500 Expansion	Avaya 9508	6242
IPOSE-Primary	Avaya Com	6239
IPOSE-Primary	Avaya 9611	6237
IPOSE-Primary	Avaya 1140E	6235
IPOSE-Primary	Avaya 1616	6233
IPOSE-Primary	Extn301	301
IP500 Expansion	Extn216	216
IP500 Expansion	Extn214	214
IP500 Expansion	Extn213	213
IP500 Expansion	Extn212	212
IP500 Expansion	Extn211	211
IP500 Expansion	Extn210	210
IP500 Expansion	Extn208	208
IP500 Expansion	Extn207	207
IP500 Expansion	Extn206	206
IP500 Expansion	Extn205	205
IP500 Expansion	Extn204	204
IP500 Expansion	Extn203	203
IP500 Expansion	Extn202	202
IP500 Expansion	Extn201	201

2. Analog (or digital) phone extension ports are either integral to the control unit or added by the installation of an analog or digital phone expansion module. Analog (or digital) extension records are automatically created for each physical extension port within the system. These ports cannot be added or deleted manually. For Server Edition, non-IP extensions are only supported on Expansion System (V2) units.
 - To edit an existing analog extension, select the appropriate extension to be configured (e.g., **6300**).



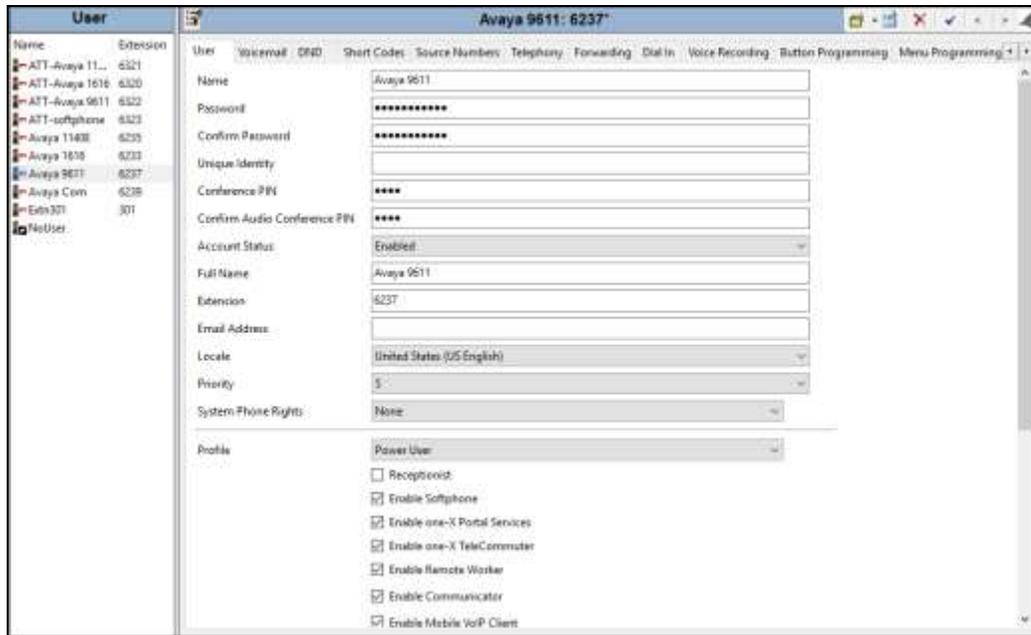
- Select the Analogue tab and verify that **Standard Telephone** is selected. Note that even though a fax machine is connected, it needs to be classified as a standard telephone.
- Click the **OK** button (not shown).



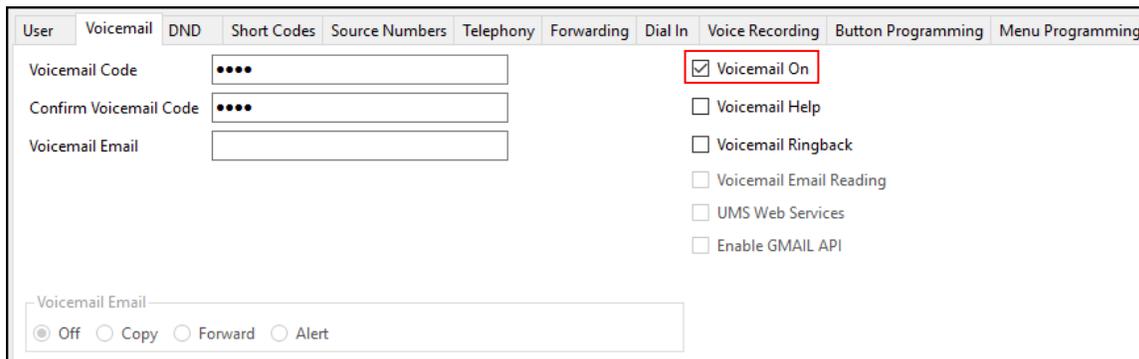
5.6.2. IP Phone User 6237

1. Following the steps shown in **Section 5.5.1**, create a 9611 H.323 IP phone user (e.g., **6237**).

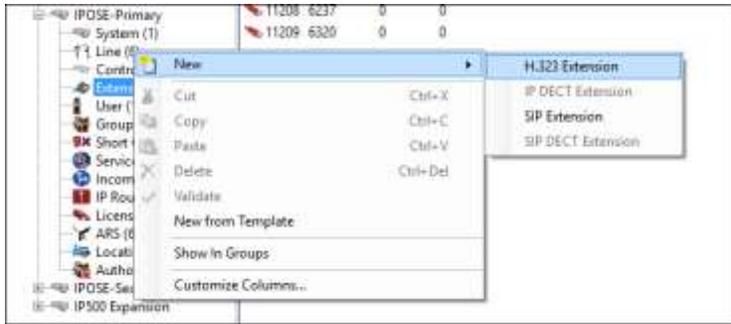
- **Password:** This password is used by user applications such as SoftConsole, one-X® Portal and TAPI, or users with Dial In access. Note that this is *not* the user's phone login code (see the information on the **Extension** tab below), or their Voicemail mailbox password (see information on the **Voicemail** tab below).
- **Conference PIN:** This is the pin number used to access the user's meet me conference.
- The **Profile** parameter is set to **Power User**. This gives this user access to additional Avaya P Office features. See [3] for more information.



The following screen shows the **Voicemail** tab for user 6237. The **Voicemail On** box is checked and a Voicemail password can be configured using the **Voicemail Code** and **Confirm Voicemail Code** parameters.



2. To create an associated extension, right click on **Extension** in the Navigation Pane, and select **New → H323 Extension**.

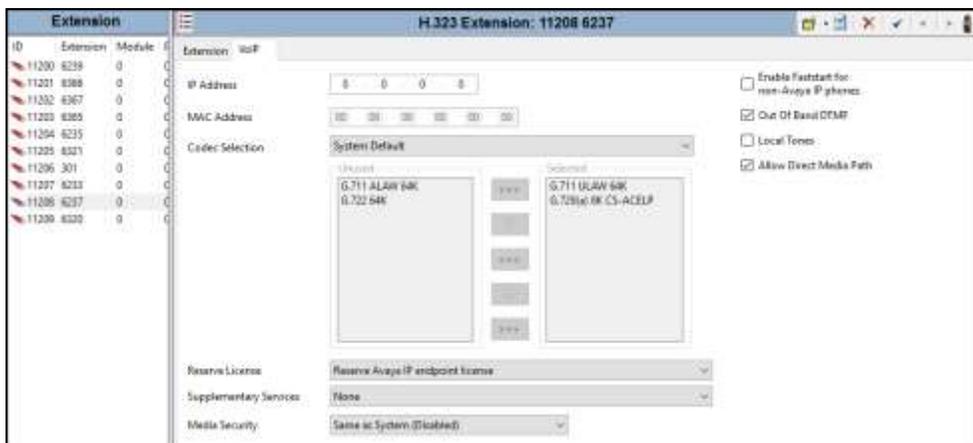


On the **Extension** tab, enter the **Base Extension** (e.g., **6237**). Note that the **Extension ID** field will auto populate. The **Phone Password** will be used by the telephone user as the phone login password.



Select the **VoIP** tab and provision the following:

- Keep the **IP Address** field as the default value (**0.0.0.0**).
- Populate the **Selected** column with **G.711 ULAW 64K** as the first codec and **G.729(a) 8K CS-ACELP** as the second codec, (see **Section 5.3.4**).
- Click the **OK** button (not shown).



5.6.3. Hunt Groups

Users may also receive incoming calls as members of a hunt group. To configure a new hunt group, right-click **HuntGroup** from the Navigation pane and select **New**. To view or edit an existing hunt group, select **HuntGroup** from the Navigation pane, and the appropriate hunt group from the Group pane.

1. The following screen shows the **Hunt Group** tab for hunt group **Call Center**. This hunt group was configured to contain various IP Office extensions. In the reference configuration, these telephones extensions are rung based on idle time, due to the **Ring Mode** setting **Longest Waiting**. Click the **Edit** button to select/deselect from the **User List** included in the Hunt Group from the list of available users.

The screenshot shows the configuration window for a Hunt Group named 'Call Center'. The 'Name' field is 'Call Center' and the 'Profile' is 'Standard Hunt Group'. The 'Extension' is '401'. The 'Ring Mode' is set to 'Longest Waiting'. The 'No Answer Time (sec)' is 'System Default (13)'. The 'Central System' is 'IPOSE-Primary'. Below these settings is a 'User List' table with the following data:

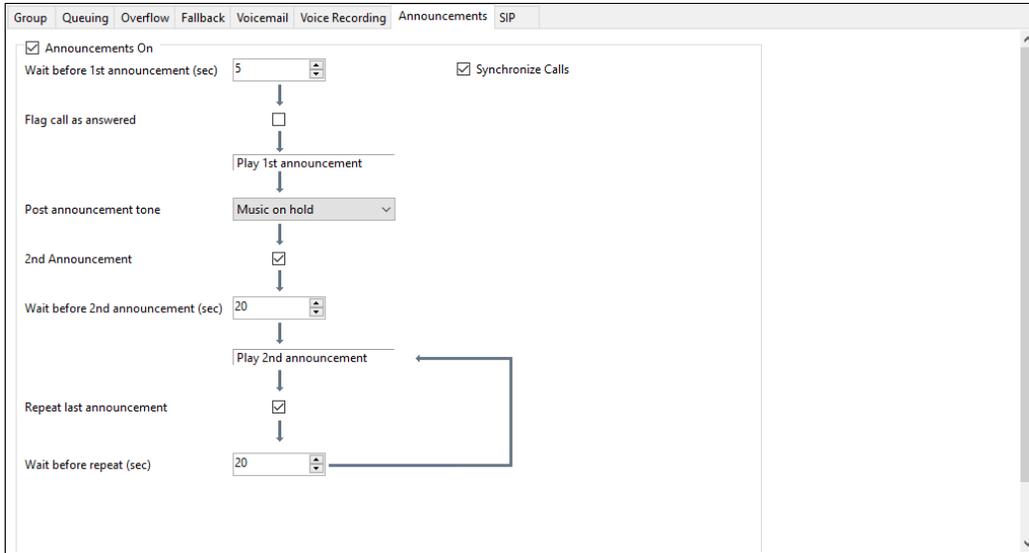
Extension	Name	System
<input checked="" type="checkbox"/>	6242	Avaya 9508 IP500 Expansion
<input checked="" type="checkbox"/>	6237	Avaya 964T IPOSE-Primary
<input checked="" type="checkbox"/>	6233	Avaya 1616 IPOSE-Primary
<input checked="" type="checkbox"/>	6235	Avaya 1140E IPOSE-Primary
<input checked="" type="checkbox"/>	6239	Avaya Com IPOSE-Primary

Buttons for 'Edit...' and 'Remove' are visible at the bottom right of the window.

2. Under the **Queuing** tab, check the **Queuing On** box and set the **Queue Length** field to any desirable value. Use the default values for all the other fields.

The screenshot shows the 'Queuing' configuration tab. The 'Queuing On' checkbox is checked. The 'Queue Length' is set to '5'. The 'Normalize Queue Length' checkbox is checked. The 'Queue Type' is set to 'Assign Call On Agent Answer'. The 'Calls In Queue Alarm' section has 'Calls In Queue Threshold' set to '1' and 'Analog Extension to Notify' set to '<None>'. The 'Group' pane at the top shows 'Queuing' selected.

- Under the **Announcements** tab, check the **Announcements On** box. The wait time can be set to any desirable value. The **Synchronize Calls** box is checked to greatly reduce the number voicemail channels needed to play announcements. These announcements are played if an agent for a particular skill is unavailable.



- Click on **OK** (not shown).

In the reference configuration, these steps were used to create additional Hunt Group “Support” (402).

5.7. Incoming Call Routes

Note – The digits defined and matched in the Incoming Call Route table, are the DNIS digits specified in the AT&T Request-URI, not the DID digits dialed by the caller.

The Incoming Call Route table will map specific AT&T DNIS numbers to an IP Office User, or Hunt Group, as well as to Voicemail Pro scripts.

To add an incoming call route, right click on **Incoming Call Route** in the Navigation pane, and select **New** (not shown). To edit an existing incoming call route, select an **Incoming Call Route** in the Navigation pane, and the associated call route information is displayed in the Group pane.

5.7.1. Calls to IP Office Stations and Hunt Groups

In the example below, the incoming number **000008885551025** is directed to H.323 phone 6237.

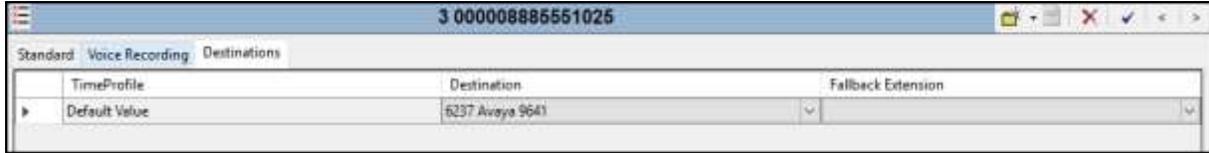
1. On the **Standard** tab enter the following:

- **Line Group ID:** Enter the SIP Line defined in **Section 5.4** (e.g., **3**).
- **Incoming Number:** Enter the associated DNIS digits sent by AT&T (e.g., **000008885551025**).
- Use default values for the remaining fields and click **OK** (not shown).

The screenshot shows the configuration page for an Incoming Call Route. The left navigation pane lists various system components, with 'Incoming Call Route' selected. The main area displays the configuration for the route with the incoming number '3 000008885551025'. The configuration is organized into tabs: 'Standard', 'Voice Recording', and 'Destinations'. The 'Standard' tab is active, showing the following fields:

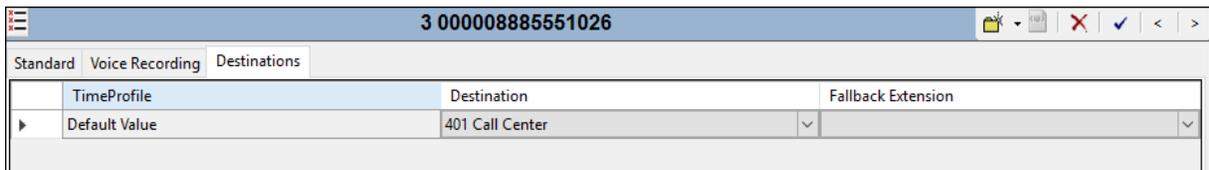
Field	Value
Bearer Capability	Any Voice
Line Group ID	3
Incoming Number	000008885551025
Incoming Sub Address	
Incoming CLI	
Locale	
Priority	1 - Low
Tag	
Hold Music Source	System Source
Ring Tone Override	None

2. On the **Destinations** tab enter the following:
 - In the **Destinations** column, select extension **6237** from the drop down menu.
 - Use default values for the remaining fields and click **OK** (not shown).



Below is an example of a call for **000008885551026** being directed to Hunt Group **401** (Call Center).

Standard	Voice Recording	Destinations
Bearer Capability	Any Voice	
Line Group ID	3	
Incoming Number	000008885551026	
Incoming Sub Address		
Incoming CLI		
Locale		
Priority	1 - Low	
Tag		
Hold Music Source	System Source	
Ring Tone Override	None	

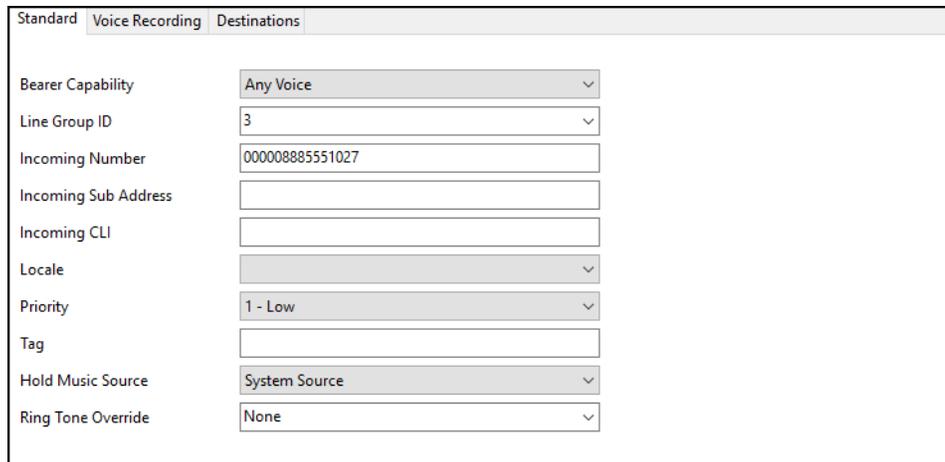


5.7.2. Calls to Voicemail Pro Scripts

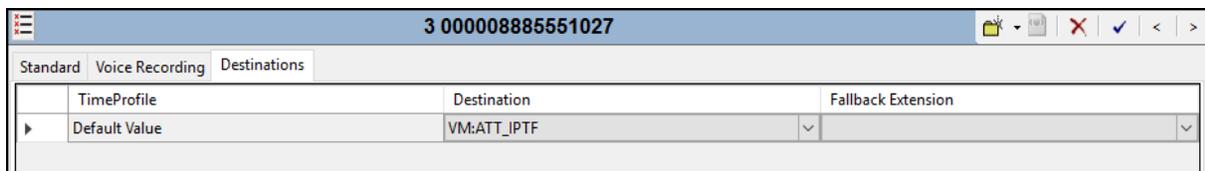
As described in **Sections 5.6.1** and **5.8**, Voicemail Pro scripts are defined with specific names. These script names are specified as destinations in the Incoming Call Route table.

In the example below, incoming number **000008885551027** is directed to the Voicemail Pro Auto-Attendant script **ATT_IPTF**.

1. On the **Standard** tab repeat the steps in **Section 5.7.1**, with the following changes:
 - **Incoming Number:** Enter the associated DNIS digits sent by AT&T (e.g., **000008885551027**).
2. On the **Destinations** tab enter the following:
 - In the **Destinations** column, enter the string **VM:ATT_IPTF** from the drop down menu (note if the voicemail module does not appear in the list, enter the value manually).
 - Use default values for the remaining fields and click **OK** (not shown).



Bearer Capability	Any Voice
Line Group ID	3
Incoming Number	000008885551027
Incoming Sub Address	
Incoming CLI	
Locale	
Priority	1 - Low
Tag	
Hold Music Source	System Source
Ring Tone Override	None



TimeProfile	Destination	Fallback Extension
Default Value	VM:ATT_IPTF	Default Value

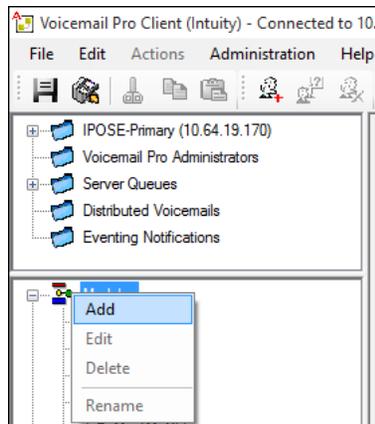
5.8. Call Center Provisioning in Voicemail Pro

Note – While Voicemail Pro provisioning and programming is beyond the scope of this document, a sample Auto-Attendant script is described below.

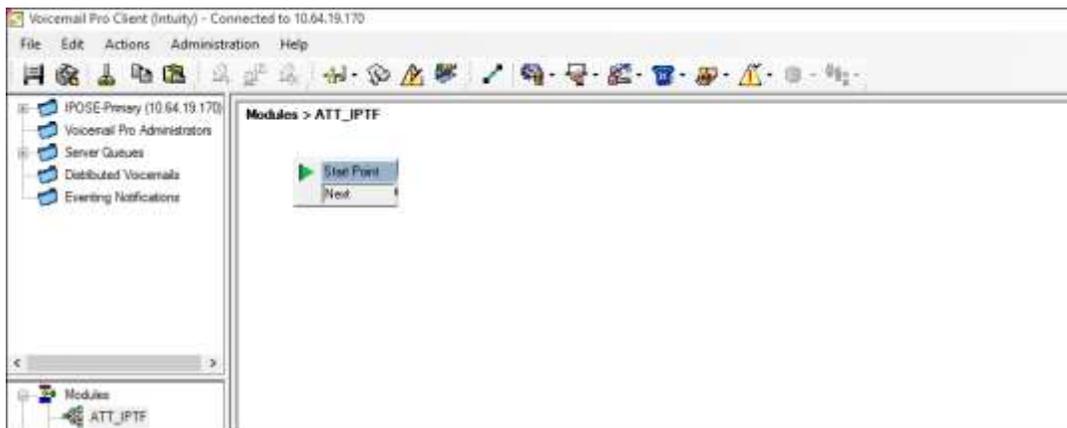
In the reference configuration, Voicemail Pro is used for Voicemail processing as well as for simulating basic Call Center functionality.

The Auto-Attendant function was provisioned to prompt callers to select a numeric option (1, 2, or 3), that would forward the call to an associated Avaya IP Office Hunt Group (Call Center, and Support), or user 6237. This is accomplished via the following steps:

1. Hunt Groups **Call Center** and **Support** are created in IP Office (**Section 5.6.3**).
2. User 6237 is created in IP Office (**Section 5.6.2**).
3. Incoming Call Route for DNIS digits **000008885551027** is defined for access to the Auto-Attendant script (**Section 5.7.2**).
4. Via the Voicemail Pro GUI interface:
 - Open the **Voicemail Pro Client** application and log in to the Voicemail Pro server (not shown).
 - Create a **Start Point** by right clicking on **Modules** and selecting **Add**.

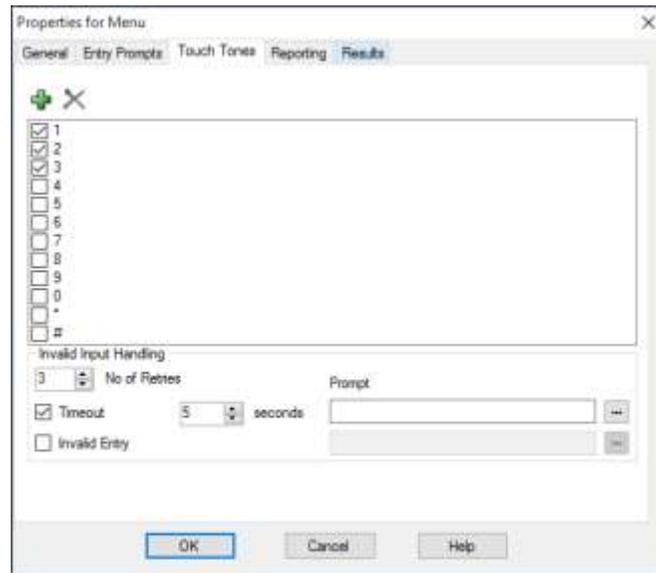


- Enter a name (e.g., **ATT_IPTF**) and click on **OK** (not shown). The new script "ATT_IPTF" will appear under Modules and a Start Point icon will appear in the work area.



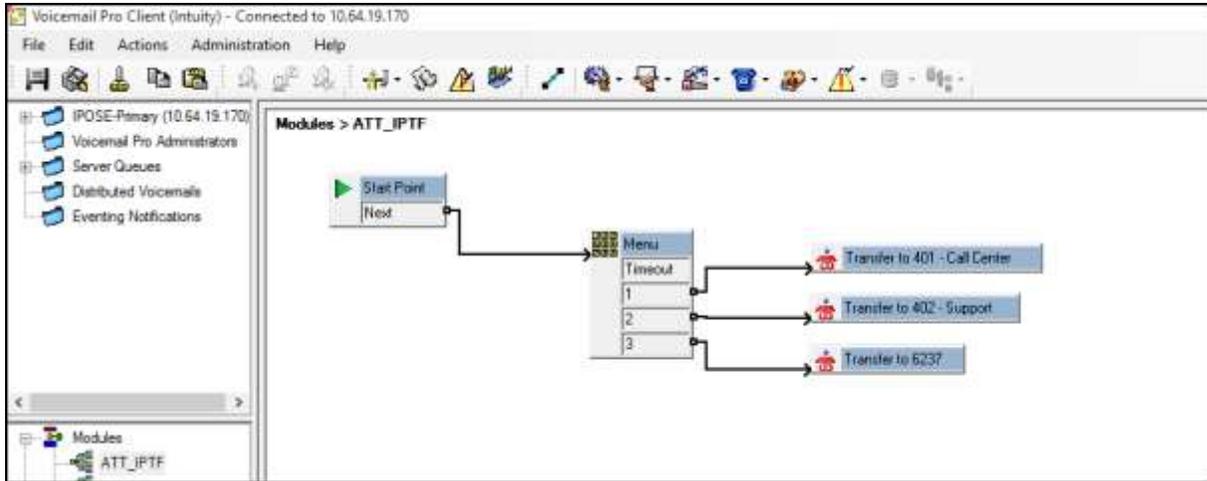
- Click on the **Start Point** icon  to activate the script options at the top of the screen. From the options, select the **Basic Actions** icon , select the **Menu** icon , and click on the work area to place the **Menu** icon.
 - i. Double click the **Start Point** icon.
 1. On the **General** tab → **Token Name**, enter **Start Point** and click **OK** (not shown).

- ii. Double click the **Menu** icon.
 1. On the **General** tab → **Token Name**, enter **Menu** (not shown).
 2. On the **Entry Prompts** tab (not shown), select or create an **Entry Prompt** that will tell the caller what digits to press (e.g., **mainmenu.wav**). To modify an existing recording, double click on the .wav file and rerecord. If no .wav files exist, double click on the  icon to open the .wav editor.
 3. On the **Touch Tone** tab:
 - a. Select **1, 2,** and **3** as the possible entry digits.
 - b. Select **3** for **No of Retries**.
 4. Click on **OK**.



- Click on the Telephony Actions icon , select the Transfer icon , and click on the work area to place the **Transfer** icon in the work area. This will be used for “Sales”. Select and place two more Transfer Icons (these will be used for “Service” and “Parts”).
 - i. Double click on the first **Transfer** icon (“**Call Center**”)
 1. On the **General** tab → **Token Name** = **Transfer to 401 - Call Center** (not shown).
 2. On the **Specific** tab → **Destination** = **401** (not shown).
 - ii. Double click on the second **Transfer** icon (“**Support**”).
 1. On the **General** tab → **Token Name** = **Transfer to 402 - Support** (not shown).
 2. On the **Specific** tab → **Destination** = **402** (not shown).
 - iii. Double Click on the third **Transfer** icon (“**Ext6237**”).
 1. On the **General** tab, **Token Name** = **Transfer to 6237** (not shown).
 2. On the **Specific** tab, **Destination** = **6237** (not shown).
- From the options bar, select the Connector icon  and:

- i. Drag a connecting flow line from the **Start Point** box to the **Menu** box (see screen shot below).
- ii. Drag connecting flow lines from each of the **Menu** options to their associated **Transfer** boxes (see screenshot below).



5. From the top menu select **File** → **Save & Make Live**, or select the  icon.

When the associated AT&T DNIS number is received (e.g., **00008885551027**), IP Office will send the call to Voicemail Pro. The caller will be prompted to enter 1, 2, or 3 to access Call Center, Support, or user 6237. The associated Avaya IP Office extension (e.g., 401, 402, or 6237) will then ring.

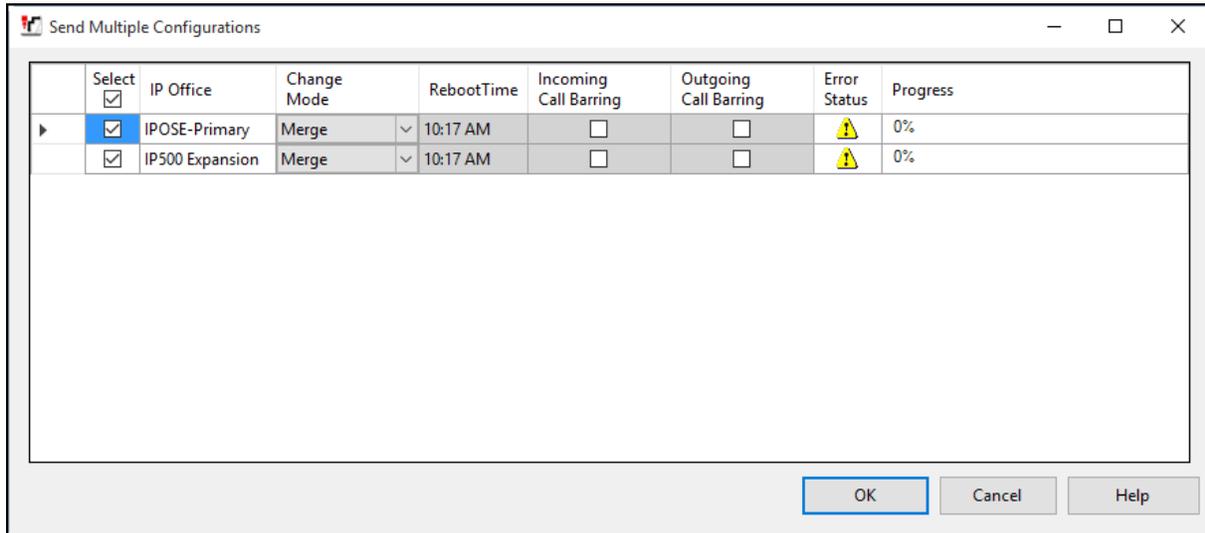
5.9. Saving Configuration Changes to Avaya IP Office

The provisioning changes made in Avaya IP Office Manager must be applied to the Avaya IP Office server in order for the changes to take effect. As noted in the previous sections, any changes made to an IP Office provisioning tab must be accepted by clicking **OK** on the associated screen. However these changes will not take effect until they are written to the IP Office configuration.

At the top of the Avaya IP Office Manager GUI, click **File → Save Configuration** (note that if that option is grayed out, no changes are pending).

A screen similar to the one below will appear, with either **Merge** or **Immediate** automatically selected, based on the nature of the configuration changes. The **Merge** option will save the configuration change with no impact to the current system operation. The **Immediate** option will save the configuration and cause the Avaya IP Office server to reboot.

Click **OK** to execute the save.



The active configuration may be saved to a file at any time by selecting **File → Save Configuration As**.

6. Avaya IP Office Expansion Configuration

Navigate to **File** → **Open Configuration** (not shown), select the proper Avaya IP Office system from the pop-up window, and log in using the appropriate credentials. Clicking the “plus” sign next to **IP500 Expansion** on the left navigation pane will expand the menu on this server.



6.1. Physical Hardware

In the sample configuration, looking at the Expansion System IP500 V2 from left to right, the first module is a **TCM 8** Digital Station Module. This module supports BCM / Norstar T-Series and M-Series telephones. The second module is a **COMBO6210/ATM4** module. This module is used to add a combination of ports to an IP500 V2 control unit and is not supported by IP500 control units. The module supports 10 voice compression channels. Codec support is G.711, G729A and G.723 with 64ms echo cancellation. G.722 is supported by IP Office Release 8.0 and higher. The “Combo” card will support 6 Digital Station ports for digital stations in slots 1-6 (except 3800, 4100, 4400, 7400, M and T-Series), 2 Analog Extension ports in slots 7-8, and 4 Analog Trunk ports in slots 9-12.



6.2. System Settings

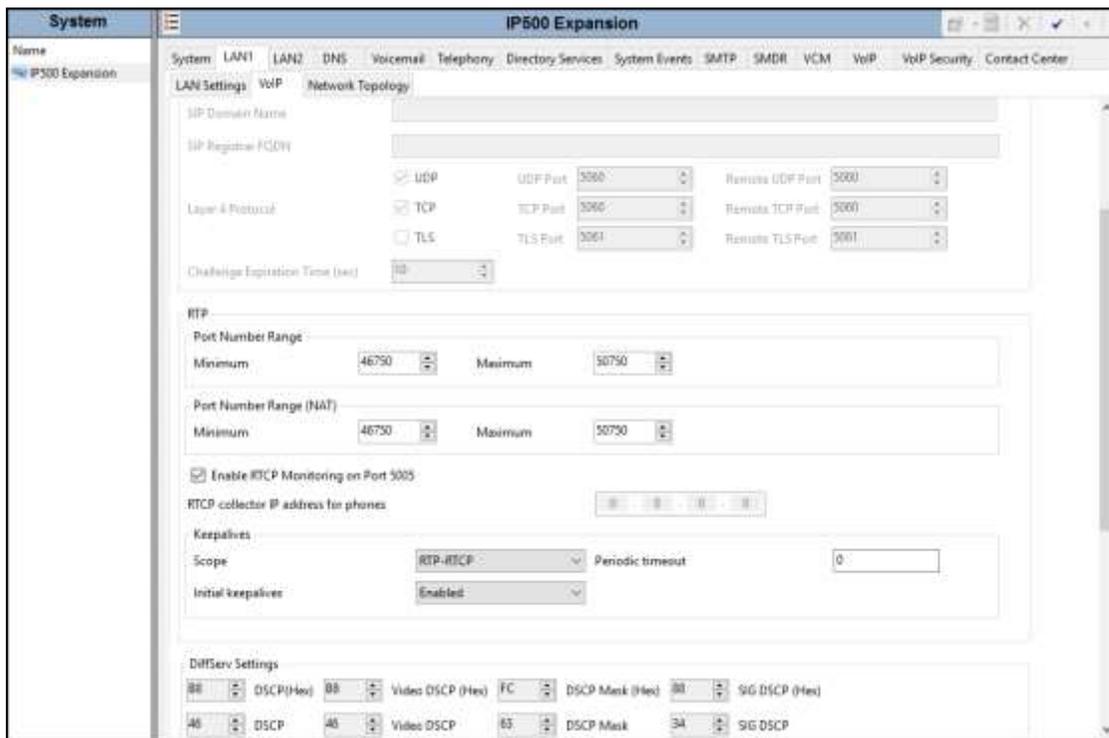
This section illustrates the configuration of system settings. Select **System** in the Navigation pane to configure these settings. The subsection order corresponds to a left to right navigation of the tabs in the Details pane for System settings. For all of the following configuration sections, the **OK** button (not shown) must be selected in order for any changes to be saved.

6.2.1. LAN Settings

In the sample configuration, LAN1 is used to connect the Expansion System to the enterprise network. To view or configure the **IP Address** of LAN1, select the **LAN1** tab followed by the **LAN Settings** tab. As shown in **Figure 1**, the IP Address of the Expansion System is **10.64.19.66**. Other parameters on this screen may be set according to customer requirements.



Select the **VoIP** tab as shown in the following screen. If desired, the **RTP Port Number Range** can be customized to a specific range of receive ports for the RTP media paths from Primary server to the Expansion System. The defaults are used here.



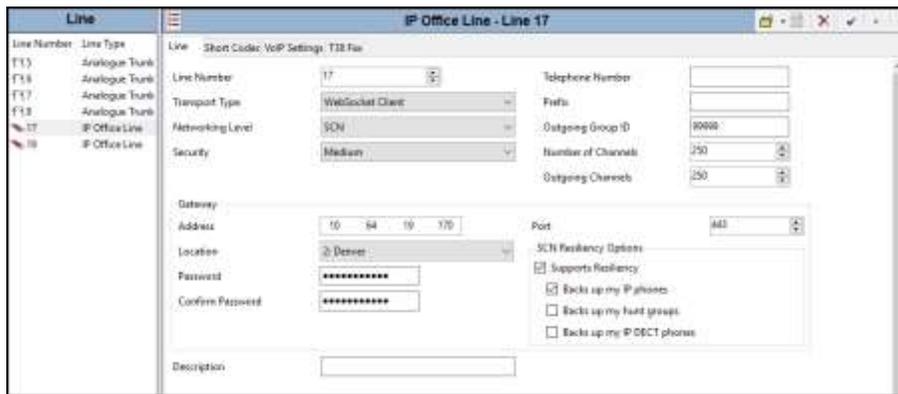
6.3. IP Route

In the sample configuration, the Expansion System LAN1 port is physically connected to the local area network switch at the IP Office customer site. The default gateway for this network is **10.64.19.1**.



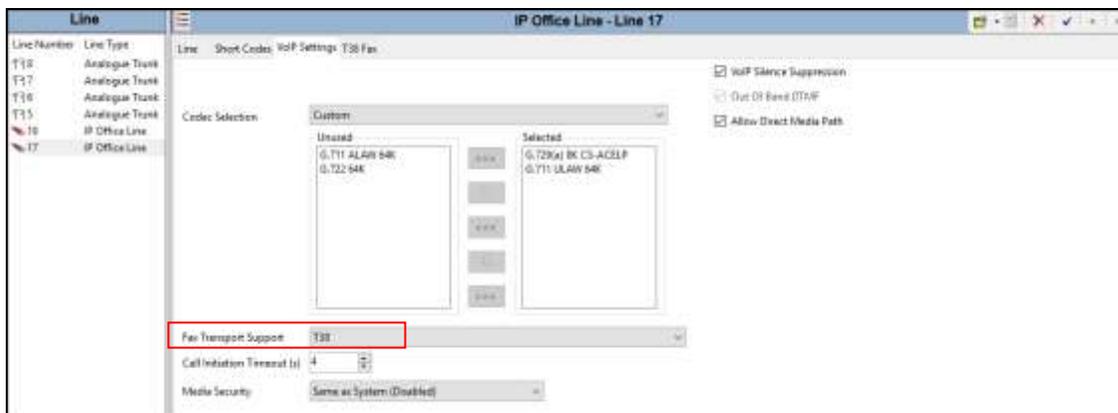
6.4. IP Office Line

The IP Office Lines are automatically created on each server when a Secondary server or Expansion System is added to the solution. Below is the IP Office Line to the Primary server.



In the reference configuration, a fax machine is connected to one of the analog ports on the Expansion System. To accommodate T.38 fax, select the **VoIP Settings** tab and configure the following:

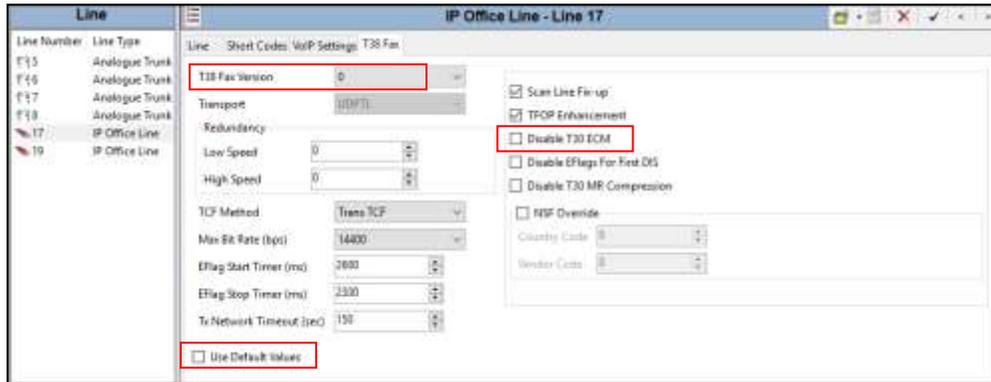
- **Fax Transport Support: T38**



Select the **T38 Fax** tab and enter the following:

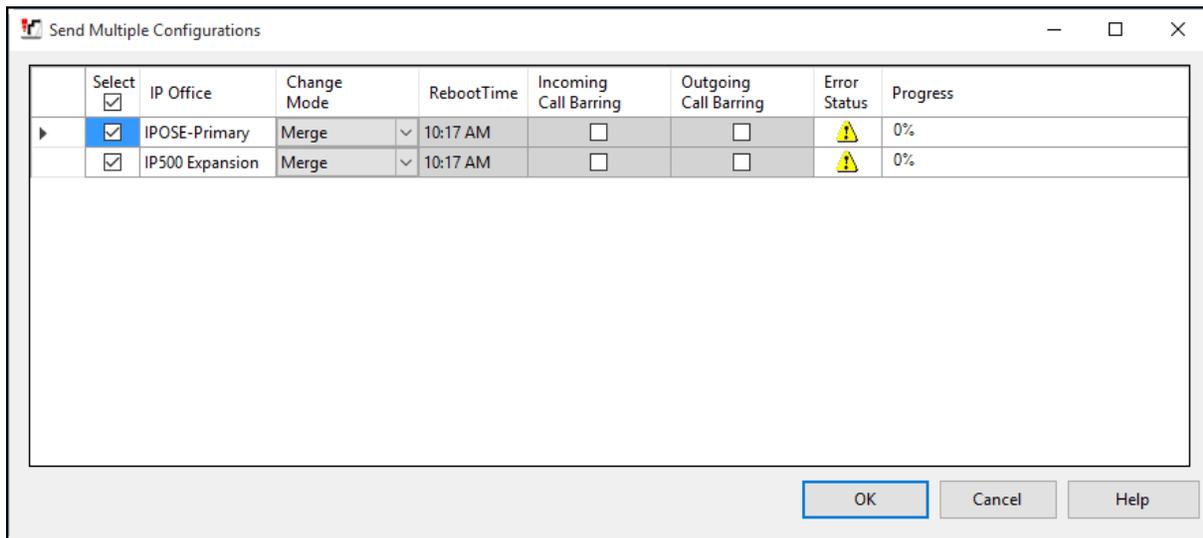
- Unselect the **Use Default Values** option.
- Set the **T38 Fax Version** option to **0** (zero). This matches the version AT&T uses.
- Verify that **Disable T30 ECM** is *not* checked,

Default values are used for the remaining fields. Select **Ok** (not shown).



6.5. Saving Configuration Changes to Avaya IP Office

Similar to **Section 5.9**, navigate to **File → Save Configuration** in the menu bar at the top of the screen to save the configuration performed in the preceding sections.



The active configuration may be saved to a file at any time by selecting **File → Save Configuration As**.

7. AT&T IP Toll Free Service Configuration

AT&T provides the IPTF service border element IP address, the access DID numbers, and the associated DNIS digits used in the reference configuration. In addition the AT&T IPTF features, and their associated access numbers, are also assigned by AT&T. AT&T requires that the Avaya IP Office public (LAN2) IP address be provided to the IPTF service, as part of the provisioning process.

8. Verification Steps

The following procedures may be used to verify the Avaya IP Office R10 with the AT&T IP Toll Free service configuration.

8.1. AT&T IP Toll Free Service

The following scenarios may be executed to verify Avaya IP Office R10 functionality with the AT&T IPTF service:

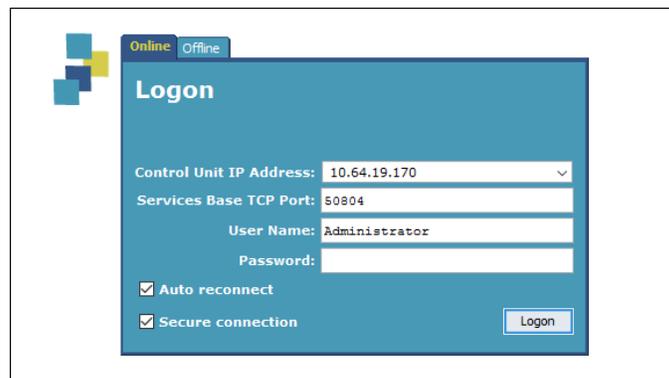
- Place inbound calls, answer the calls, and verify that two-way talk path exists. Verify that the calls remain stable for several minutes and disconnects properly.
- Incoming calls using the G.729A and G.711 ULAW codecs.
- Verify basic call functions such as hold, transfer, and conference.
- Place an inbound call to a telephone, but do not answer the call. Verify that the call covers to voicemail (e.g., Voicemail Pro). Retrieve the message either locally or from PSTN.
- Using the appropriate IPTF access numbers and codes, verify the “Legacy Transfer Connect” DTMF initiated features.
- Inbound fax using T.38 or G.711.
- SIP OPTIONS monitoring of the health of the SIP trunk.

8.2. Avaya IP Office 10

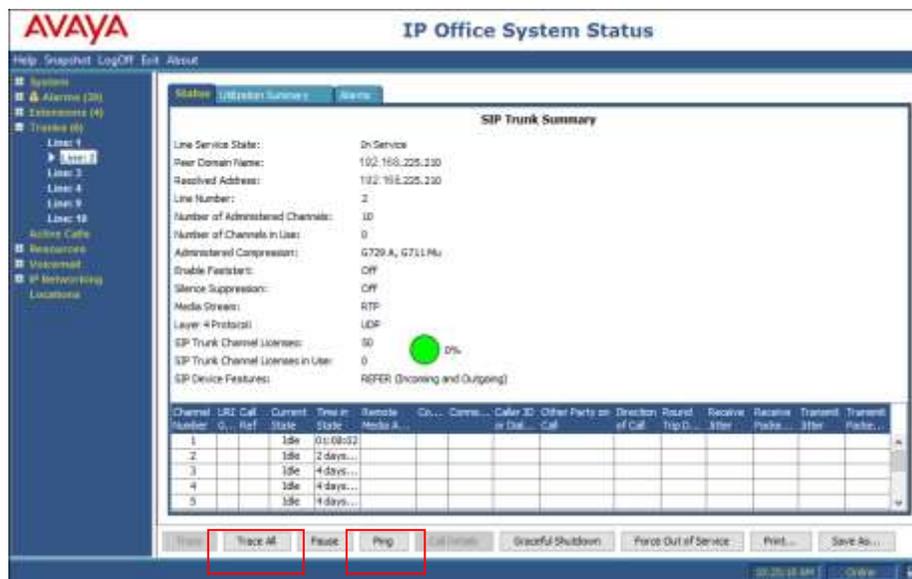
The following items may be used to analyze/troubleshoot Avaya IP Office operations.

8.2.1. System Status Application

The System Status application can be used to monitor or troubleshoot Avaya IP Office. The System Status application can typically be accessed from **Start → Programs → Avaya IP Office → System Status**. The following screen shows an example **Logon** screen. Enter the Avaya IP Office IP address in the **Control Unit IP Address** field, and enter an appropriate **User Name** and **Password**. Click **Logon**.

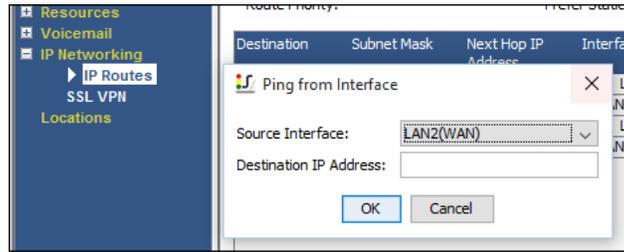


After logging in, select **Trunks → Line: 2** from the left navigation menu. (SIP Line 2 is configured in **Section 5.4**). A screen such as the one shown below is displayed. In the lower left, the **Trace All** button may be pressed to display tracing information as calls are made using this SIP Line. The **Ping** button can be used to ping the other end of the SIP trunk (e.g., the AT&T Border Element, however the AT&T Border Element may not be configured to respond to pings).



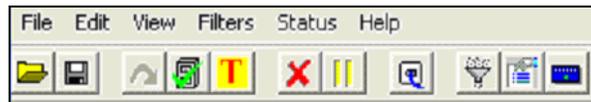
Channel Number	DRZ	Call Ref	Current State	Orig. In State	Service Media	Call ID	Caller ID	Other Party	Director of Call	Round Trip D.	Receive Other	Receive Other	Transmit Other	Transmit Other
1			Idle	01:08:02										
2			Idle	2 days...										
3			Idle	4 days...										
4			Idle	4 days...										
5			Idle	4 days...										

By navigating to **IP Networking** → **IP Routes**, and clicking on **Ping**, an IP Office **Source Interface**, and any **Destination IP Address**, may be specified for a ping by clicking **OK**.



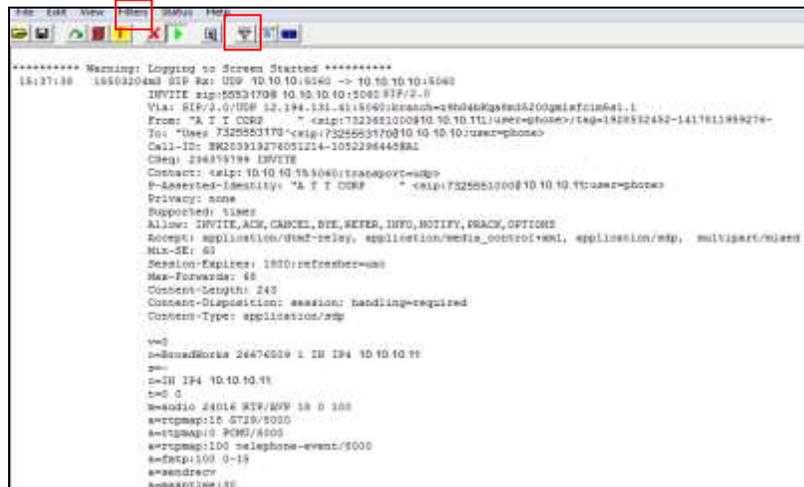
8.2.2. System Monitor Application

The System Monitor application can also be used to monitor or troubleshoot Avaya IP Office functionality (see reference [3]). The System Monitor application can typically be accessed from **Start** → **Programs** → **Avaya IP Office** → **Monitor**.



The Monitor will be active at startup. To pause the Monitor, press the Pause  button.

The pause button will be replaced with the Start  button. Press this button to resume the monitoring. To clear the Monitor display, press the Clear  button. Below is a sample of a monitored inbound call.



The displayed data may be customized. Select the **Options** button , or select **Filters** → **Trace Options**. The following screen shows the **SIP** tab, allowing configuration of SIP monitoring. In this example, only the **SIP Rx** and **SIP Tx** boxes are selected.



9. Conclusion

As illustrated in these Application Notes, Avaya IP Office R10 can be configured to interoperate successfully with the AT&T IP Toll Free service using **AVPN** or **MIS/PNT** transport connections, utilizing service features listed in **Section 2.1**, and within the limitations described in **Section 2.2**.

The reference configuration shown in these Application Notes is representative of a basic enterprise customer configuration and is intended to provide configuration guidance to supplement other Avaya product documentation. It is based upon formal interoperability compliance testing as part of the Avaya DevConnect Service Provider program.

10. References

Avaya:

Avaya product documentation is available at <http://support.avaya.com>

- [1] *IP Office™ Platform 10.0, Deploying Avaya IP Office™ Platform Servers as Virtual Machines*, Document Number 15-601011, Issue 04d, July 2016
- [2] *IP Office™ Platform 10.0, Deploying Avaya IP Office™ Platform IP500 V2*, Document Number 15-601042, Issue 31h, Aug 2016
- [3] *Administering Avaya IP Office™ Platform with Manager*, Release 10.0, August 2016
- [4] Additional Avaya IP Office information can be found at:
<http://marketingtools.avaya.com/knowledgebase/>

AT&T IPTF Service:

- [5] AT&T IP Toll Free Service description -
<http://www.business.att.com/enterprise/Service/voice-services/contact-center-solutions/ip-toll-free/>

©2016 Avaya Inc. All Rights Reserved.

Avaya and the Avaya Logo are trademarks of Avaya Inc. All trademarks identified by TM and [®] are registered trademarks or trademarks, respectively, of Avaya Inc. All other trademarks are the property of their respective owners. The information provided in these Application Notes is subject to change without notice. The configurations, technical data, and recommendations provided in these Application Notes are believed to be accurate and dependable, but are presented without express or implied warranty. Users are responsible for their application of any products specified in these Application Notes.

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect program at devconnect@avaya.com.