



Avaya Solution & Interoperability Test Lab

Application Notes for Avaya IP Office Release 10.1 with AT&T IP Toll Free Service – Issue 1.0

Abstract

These Application Notes describe the steps for configuring Avaya IP Office Release 10.1 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 Release 10.1 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 10.1 (see **Section 3.2** for call flow examples).

Avaya recommends our customers implement Avaya solutions using appropriate security and encryption capabilities enabled by our products. The testing referenced in this DevConnect Application Note 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.

¹ AVPN uses compressed RTP (cRTP).

² MIS/PNT does not support cRTP.

The test environment described in these Application Notes consisted of:

- A simulated enterprise with Avaya IP Office 10.1, 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.

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.
4. **G.729b is not supported on IP Office Server Edition.** There is no VoIP Silence Suppression parameter for the SIP Line with Server Edition. Therefore G.729 annex B (G.729b) was not tested.
5. **IP Office determines the codec priority.** IP Office will follow the codec priority based on the Codec Selection on the SIP Line VoIP tab, see **Section 5.4.6**. It will not follow the codec priority set by the IPTF service.
6. **Inbound User-to-User Information is not supported with IP Office.** User-to-User Information (UUI) is not supported on inbound SIP trunk calls. IP Office is able to successfully receive an inbound call from AT&T containing UUI, but the UUI data is simply 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:

- Avaya IP Office provides the voice communications services for a particular enterprise site. In the reference configuration, Avaya IP Office runs on an IP 500 V2 platform.
- A single IP Office Application Server was used to provide Voicemail Pro, one-X® Portal, and WebRTC gateway. Voicemail Pro provided the voice messaging capabilities, while one-X® Portal, and WebRTC gateway provided Avaya Communicator for Web capabilities.
- 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) and Avaya Communicator for Web (WebRTC). 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.

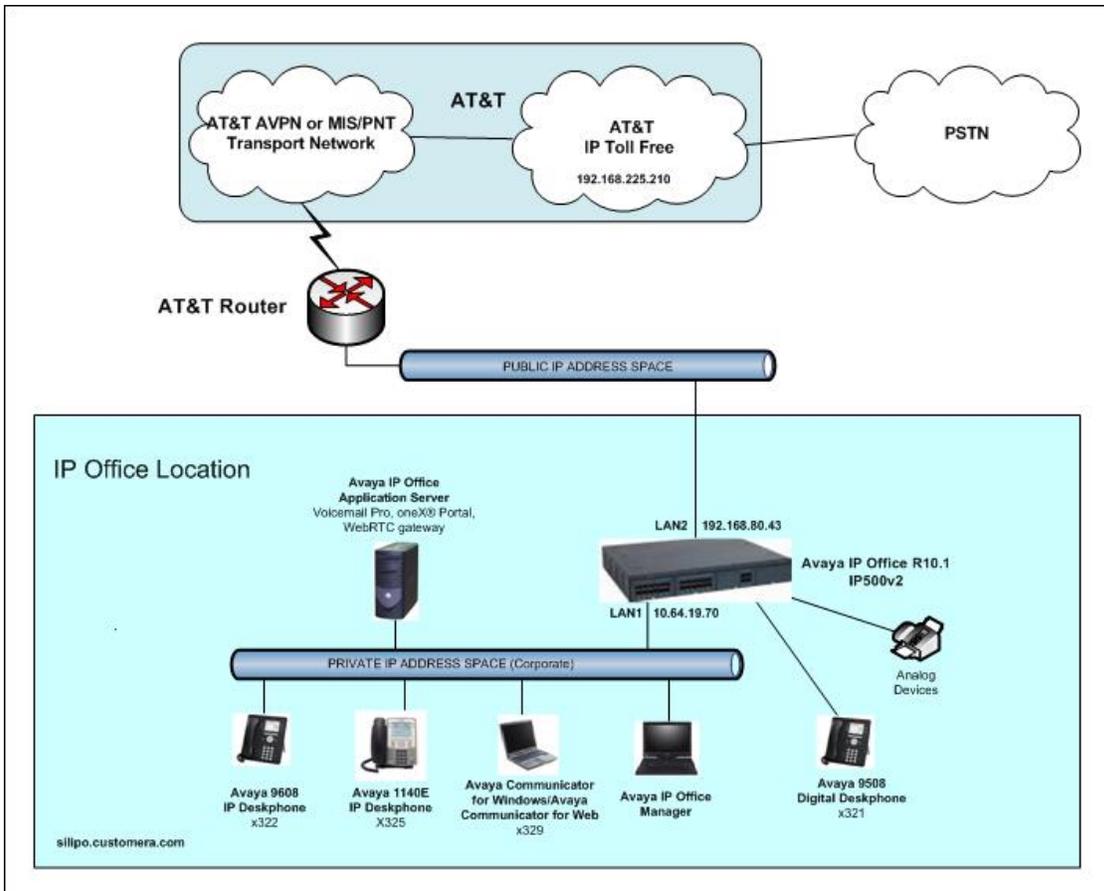


Figure 1: Reference Configuration

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 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 **192.168.80.43** (Avaya IP Office 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 500 V2 Platform	
Private network LAN1 interface	10.64.19.70
Public network LAN2 interface	192.168.80.43
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.

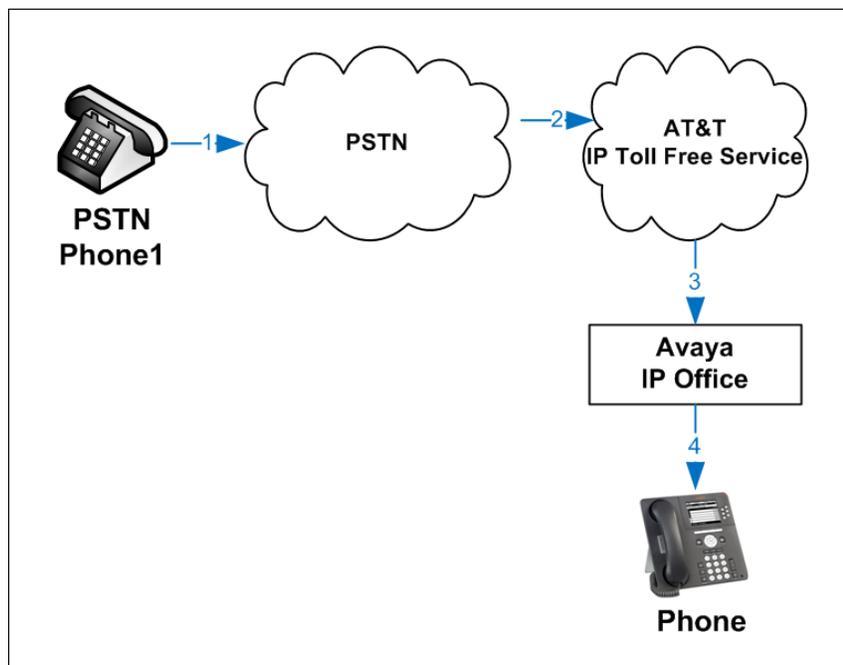


Figure 2: Inbound AT&T IPTF Call

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 IP Office Application 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.

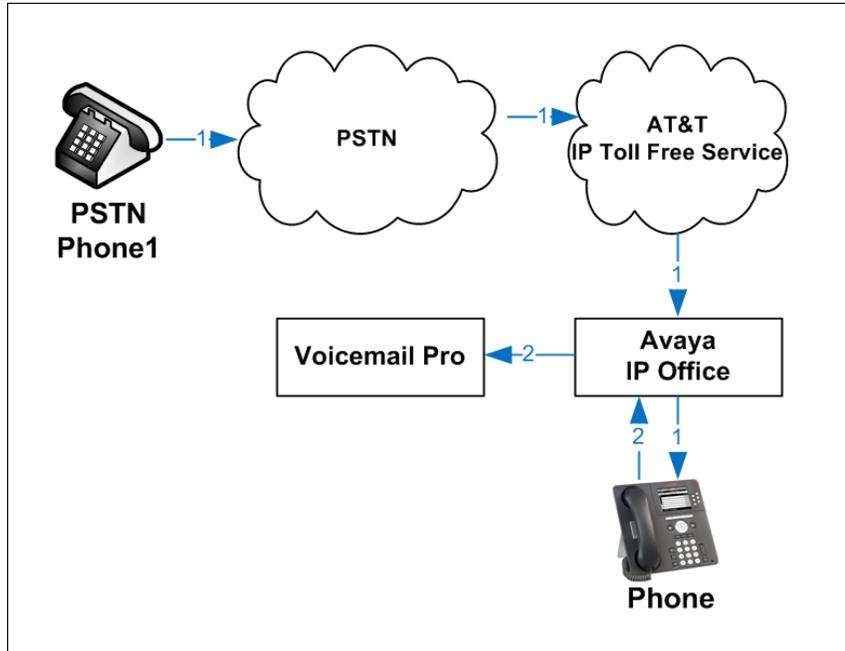


Figure 3: Coverage to Voicemail (Voicemail Pro)

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 500 V2	10.1.0.0.0 build 237
Avaya IP Office Application Server	10.1.0.0.0 build 237
▪ Voicemail Pro	10.1.0.0.0 build 241
▪ Avaya WebRTC Gateway	10.1.0.0.0 build 13
▪ Avaya one-X® Portal for IP Office	10.1.0.0.0 build 305
Avaya Communicator for Windows (SIP)	2.1.30
Avaya Communicator for Web	1.0.17.1914
Avaya 9641G (H.323) IP Deskphone	6.6401
Avaya 1616 (H.323) Telephone	Ha1616ua1_390A.bin
Avaya 1140E (SIP) Telephone	04.04.23
Avaya 9508 Digital Telephone	0.59
Analog Fax device	Ventafax 7.9

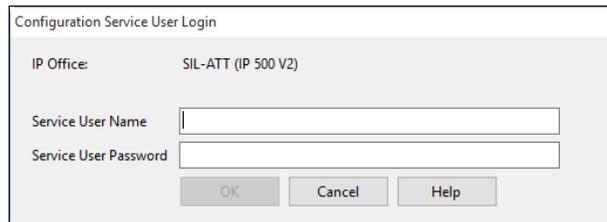
Table 2: Equipment and Software Versions

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 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.



Configuration Service User Login

IP Office: SIL-ATT (IP 500 V2)

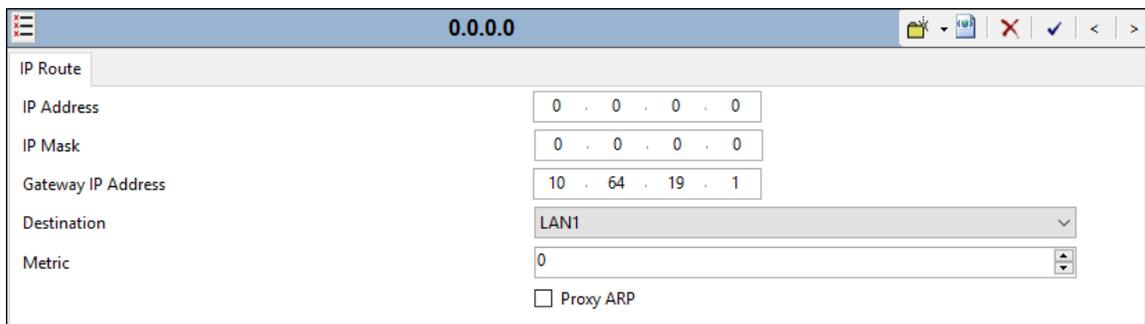
Service User Name:

Service User Password:

OK Cancel Help

5.1. IP Route

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



0.0.0.0

IP Route

IP Address: 0 . 0 . 0 . 0

IP Mask: 0 . 0 . 0 . 0

Gateway IP Address: 10 . 64 . 19 . 1

Destination: LAN1

Metric: 0

Proxy ARP

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).

192.168.0.0

IP Route

IP Address: 192 . 168 . 0 . 0

IP Mask: 255 . 255 . 0 . 0

Gateway IP Address: 192 . 168 . 80 . 1

Destination: LAN2

Metric: 0

Proxy ARP

5.2. Licensing

In the sample configuration, **SIL-ATT** was used as the system name of the IP Office. 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).

IP Offices	License	License Remote Server																																																																					
<ul style="list-style-type: none"> BOOTP (22) Operator (3) SIL-ATT <ul style="list-style-type: none"> System (1) Line (8) Control Unit (3) Extension (23) User (25) Group (5) Short Code (78) Service (0) RAS (1) Incoming Call Route WAN Port (0) Directory (0) Time Profile (0) Firewall Profile (1) IP Route (4) Account Code (0) License (34) Tunnel (0) User Rights (8) ARS (6) Location (2) Authorization Code (0) 	License Type: S	License Mode: License Normal	Licensed Version: 10.0	PLDS Host ID: 111309066506	PLDS File Status: Valid																																																																		
						<table border="1"> <thead> <tr> <th>Feature</th> <th>Instances</th> <th>Status</th> <th>Expiration Date</th> <th>Source</th> </tr> </thead> <tbody> <tr> <td>IPSec Tunnelling</td> <td>1</td> <td>Valid</td> <td>Never</td> <td>PLDS Nodal</td> </tr> <tr> <td>Power User</td> <td>384</td> <td>Valid</td> <td>Never</td> <td>PLDS Nodal</td> </tr> <tr> <td>Customer Service Agent</td> <td>100</td> <td>Valid</td> <td>Never</td> <td>PLDS Nodal</td> </tr> <tr> <td>Customer Service Supervisor</td> <td>100</td> <td>Valid</td> <td>Never</td> <td>PLDS Nodal</td> </tr> <tr> <td>Avaya IP endpoints</td> <td>384</td> <td>Valid</td> <td>Never</td> <td>PLDS Nodal</td> </tr> <tr> <td>IP500 Voice Networking Channels</td> <td>32</td> <td>Valid</td> <td>Never</td> <td>PLDS Nodal</td> </tr> <tr> <td>SIP Trunk Channels</td> <td>128</td> <td>Valid</td> <td>Never</td> <td>PLDS Nodal</td> </tr> <tr> <td>IP500 Universal PRI (Additional cha...</td> <td>100</td> <td>Valid</td> <td>Never</td> <td>PLDS Nodal</td> </tr> <tr> <td>CTI Link Pro</td> <td>1</td> <td>Valid</td> <td>Never</td> <td>PLDS Nodal</td> </tr> <tr> <td>Wave User</td> <td>16</td> <td>Valid</td> <td>Never</td> <td>PLDS Nodal</td> </tr> <tr> <td>3rd Party IP Endpoints</td> <td>384</td> <td>Valid</td> <td>Never</td> <td>PLDS Nodal</td> </tr> <tr> <td>Essential Edition</td> <td>1</td> <td>Valid</td> <td>Never</td> <td>PLDS Nodal</td> </tr> </tbody> </table>	Feature	Instances	Status	Expiration Date	Source	IPSec Tunnelling	1	Valid	Never	PLDS Nodal	Power User	384	Valid	Never	PLDS Nodal	Customer Service Agent	100	Valid	Never	PLDS Nodal	Customer Service Supervisor	100	Valid	Never	PLDS Nodal	Avaya IP endpoints	384	Valid	Never	PLDS Nodal	IP500 Voice Networking Channels	32	Valid	Never	PLDS Nodal	SIP Trunk Channels	128	Valid	Never	PLDS Nodal	IP500 Universal PRI (Additional cha...	100	Valid	Never	PLDS Nodal	CTI Link Pro	1	Valid	Never	PLDS Nodal	Wave User	16	Valid	Never	PLDS Nodal	3rd Party IP Endpoints	384	Valid	Never	PLDS Nodal	Essential Edition	1	Valid	Never	PLDS Nodal
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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.

The screenshot displays the Avaya IP Office configuration interface. On the left is a navigation tree under 'IP Offices' containing various system components like BOOTP, Operator, and System. The main area is titled 'System' and 'SIL-ATT'. A sub-pane shows 'LAN Settings' for 'LAN1' with the following configuration:

IP Address	10 . 64 . 19 . 70
IP Mask	255 . 255 . 255 . 0
Primary Trans. IP Address	0 . 0 . 0 . 0
RIP Mode	None
Enable NAT	<input type="checkbox"/>
Number Of DHCP IP Addresses	200
DHCP Mode	<input type="radio"/> Server <input type="radio"/> Client <input type="radio"/> Dial In <input checked="" type="radio"/> Disabled

An 'Advanced' button is visible at the bottom right of the configuration pane.

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 9600-Series Telephones used in the reference configuration.
- The H.323 Signaling over TLS should be set based on customer needs. In the reference configuration it is set to **Preferred**.
- The **SIP Registrar Enable** box is checked to allow Avaya 11xx (SIP) and Avaya Communicator for Windows (SIP) usage.
- The **SIP Domain Name** used in the reference configuration is **silipo.customera.com**.
- The **SIP Registrar FQDN** used in the reference configuration is **silipo.customera.com**.
- In the **Layer 4 Protocol** section, select **UDP/5060, TCP/5060, and TLS/5061**.
- Let all other values default.

The screenshot displays the Avaya configuration interface for the LAN1 VoIP tab. The interface includes a navigation bar at the top with tabs for System, LAN1, LAN2, DNS, Voicemail, Telephony, Directory Services, System Events, SMTP, SMDR, VCM, VoIP, VoIP Security, and Contact Center. The LAN1 tab is selected, and the VoIP sub-tab is active. The configuration is organized into several sections:

- H.323 Settings:** Includes checkboxes for H.323 Gatekeeper Enable (checked), Auto-create Extension (unchecked), Auto-create User (unchecked), and H.323 Remote Extension Enable (unchecked). The H.323 Signaling over TLS is set to Preferred, and the Remote Call Signaling Port is 1720.
- SIP Settings:** Includes checkboxes for SIP Trunks Enable (checked), SIP Registrar Enable (checked), Auto-create Extension/User (unchecked), and SIP Remote Extension Enable (unchecked). The SIP Domain Name and SIP Registrar FQDN are both set to silipo.customera.com.
- Layer 4 Protocol:** Includes checkboxes for UDP (checked), TCP (checked), and TLS (checked). The corresponding ports are: UDP Port 5060, Remote UDP Port 5060, TCP Port 5060, Remote TCP Port 5060, and TLS Port 5061, Remote TLS Port 5061.
- Challenge Expiration Time (sec):** Set to 10.
- RTP:** Includes a Port Number Range section with Minimum set to 49152 and Maximum set to 53246.

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.

The screenshot shows the configuration interface for the LAN 2 interface. The top navigation bar includes tabs for System, LAN1, LAN2, DNS, Voicemail, Telephony, Directory Services, System Events, SMTP, SMDR, and VCM. The LAN2 tab is selected, and the LAN Settings sub-tab is active. The configuration fields are as follows:

IP Address	192 . 168 . 80 . 43
IP Mask	255 . 255 . 255 . 128
Primary Trans. IP Address	0 . 0 . 0 . 0
Firewall Profile	<None>
RIP Mode	None
Enable NAT	<input checked="" type="checkbox"/>
Number Of DHCP IP Addresses	1
DHCP Mode	<input type="radio"/> Server <input type="radio"/> Client <input type="radio"/> Dial In <input checked="" type="radio"/> Disabled

An **Advanced** button is located at the bottom right of the configuration area.

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.

The screenshot displays the Avaya configuration interface for the LAN2 VoIP tab. The interface includes a navigation bar at the top with tabs for System, LAN1, LAN2, DNS, Voicemail, Telephony, Directory Services, System Events, SMTP, SMDR, VCM, VoIP, VoIP Security, and Contact Center. The LAN2 tab is selected, and the VoIP sub-tab is active. The configuration is organized into several sections:

- H.323 Gatekeeper Enable:** Includes checkboxes for H.323 Gatekeeper Enable, Auto-create Extension, Auto-create User, and H.323 Remote Extension Enable. The H.323 Signaling over TLS is set to Disabled, and the Remote Call Signaling Port is 1720.
- SIP Trunks Enable:** The checkbox is checked. Below it are fields for SIP Domain Name and SIP Registrar FQDN.
- SIP Registrar Enable:** Includes checkboxes for SIP Registrar Enable, Auto-create Extension/User, and SIP Remote Extension Enable.
- Layer 4 Protocol:** Includes checkboxes for UDP, TCP, and TLS. The ports are configured as follows:
 - UDP: 5060 (Local), 5060 (Remote)
 - TCP: 5060 (Local), 5060 (Remote)
 - TLS: 5061 (Local), 5061 (Remote)
- Challenge Expiration Time (sec):** Set to 10.
- RTP Port Number Range:** The Minimum port is 16384 and the Maximum port is 32766.

- 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.

RTP

Port Number Range

Minimum Maximum

Port Number Range (NAT)

Minimum Maximum

Enable RTCP Monitoring on Port 5005

RTCP collector IP address for phones

Keepalives

Scope Periodic timeout

Initial keepalives

DiffServ Settings

DSCP(Hex) Video DSCP (Hex) DSCP Mask (Hex) SIG DSCP (Hex)

DSCP Video DSCP DSCP Mask SIG DSCP

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**.

The screenshot shows the 'Network Topology' configuration window. At the top, there are tabs for 'System', 'LAN1', 'LAN2', 'DNS', 'Voicemail', 'Telephony', 'Directory Services', 'System Events', 'SMTP', 'SMDR', 'VCM', 'VoIP', 'VoIP Security', and 'Contact Center'. Below these, there are sub-tabs for 'LAN Settings', 'VoIP', and 'Network Topology'. The 'Network Topology Discovery' section contains the following fields and controls:

- STUN Server Address:** An empty text input field.
- STUN Port:** A dropdown menu set to '3478'.
- Firewall/NAT Type:** A dropdown menu set to 'Open Internet'.
- Binding Refresh Time (sec):** A dropdown menu set to '120'.
- Public IP Address:** A text input field containing '192 . 168 . 80 . 43'.
- Public Port:** A group of three dropdown menus: 'UDP' set to '5060', 'TCP' set to '0', and 'TLS' set to '0'.
- Run STUN on startup:** An unchecked checkbox.
- Buttons:** 'Run STUN' and 'Cancel' buttons are located to the right of the IP address field.

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.

The screenshot shows the 'Telephony' configuration window with several tabs: 'Telephony', 'Park & Page', 'Tones & Music', 'Ring Tones', 'SM', 'Call Log', and 'TUI'. The 'Telephony' tab is active. On the left, under 'Analogue Extensions', there are settings for 'Default Outside Call Sequence' (Normal), 'Default Inside Call Sequence' (Ring Type 1), 'Default Ring Back Sequence' (Ring Type 2), and 'Restrict Analogue Extension Ringer Voltage' (unchecked). Below this are various time and delay settings: 'Dial Delay Time (sec)' (4), 'Dial Delay Count' (0), 'Default No Answer Time (sec)' (15), 'Hold Timeout (sec)' (0), 'Park Timeout (sec)' (300), 'Ring Delay (sec)' (5), 'Call Priority Promotion Time (sec)' (Disabled), 'Default Currency' (USD), 'Default Name Priority' (Favor Trunk), 'Media Connection Preservation' (Disabled), 'Phone Failback' (Manual), and 'Login Code Complexity'. On the right, the 'Companding Law' section is highlighted with a red box. It has two columns: 'Switch' and 'Line'. Both columns have 'U-Law' selected with a radio button, and 'A-Law' and 'A-Law Line' are unselected. Below this, there are several other checkboxes: 'DSS Status' (unchecked), 'Auto Hold' (checked), 'Dial By Name' (checked), 'Show Account Code' (checked), 'Inhibit Off-Switch Forward/Transfer' (unchecked and highlighted with a red box), 'Restrict Network Interconnect' (unchecked), 'Include location specific information' (unchecked), 'Drop External Only Impromptu Conference' (unchecked), 'Visually Differentiate External Call' (unchecked), 'Unsupervised Analog Trunk Disconnect Handling' (unchecked), 'High Quality Conferencing' (checked), and 'Digital/Analogue Auto Create User' (checked).

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 codecs that meet the needs of the customer. In the reference configuration, the default selection was used.

- 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 Error! Reference source not found.**) also specify the default selection, 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.722 first, and SIP trunk calls will offer G.729A first.

The screenshot shows the configuration interface for a SIP Line. The 'VoIP' tab is active. The 'RFC2833 Default Payload' is set to 100. The 'Available Codecs' list includes G.711 ULAW 64K, G.711 ALAW 64K, G.722 64K, G.729(a) 8K CS-ACELP, and G.723.1 6K3 MP-MLQ. The 'Default Codec Selection' shows 'Unused' in the 'Unused' list and 'Selected' in the 'Selected' list, which includes G.722 64K, G.711 ULAW 64K, G.711 ALAW 64K, G.729(a) 8K CS-ACELP, and G.723.1 6K3 MP-MLQ.

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

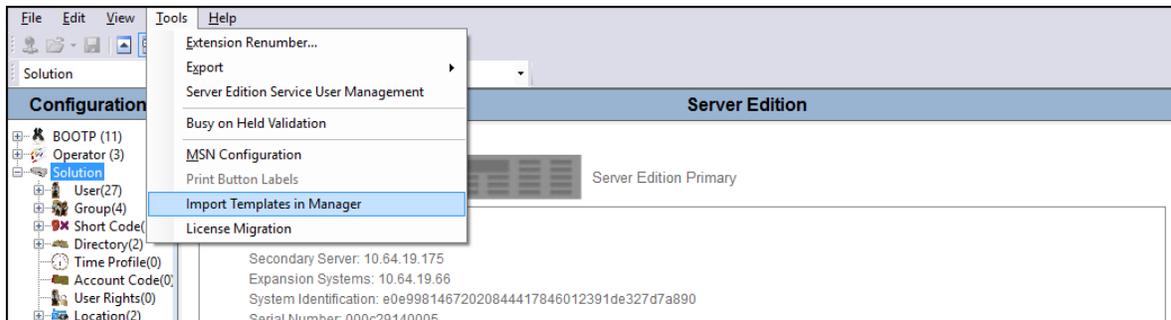
- Transport – Second Explicit DNS Server
- SIP Credentials – Registration Requirements
- SIP Advanced Engineering

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

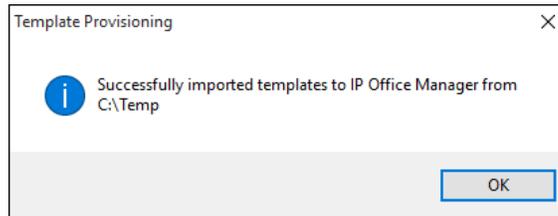
Note – DevConnect generated SIP Line templates are always exported in an XML format. These XML templates do not include sensitive customer specific information and are therefore suitable for distribution. The XML format templates can be used to create SIP trunks on both IP Office Standard Edition (IP500 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**.



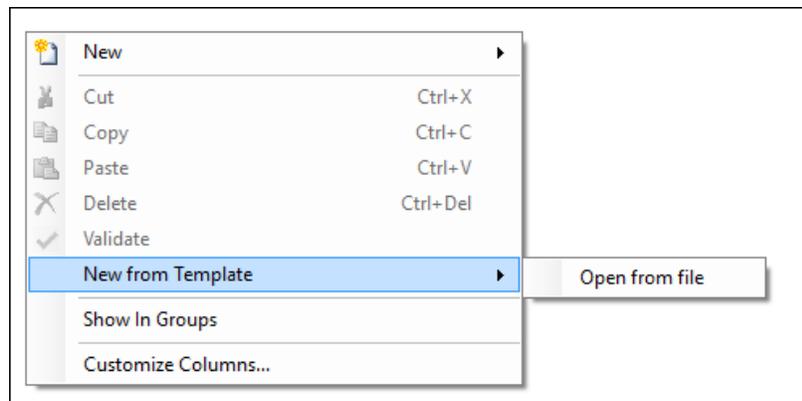
3. A folder browser will open (not shown). Select the directory used in **step 1** to store the template(s) (e.g., *\temp*). In the reference configuration, template file **IPO10TF.xml** was imported. The template files are automatically copied into the IP Office default template location, **C:\Program 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.

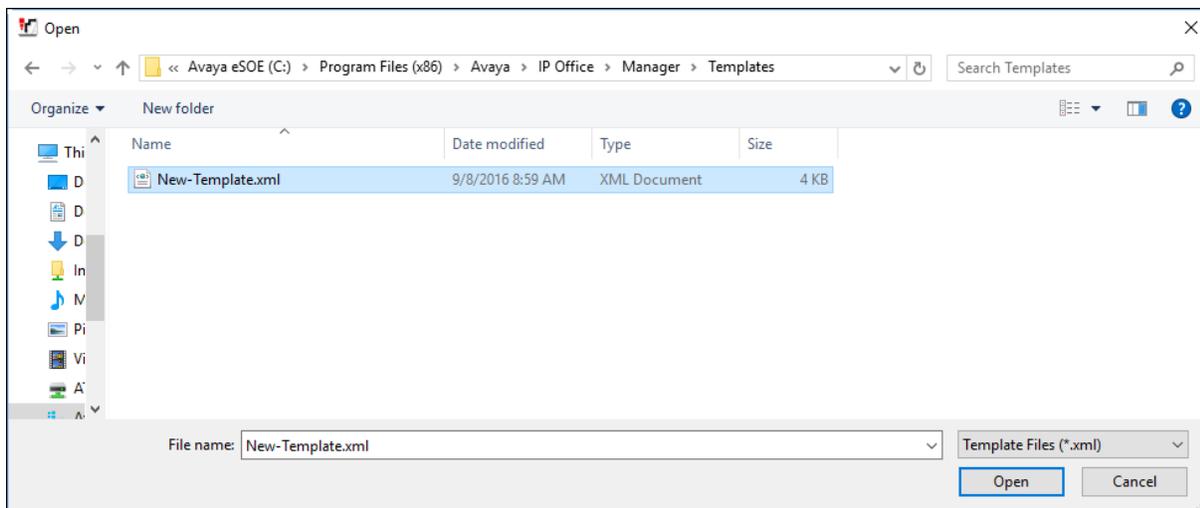


5.4.2. Creating a SIP Trunk from an XML Template

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 22 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).

SIP Line	Transport	SIP URI	VoIP	T38 Fax	SIP Credentials	SIP Advanced	Engineering
Line Number	22						<input checked="" type="checkbox"/>
ITSP Domain Name	192.168.225.210						Check OOS <input checked="" type="checkbox"/>
Local Domain Name	192.168.80.43						
URI Type	SIP URI						
Location	Cloud						
Prefix							
National Prefix	0						
International Prefix	00						
Country Code							
Name Priority	System Default						
Description	AT&T IPTF						
							Session Timers
							Refresh Method
							Timer (sec)
							Redirect and Transfer
							Incoming Supervised REFER
							Outgoing Supervised REFER
							Send 302 Moved Temporarily
							Outgoing Blind REFER

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 **None**.
- **Calls Route via Registrar:** Verify this field is checked (default).
- **Click OK** (not shown).

The screenshot shows the 'Transport' tab of the SIP Line configuration window. The 'ITSP Proxy Address' field is set to '192.168.225.210'. Under the 'Network Configuration' section, 'Layer 4 Protocol' is set to 'UDP', 'Send Port' is '5060', and 'Use Network Topology Info' is set to 'None'. 'Listen Port' is also '5060'. 'Explicit DNS Server(s)' are set to '0 . 0 . 0 . 0'. 'Calls Route via Registrar' is checked. The 'Separate Registrar' field is empty.

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**.
- 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., **22**. This value references the table created with **Incoming Call Routes** in **Section 5.6**.
- **Outgoing Group**: Set to an unused group number, e.g., **22**.
- **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	Credential	Max Calls
1	22, 22	Auto	Auto	Auto	None	PAI		None	None	0: <Non...	10

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: 22

Outgoing Group: 22

Max Sessions: 10

Buttons: Add..., Remove, Edit..., 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.
- 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 in 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, G.722 64K, and G.723.1 6K3 MP-MLQ. The 'Selected' list contains G.729(a) 8K CS-ACELP and G.711 ULAW 64K. There are navigation buttons between the lists: >>> (move right), <<< (move left), ↑ (move up), ↓ (move down), and >>> (move right). To the right of the lists are several checkboxes: VoIP Silence Suppression, Local Hold Music, Re-invite Supported (checked), Codec Lockdown, Allow Direct Media Path, Force direct media with phones, PRACK/100rel Supported, and G.711 Fax ECAN. Below the lists are three dropdown menus: Fax Transport Support (set to T38), DTMF Support (set to RFC2833), and Media Security (set to Disabled).

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 Error! Reference source not found.** for T.38 fax settings.

Select the **SIP Line** → **T.38 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).

The screenshot shows the 'T38 Fax' configuration tab for a SIP line. The 'Use Default Values' checkbox is highlighted with a red box. The configuration includes the following settings:

Field	Value
T38 Fax Version	0
Transport	UDPTL
Redundancy	Low Speed: 0, High Speed: 0
TCF Method	Trans TCF
Max Bit Rate (bps)	14400
EFlag Start Timer (ms)	2600
EFlag Stop Timer (ms)	2300
Tx Network Timeout (sec)	150
Scan Line Fix-up	Checked
TFOP Enhancement	Checked
Disable T30 ECM	Unchecked
Disable EFlags For First DIS	Unchecked
Disable T30 MR Compression	Unchecked
NSF Override	Country Code: 0, Vendor Code: 0

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 Advanced' configuration tab for a SIP Line. The interface is divided into several sections:

- Addressing:** Association Method (By Source IP address), Call Routing Method (Request URI), Suppress DNS SRV Lookups (unchecked).
- Identity:** A list of checkboxes for various identity-related settings, including 'Use "phone-context"', 'Add user=phone', 'Use + for International', 'Use PAI for Privacy', 'Use Domain for PAI', 'Swap From and PAI/Diversion', 'Caller ID from From header', 'Send From In Clear', 'Cache Auth Credentials' (checked), 'User-Agent and Server Headers' (text input), 'Send Location Info' (Never), 'Add UUI header', and 'Add UUI header to redirected calls'.
- Media:** A list of checkboxes and dropdowns for media-related settings, including 'Allow Empty INVITE', 'Send Empty re-INVITE', 'Allow To Tag Change', 'P-Early-Media Support' (None), 'Send SilenceSup=Off', 'Force Early Direct Media', 'Media Connection Preservation' (Disabled), and 'Indicate HOLD' (checked and highlighted with a red box).
- Call Control:** A list of settings for call control, including 'Call Initiation Timeout (s)' (4), 'Call Queuing Timeout (mins)' (5), 'Service Busy Response' (486 - Busy Here), 'on No User Responding Send' (408-Request Timeout), 'Action on CAC Location Limit' (Allow Voicemail), 'Suppress Q.850 Reason Header', 'Emulate NOTIFY for REFER', and 'No REFER if using Diversion'.

5.5. Users, Extensions, and Hunt Groups

In this section, examples of IP Office Users, Extensions, and Groups will be illustrated. In the interests of brevity, not all users and extensions shown in **Figure 1** will be presented, since the configuration can be easily extrapolated to other users. To add a User, right click on **User** in the **Navigation** pane, and select **New**. To edit an existing User, select **User** in the **Navigation** pane, and select the appropriate user to be configured in the **Group** pane.

5.5.1. User 321 (Digital)

The following screen shows the **User** tab for User 321. As shown in **Figure 1**, this user corresponds to the Avaya Digital 9508.

The screenshot displays the configuration page for a user in the Avaya IP Office system. The interface includes a navigation bar at the top with tabs for various settings: User, Voicemail, DND, Short Codes, Source Numbers, Telephony, Forwarding, Dial In, Voice Recording, Button Programming, Menu Programming, Mobility, and Group. The 'User' tab is active. The configuration fields are as follows:

Name	Avaya9508
Password	••••
Confirm Password	••••
Unique Identity	
Conference PIN	
Confirm Audio Conference PIN	
Account Status	Enabled
Full Name	
Extension	321
Email Address	
Locale	
Priority	5
System Phone Rights	None
Profile	Basic User

Below the profile dropdown, there is a list of checkboxes for additional features:

- Receptionist
- Enable Softphone
- Enable one-X Portal Services
- Enable one-X TeleCommuter
- Enable Remote Worker
- Enable Communicator
- Enable Mobile VoIP Client
- Send Mobility Email
- Web Collaboration

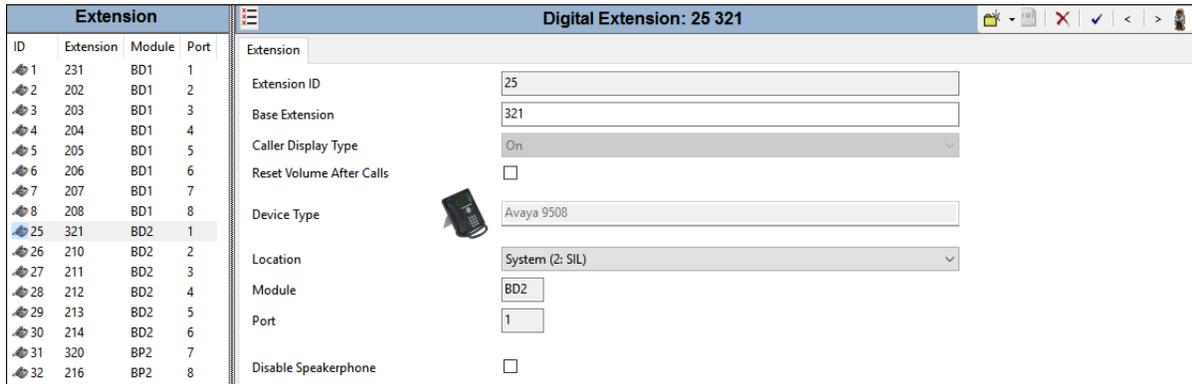
The following screen shows the **SIP** tab for User 321. The **SIP Name** and **Contact** parameters are configured with the DID number of the user, 303-555-9321. These parameters configure the user part of the SIP URI in the From header for outgoing SIP trunk calls, and allow matching of the SIP URI for incoming calls, without having to enter this number as an explicit SIP URI for the SIP Line. The **SIP Display Name (Alias)** parameter can optionally be configured with a descriptive name. If all calls involving this user and a SIP Line should be considered private, then the **Anonymous** box may be checked to withhold the user's information from the network. See **Section Error! Reference source not found.** for a method of using a short code (rather than static user provisioning) to place an anonymous call.

Voice Recording	Button Programming	Menu Programming	Mobility	Group Membership	Announcements	SIP	Personal Directory	Web Self-Administrati...
SIP Name	3035559321							
SIP Display Name (Alias)	Avaya9508							
Contact	3035559321							
<input type="checkbox"/> Anonymous								

From **Figure 1**, note that user 321 will use the Mobile Twinning feature. The following screen shows the **Mobility** tab for user 321. The **Mobility Features** and **Mobile Twinning** boxes are checked. The **Twinned Mobile Number** field is configured with the number to dial to reach the twinned mobile telephone, in this case 913035582177. Other options can be set according to customer requirements.

Voiceemail	DND	Short Codes	Source Numbers	Telephony	Forwarding	Dial In	Voice Recording	Button Programming	Menu Programming	Mobility
<input type="checkbox"/> Internal Twinning										
Twinned Handset: <None>										
Maximum Number of Calls: 1										
<input type="checkbox"/> Twin Bridge Appearances										
<input type="checkbox"/> Twin Coverage Appearances										
<input type="checkbox"/> Twin Line Appearances										
<input checked="" type="checkbox"/> Mobility Features										
<input type="checkbox"/> Mobile Twinning										
Twinned Mobile Number (including dial access code): 91303552177										
Twinning Time Profile: <None>										
Mobile Dial Delay (sec): 0										
Mobile Answer Guard (sec): 0										
<input type="checkbox"/> Hunt group calls eligible for mobile twinning										
<input type="checkbox"/> Forwarded calls eligible for mobile twinning										
<input type="checkbox"/> Twin When Logged Out										
<input type="checkbox"/> one-X Mobile Client										
<input checked="" type="checkbox"/> Mobile Call Control										
<input checked="" type="checkbox"/> Mobile Callback										

The following screen shows the Extension information for this user. To view, select **Extension** from the **Navigation** pane, and the appropriate extension from the **Group** pane.



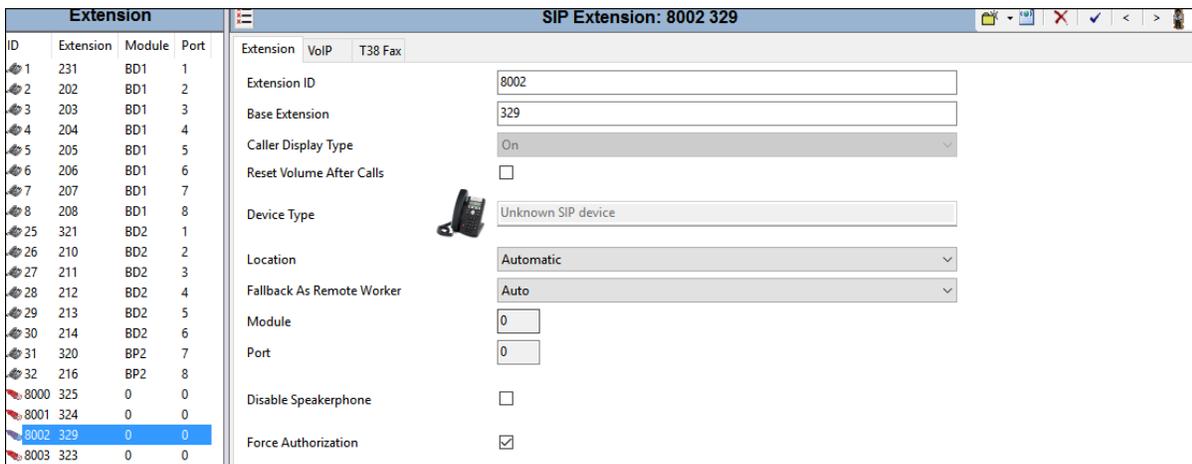
ID	Extension	Module	Port
1	231	BD1	1
2	202	BD1	2
3	203	BD1	3
4	204	BD1	4
5	205	BD1	5
6	206	BD1	6
7	207	BD1	7
8	208	BD1	8
25	321	BD2	1
26	210	BD2	2
27	211	BD2	3
28	212	BD2	4
29	213	BD2	5
30	214	BD2	6
31	320	BP2	7
32	216	BP2	8

Digital Extension: 25 321

Extension ID: 25
 Base Extension: 321
 Caller Display Type: On
 Reset Volume After Calls:
 Device Type: Avaya 9508
 Location: System (2: SIL)
 Module: BD2
 Port: 1
 Disable Speakerphone:

5.5.2. User 329 (Avaya Communicator for Windows)

A new SIP extension may be added by right-clicking on **Extension** in the **Navigation** pane and selecting **New SIP Extension**. Alternatively, an existing SIP extension may be selected in the **Group** pane. The following screen shows the **Extn** tab for the extension corresponding to an Avaya Communicator for Windows. The **Base Extension** field is populated with 329, the extension assigned to the Avaya Communicator for Windows. Ensure the **Force Authorization** box is checked.



ID	Extension	Module	Port
1	231	BD1	1
2	202	BD1	2
3	203	BD1	3
4	204	BD1	4
5	205	BD1	5
6	206	BD1	6
7	207	BD1	7
8	208	BD1	8
25	321	BD2	1
26	210	BD2	2
27	211	BD2	3
28	212	BD2	4
29	213	BD2	5
30	214	BD2	6
31	320	BP2	7
32	216	BP2	8
8000	325	0	0
8001	324	0	0
8002	329	0	0
8003	323	0	0

SIP Extension: 8002 329

Extension ID: 8002
 Base Extension: 329
 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 Authorization:

The following screen shows the **VoIP** tab for the extension. The **IP Address** field may be left blank. For the **Reserve License** parameter, select **Reserve Avaya IP endpoint license** from the drop-down box. The **Codec Selection** parameter may retain the default setting **System Default** to follow the system configuration shown in **Section Error! Reference source not found.** Alternatively, “Custom” may be selected to allow the codecs to be configured for this extension, using the arrow keys to select and order the codecs. Other fields may retain default values.

The screenshot displays the VoIP configuration page for an extension. At the top, there are tabs for 'Extension', 'VoIP', and 'T38 Fax'. The 'VoIP' tab is active. The 'IP Address' field contains '0 . 0 . 0 . 0'. The 'Codec Selection' dropdown is set to 'System Default'. Below this, there are two lists: 'Unused' (empty) and 'Selected' (containing G.722 64K, G.711 ULAW 64K, G.711 ALAW 64K, G.729(a) 8K CS-ACELP, and G.723.1 6K3 MP-MLQ). The 'Reserve License' dropdown is set to 'Reserve Avaya IP endpoint license'. Other dropdowns include 'Fax Transport Support' (None), 'TDM->IP Gain' (Default), 'IP->TDM Gain' (Default), 'DTMF Support' (RFC2833), and '3rd Party Auto Answer' (None). The 'Media Security' dropdown is set to 'Same as System (Preferred)'. An expanded section for 'Advanced Media Security Options' shows 'Same As System' checked, with 'Encryptions' (RTP, RTCP), 'Authentication' (RTP, RTCP), 'Replay Protection', 'SRTP Window Size' (64), and 'Crypto Suites' (SRTP_AES_CM_128_SHA1_80, SRTP_AES_CM_128_SHA1_32) all checked.

The following screen shows the **User** tab for user 329 corresponding to an Avaya Communicator for Windows. The **Extension** parameter is populated with extension 329. The **Password** fields will be used by the Avaya Communicator for Windows user as the login password. In the reference configuration, the **Profile** is set to **Power User**, with **Enable Softphone**, and **Enable Communicator** checked.

Name	Mobile
Password	••••••••
Confirm Password	••••••••
Unique Identity	
Conference PIN	
Confirm Audio Conference PIN	
Account Status	Enabled
Full Name	onex mobile
Extension	329
Email Address	
Locale	
Priority	5
System Phone Rights	None
Profile	Power User
	<input type="checkbox"/> Receptionist <input checked="" type="checkbox"/> Enable Softphone <input checked="" type="checkbox"/> Enable one-X Portal Services <input checked="" type="checkbox"/> Enable one-X TeleCommuter <input checked="" type="checkbox"/> Enable Remote Worker <input checked="" type="checkbox"/> Enable Communicator <input checked="" type="checkbox"/> Enable Mobile VoIP Client <input type="checkbox"/> Send Mobility Email <input type="checkbox"/> Web Collaboration

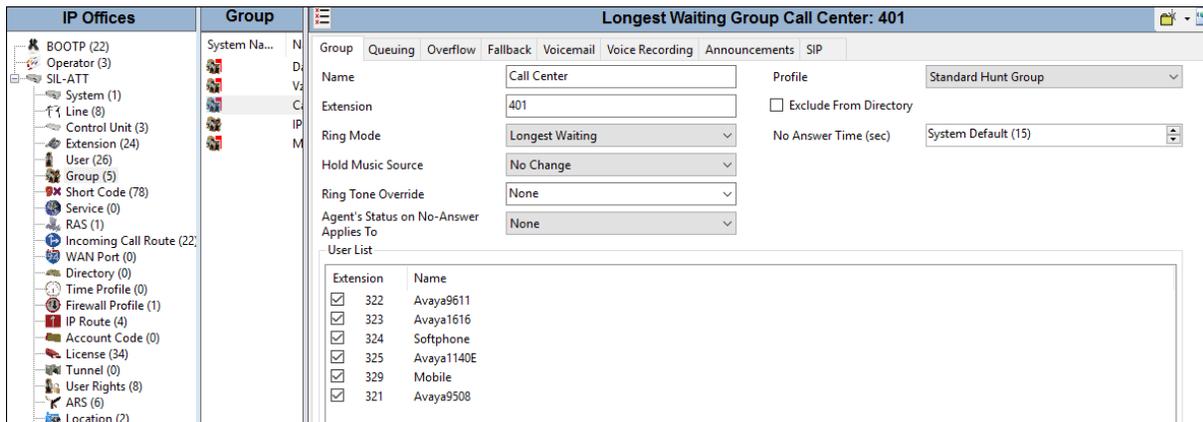
Like other users previously illustrated, the **SIP** tab for the user with extension 329 is configured with a **SIP Name** and **Contact** specifying the user's AT&T DID number.

SIP Name	4695554881
SIP Display Name (Alias)	Mobile
Contact	4695554881
	<input type="checkbox"/> Anonymous

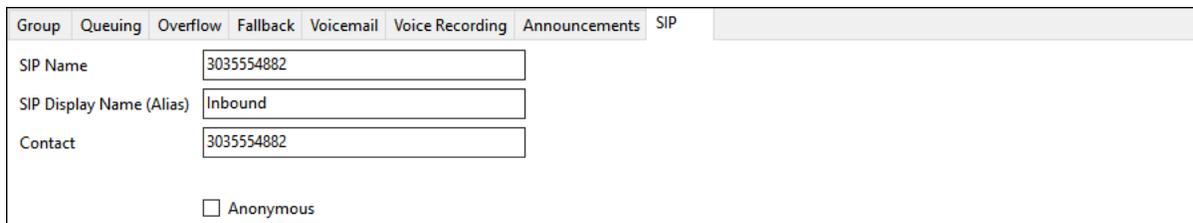
5.5.3. Hunt Groups

During the verification of these Application Notes, users could also receive incoming calls as members of a hunt group. To configure a new hunt group, right-click **Group** 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.

The following screen shows the **Group** tab for hunt group 401. The telephone extensions in the **User List** are rung based on the extension that has been unused for the longest period, due to the **Ring Mode** setting **Longest Waiting** (i.e., most idle user to receive the next call). Click the **Edit** button to change the **User List**.



The following screen shows the **SIP** tab for hunt group 401. The **SIP Name** and **Contact** are configured with AT&T DID 3035554882. Later, in **Section 5.7**, an Incoming Call Route will map 3035554882 to this hunt group based on the information entered on this tab.



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

5.6. 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.

5.6.1. Calls to IP Office Stations and Hunt Groups

In the example below, the incoming number **000008885551025** is directed to H.323 phone 322.

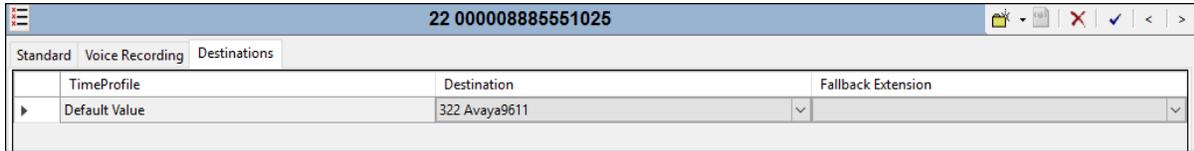
1. On the **Standard** tab enter the following:

- **Line Group ID:** Enter the SIP Line defined in **Section 5.4** (e.g., **22**).
- **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 displays the configuration interface for an Incoming Call Route. On the left is a navigation tree under 'IP Offices' with categories like BOOTP, Operator, System, Line, Control Unit, Extension, User, Group, Short Code, Service, RAS, and Incoming Call Route. The 'Incoming Call Route' category is expanded, showing several entries, with '22 000008885551025' selected. The main pane shows the configuration for this route, with tabs for 'Standard', 'Voice Recording', and 'Destinations'. The 'Standard' tab is active, showing the following fields:

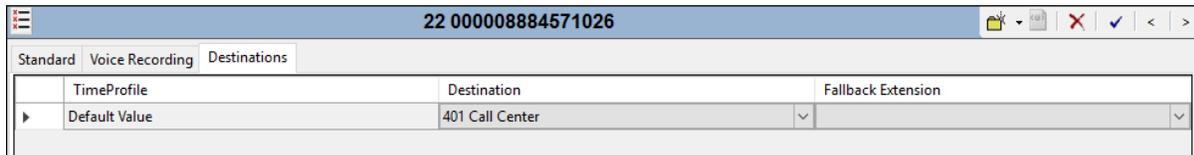
Field	Value
Bearer Capability	Any Voice
Line Group ID	22
Incoming Number	000008885551025
Incoming Sub Address	
Incoming CLI	
Locale	
Priority	1 - Low
Tag	
Hold Music Source	System Source
Ring Tone Override	None

2. On the **Destinations** tab enter the following:
 - In the **Destinations** column, select extension **322** 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	22	
Incoming Number	000008884571026	
Incoming Sub Address		
Incoming CLI		
Locale		
Priority	1 - Low	
Tag	732-555-4029	
Hold Music Source	System Source	
Ring Tone Override	None	

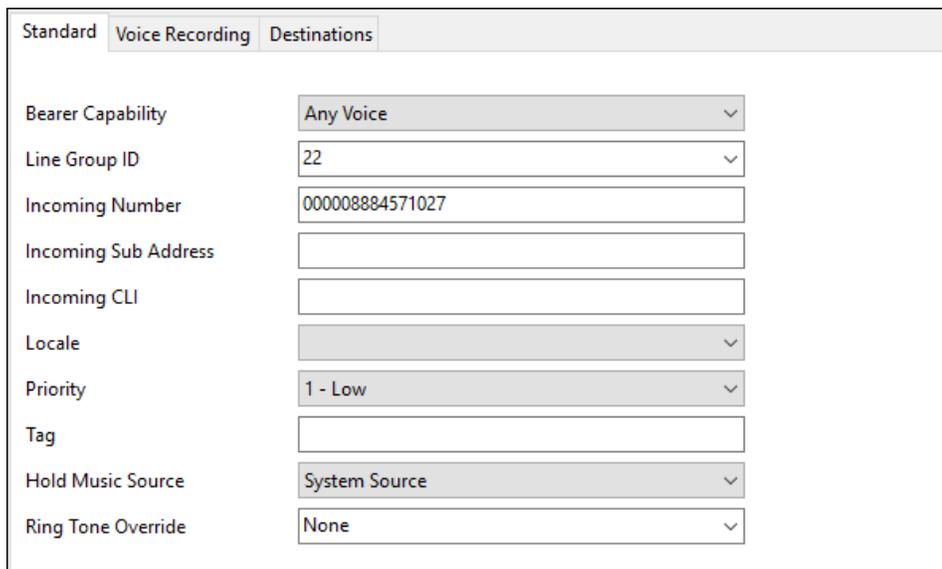


5.6.2. Calls to Voicemail Pro Scripts

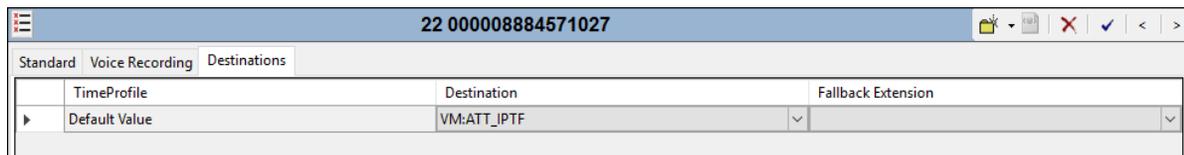
As described in **Sections 5.6.1** and **5.8**, Voicemail Pro scripts are defined with specific names. These script names are specified as destinations in the Incoming Call Route table.

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 5.6.1**, 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).



Field	Value
Bearer Capability	Any Voice
Line Group ID	22
Incoming Number	000008884571027
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	

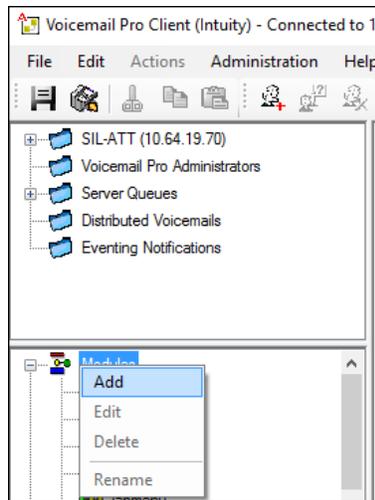
5.7. Call Center Provisioning in Voicemail Pro

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

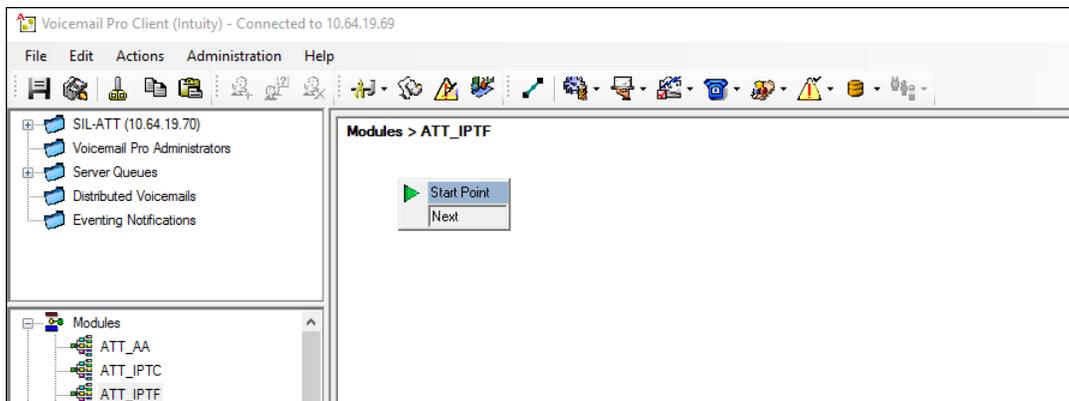
In the reference configuration, Voicemail Pro is used for Voicemail processing as well as for simulating basic Call Center functionality.

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:

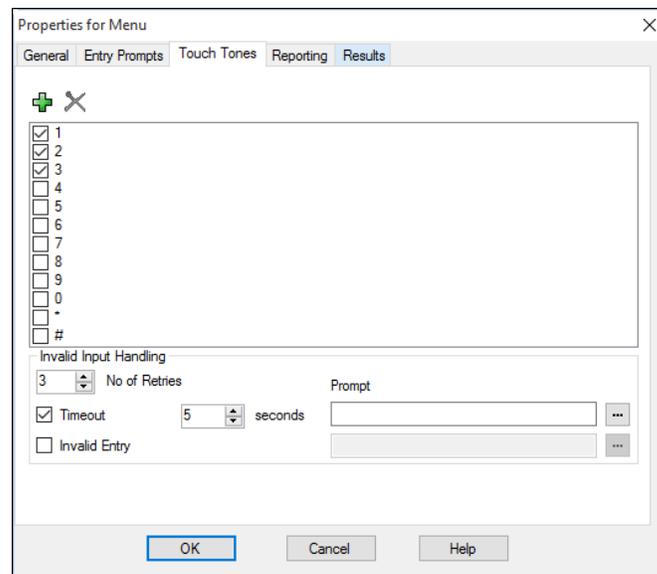
1. Hunt Groups **Call Center** and **Support** are created in IP Office (**Section Error!** Reference source not found.).
2. User 329 is created in IP Office (**Section Error!** Reference source not found.).
3. Incoming Call Route for DNIS digits **000008885551027** is defined for access to the Auto-Attendant script (**Section 5.6.2**).
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.

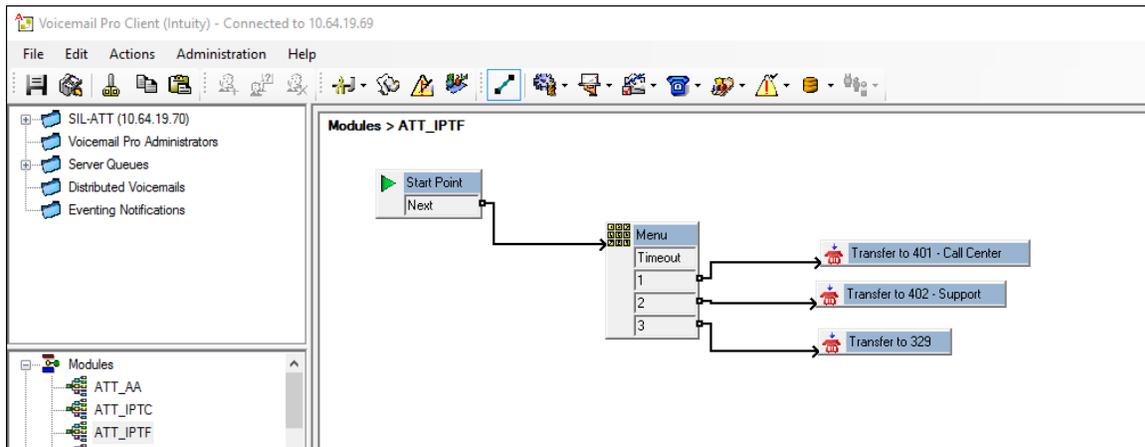


- Click on the **Start Point** icon  to activate the script options at the top of the screen. From the options, select the **Basic Actions** icon , select the **Menu** icon , and click on the work area to place the **Menu** icon.
 - Double click the **Start Point** icon.
 - On the **General** tab → **Token Name**, enter **Start Point** and click **OK** (not shown).
 - Double click the **Menu** icon.
 - On the **General** tab → **Token Name**, enter **Menu** (not shown).
 - 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.
 - On the **Touch Tone** tab:
 - Select **1, 2,** and **3** as the possible entry digits.
 - Select **3** for **No of Retries**.
 - 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”).
 - Double click on the first **Transfer** icon (“**Call Center**”).
 - On the **General** tab → **Token Name** = **Transfer to 401 - Call Center** (not shown).
 - On the **Specific** tab → **Destination** = **401** (not shown).
 - 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 (“Ext329”).
1. On the **General** tab, **Token Name = Transfer to 329** (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., **000008885551027**), 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.

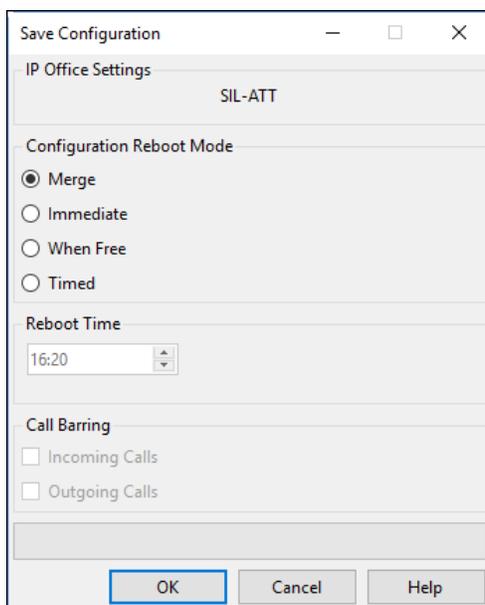
5.8. Saving Configuration Changes to Avaya IP Office

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. 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.

7. Verification Steps

The following procedures may be used to verify the Avaya IP Office R10.1 with the AT&T IP Toll Free service configuration.

7.1. AT&T IP Toll Free Service

The following scenarios may be executed to verify Avaya IP Office R10.1 functionality with the AT&T IPTF service:

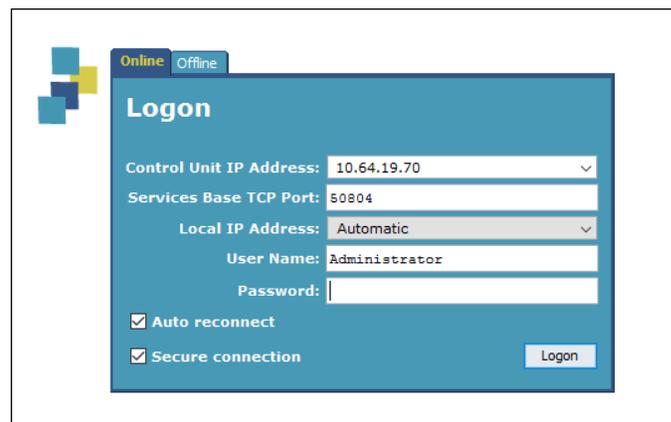
- Place inbound calls, answer the calls, and verify that two-way talk path exists. Verify that the calls remain stable for several minutes and disconnects properly.
- Incoming calls using the G.729A and G.711 ULAW codecs.
- Verify basic call functions such as hold, transfer, and conference.
- Place an 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.

7.2. Avaya IP Office 10.1

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

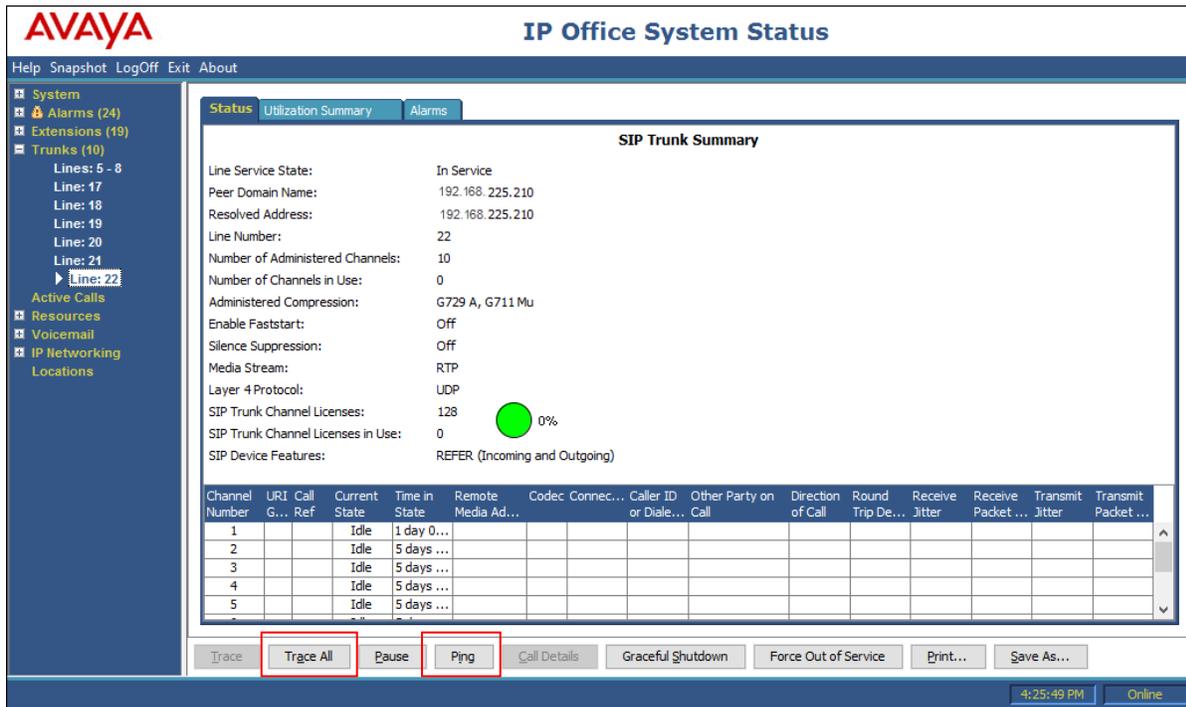
7.2.1. System Status Application

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**.

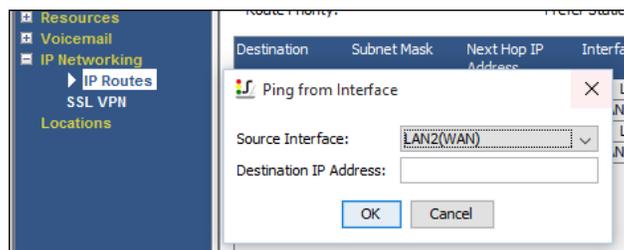


The screenshot shows the 'Logon' screen of the Avaya IP Office System Status application. The interface is blue and white. At the top left, there are 'Online' and 'Offline' status indicators. The main title is 'Logon'. Below the title, there are several input fields: 'Control Unit IP Address' (with the value '10.64.19.70'), 'Services Base TCP Port' (with the value '50804'), 'Local IP Address' (with the value 'Automatic'), 'User Name' (with the value 'Administrator'), and 'Password'. There are two checkboxes: 'Auto reconnect' and 'Secure connection', both of which are checked. A 'Logon' button is located at the bottom right of the form.

After logging in, select **Trunks** → **Line: 2** from the left navigation menu. (SIP Line 22 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).

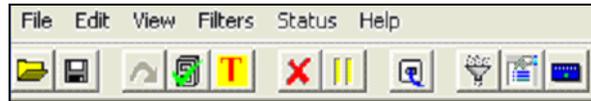


By navigating to **IP Networking** → **IP Routes**, and clicking on **Ping**, an IP Office **Source Interface**, and any **Destination IP Address**, may be specified for a ping by clicking **OK**.



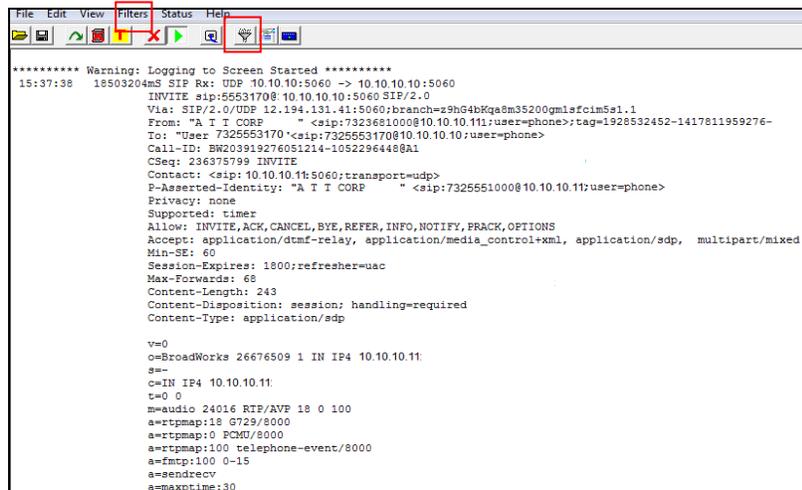
7.2.2. System Monitor Application

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**.

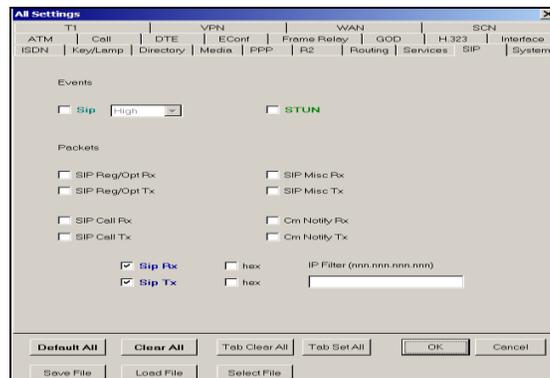


The Monitor will be active at startup. To pause the Monitor, press the Pause  button.

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.



8. Conclusion

As illustrated in these Application Notes, Avaya IP Office R10.1 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.

9. References

Avaya:

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

- [1] *Deploying IP Office™ Platform Server Edition Solution*, Release 10.1, June 2017
- [2] *IP Office™ Platform 10.1, Deploying Avaya IP Office™ Platform IP500 V2*, September 2017
- [3] *Administering Avaya IP Office™ Platform with Web Manager*, Release 10.1, June 2017.
- [4] *Administering Avaya IP Office™ Platform with Manager*, Release 10.1, June 2017.

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/>

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