Avaya Solution & Interoperability Test Lab

Application Notes for Avaya IP Office Release 11.0 with AT&T IP Toll Free Service – Issue 1.0

Abstract

These Application Notes describe the steps for configuring Avaya IP Office Release 11.0 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.

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1. Introduction

These Application Notes describe the steps for configuring Avaya IP Office Release 11.0 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 AVPN[1] or MIS/PNT[2] 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 11.0 (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.

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The test environment described in these Application Notes consisted of:

- A simulated enterprise with Avaya IP Office 11.0, 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.

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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. **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.

5. **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.

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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 IP Office Web Client 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 J129 SIP set, as well as Avaya Equinox™ for Windows (SIP) and Avaya IP Office Web Client (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 "LAN1" is connected to the private CPE, and interface "LAN2" 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 LAN2 (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 10 digits.

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Figure 1: Reference Configuration

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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 LAN2 interface is defined as the SIP trunk (see **Section 5.3.2**) 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 LAN2 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

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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

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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)

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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	11.0.0.1.0 Build 8
Avaya IP Office Application Server ▪ Voicemail Pro ▪ Avaya WebRTC Gateway ▪ Avaya one-X® Portal for IP Office	11.0.0.1.0 Build 8 11.0.0.1.0 Build 3 11.0.0.0.1 Build 54 11.0.0.1.0 Build 38
Avaya Equinox™ for Windows (SIP)	3.4.0.152
Avaya 9641G (H.323) IP Deskphone	6.6506
Avaya 1616 (H.323) Telephone	Ha1616ua1_3100A.bin
Avaya J129 IP Telephone (SIP)	3.0.0.0.20
Avaya 9508 Digital Telephone	0.60
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.

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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.

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**.

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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, **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).

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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. LAN1 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 LAN1 – LAN Settings Tab

To view or configure the LAN1 IP address, select the **LAN1 → LAN Settings** tab, and enter the following:

- **IP Address:** Set to **10.64.19.70** as specified in the reference configuration.

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5.3.1.2 LAN1 - 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 J129 (SIP) and Avaya Equinox™ 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.

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5.3.2. LAN2 Tab

The LAN2 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 LAN2 - 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.

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5.3.2.2 LAN2 - 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.

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- 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**
    - **Periodic Timeout**: Enter **30**
    - **Initial keepalives**: Select **Enabled**
- Other parameters on this screen are set to the defaults.

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5.3.2.3 LAN2 - 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** 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**.

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

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- 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 5.5.1**) 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.

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5.3.5. VoIP Security

For the compliance test, SRTP was used internal to the enterprise wherever possible. To view or configure the media encryption settings, select the **VoIP Security** tab. Set the **Media** drop-down menu to **Preferred** to have IP Office attempt use encrypted RTP for devices that support it and fall back to RTP for devices that do not support encryption. Under **Media Security Options**, select **RTP** for the **Encryptions** and **Authentication** fields. Under **Crypto Suites**, select **SRTP_AES_CM_128_SHA1_80**. Click **OK** to commit (not shown).

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:

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- 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 **IPO11_0TF.xml** was imported. The template files are automatically copied into the IP Office default template location, **C:\Program Files\Avaya\IP Office\Manager\Templates**.

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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**.

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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 22**, 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 **Reinvite**, as AT&T does not support UPDATE.
- **Timer (seconds):** Set to **1800** during testing.
- **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).

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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).

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5.4.5. SIP Line - Call Details tab

Select the **SIP Line → Call Details** tab.

To add a new SIP URI, click the **Add…** button. A **New URI** window will open. Configure the following:

- **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**.
- **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.
- **Outgoing Group:** Set to an unused group number, e.g., **22**.
- **Local URI** and **Contact** fields for the **Display** and **Content** columns: Set these fields to **Auto**.
- **Field meaning** section: Set to the values shown in the screenshot below.
- Click **OK**.

- 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).

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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 with **G729(a) 8K CS-ACELP** and **G.711 ULAW 64K** 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).

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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**).

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).

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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**.

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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. SIP User

The following screen shows the **User** tab for user 326. As shown in **Figure 1**, this user corresponds to the Avaya J129 SIP endpoint.

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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.

The following screen shows the **VoIP** tab for the extension. The **IP Address** field may be left blank. Check the **Reserve Avaya IP endpoint license** box. The **Codec Selection** parameter may retain the default setting "**System Default**" to follow the system configuration shown in **Section 5.3.4**. The Media Security parameter may also retain the default setting "**Same as System (Preferred)**" to follow the system configuring shown in **Section 5.3.5**.

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5.5.2. User 329

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.

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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 5.3.4**. 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.

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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 one-X Portal Services** checked.

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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 (not shown) to change the **User List**.

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

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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 **0000011041** 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., **0000011041**).
   - Use default values for the remaining fields and click **OK** (not shown).

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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 **0000051045** being directed to Hunt Group **401** (Call Center).

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5.6.2. Calls to Voicemail Pro Scripts

As described in **Section 5.6.1** and **5.7**, 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 **0000021042** 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., **0000021042**).
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).

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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 transfer the call to an associated Avaya IP Office Hunt Group (Call Center, and Support), or to a specific extension. This is accomplished via the following steps:

1. Hunt Groups **Call Center** and **Support** are created in IP Office (**Section 5.5.3**).
2. User 329 is created in IP Office (**Section 5.5.2**).
3. Incoming Call Route for DNIS digits **0000021042** 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.

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- 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**.

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- 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 ("**Ext329**").
        1. On the **General** tab, **Token Name** = **Transfer to 329** (not shown).
        2. On the **Specific** tab, **Destination = 329** (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).

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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., **0000021042**), 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 329. The associated Avaya IP Office extension (e.g., 401, 402, or 329) will then ring.

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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.

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7. Verification Steps

The following procedures may be used to verify the Avaya IP Office Release 11.0 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 Release 11.0 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 11.0

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**.

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After logging in, select **Trunks → Line: 22** 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**.

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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 **Trace Options** button or select **Filters → Trace Options**.

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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 Release 11.0 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.

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9. References

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

[1] *Deploying IP Office™ Platform Server Edition Solution*, Release 11.0, May 2018
[2] *IP Office™ Platform 11.0, Deploying Avaya IP Office™ Platform IP500 V2*, May 2018
[3] *Administering Avaya IP Office™ Platform with Web Manager*, Release 11.0, May 2018.
[4] *Administering Avaya IP Office™ Platform with Manager*, Release 11.0, May 2018.

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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