Avaya Solution & Interoperability Test Lab

Application Notes for Avaya IP Office Release 9.1 with AT&T IP Flexible Reach - Enhanced Features – Issue 1.1

Abstract

These Application Notes describe the steps for configuring Avaya IP Office R9.1 with the AT&T IP Flexible Reach - Enhanced Features service using AVPN or MIS/PNT transport connections.

The AT&T IP Flexible Reach - Enhanced Features service is one of several SIP-based Voice over IP (VoIP) services offered to enterprises for a variety of voice communications needs. The AT&T IP Flexible Reach service allows enterprises in the U.S.A. to place outbound local and long distance calls, receive inbound Direct Inward Dialing (DID) calls from the PSTN, and place calls between an enterprise's sites.

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.

Table of Contents

1. Introduction

These Application Notes describe the steps for configuring Avaya IP Office R9.1 with the AT&T IP Flexible Reach - Enhanced Features 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 Flexible Reach - Enhanced Features service is one of several SIP-based Voice over IP (VoIP) services offered to enterprises for a variety of voice communications needs. The AT&T IP Flexible Reach - Enhanced Features service allows enterprises in the U.S.A. to place outbound local and long distance calls, receive inbound Direct Inward Dialing (DID) calls from the PSTN, and place calls between an enterprise's sites. The AT&T IP Flexible Reach - Enhanced Features service utilizes AVPN[1] or MIS/PNT[2] transport services.

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

Note – The AT&T IP Flexible Reach - Enhanced Features service will be referred to as IPFR-EF in the remainder of this document.

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 IPFR-EF and the Customer Premises Equipment (CPE) containing the Avaya IP Office 9.1 (see **Section 3.2** for call flow examples).

The test environment described in these Application Notes consisted of:

- A simulated enterprise with Avaya IP Office 9.1, Avaya SIP, H.323 and Analog telephones, as well as a fax machine emulator (Ventafax).
- An IPFR-EF production circuit, to which the simulated enterprise was connected via AVPN transport.

[1] AVPN uses compressed RTP (cRTP).
[2].MIS/PNT does not support cRTP.

2.1. Interoperability Compliance Testing

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

The following SIP trunking VoIP features were tested with the IPFR-EF service:

- Incoming and outgoing voice calls between PSTN, the IPFR-EF service, and Avaya IP Office, utilizing Avaya SIP telephones (desk and softphone), H.323 telephones (desk).
- Inbound/Outbound fax calls using T.38 or G.711U.
- Various outbound PSTN destinations including local, long distance, international, and toll-free.
- Requests for privacy (i.e., caller anonymity) for Avaya IP Office outbound calls to the PSTN, as well as privacy requests for inbound calls from the PSTN to Avaya IP Office users.
- SIP OPTIONS messages used to monitor the health of the SIP trunk from both Avaya IP Office and AT&T.
- Incoming and outgoing calls using the G.729(A & B) and G.711 ULAW codecs.
- Call redirection with Diversion Header.
- Long duration calls.
- DTMF transmission (RFC 2833) for successful PSTN and Avaya IP Office menu navigation.
- Telephony features such as hold, transfer, and conference.
- Avaya IP Office Mobile Twinning to a mobile phone when the associated Avaya IP Office extension is called, as well as Mobility features such as Mobile Callback and Mobile Call Control.
- AT&T IPFR-EF service features such as:
    - Simultaneous Ring
    - Sequential Ring
    - Call Forward – Always
    - Call Forward – Busy
    - Call Forward – Ring No Answer
    - Blind and Attended transfers utilizing SIP REFER messaging.

2.2. Test Results

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

1. **Avaya IP Office SIP endpoints use different RFC2833 Telephone Event types than defined in Avaya IP Office provisioning –** Although Avaya IP Office can specify the RFC2833 Telephone Event to use for Analog/Digital and H.323 sets, (see **Section 5.3.7**), it was found that the 1120E SIP desk phone, and the Avaya Communicator SIP softphone, use Telephone Events 101 and 120 respectively.
    - No issues were found during testing as a result of this behavior.

2. **Avaya Communicator SIP softphone does not change the RFC2833 telephone event type, even though a successful telephone event type change has been negotiated** – While performing an AT&T IPFR-EF Simultaneous Ring feature call, it was found that if the Avaya Communicator SIP softphone was defined as the secondary number, and it answered the inbound call, the Avaya Communicator SIP softphone could not generate the confirmation DTMF digit. As a result the call could not be answered by the Avaya Communicator SIP softphone.
    - The AT&T network plays an announcement asking for a confirmation digit. When doing so, the network ReInvites, requesting that the RFC2833 telephone event type be changed from the value specified in the original Invite. The Avaya Communicator SIP softphone accepts this change, but when the DTMF digit is issued, the Avaya Communicator SIP softphone still uses the telephone event type specified in the original Invite.
    - An MR was opened with Avaya IP Office support.
    - As a result the Avaya Communicator SIP softphone should not be used as an AT&T IPFR-EF Simultaneous Ring secondary destination.

3. **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.

4. **Avaya IP Office Direct Media feature cannot be used in the reference configuration** – If the Direct Media feature is enabled, Avaya IP Office will send AT&T the IP address of the calling party (in a Re-Invite) or called party (in the initial 200OK) Avaya IP Office IP H.323 or SIP telephone. As this address would be a private IP address (see the **Section 3**), the resulting media would be unroutable between Avaya IP Office and AT&T.
    - Direct Media cannot be selected when T.38 fax is specified. However it could be specified if G.711 fax is used. Therefore the Avaya IP Office Direct Media feature should not be used in the reference configuration (see **Section 5.4.6**).

5. **Inbound T.38 or G.711 fax calls fail when the sender and receiver are both Super G3 (SG3) fax devices** – During testing it was found that when the sender and receiver both used SG3 fax devices, and an inbound fax call was placed to Avaya IP Office using either T.38 or G.711, approximately 80% of the fax calls failed to connect.
    - It was found that during SG3/SG3 inbound fax calls, Avaya IP Office took between 15 and 20 seconds to establish the fax connection (non SG3/SG3 calls took half this time).
    - An MR was opened with Avaya IP Office support.
    - UPDATE – This issue is resolved in Avaya IP Office 9.1, Service Pack 1.

6. **Avaya IP Office issues SIP Invites with incorrect Host field contents in the From Header** – If the SIP Line *ITSP Domain Name* field (see **Section 5.4.3**), is populated per the system Help file (e.g., domain or IP address of the Service Provider), then Avaya IP Office will populate outbound Invite From headers with this value. Instead, the From header should be populated with the IP address of the Avaya IP Office SIP Trunk interface (LAN 2 in the reference configuration).
    - An MR was opened with Avaya IP Office support.
    - A workaround is to populate the *ITSP Domain Name* field with the LAN 2 IP address.
        - Note that this workaround will cause Avaya IP Office to send OPTIONS messages to AT&T with the LAN 2 IP address in the R-URI and To headers. However as the OPTIONS messages are used for trunk "keep-alive" purposes only, this is not an issue.

7. **Emergency 911/E911 Services Limitations and Restrictions** – Although AT&T provides 911/E911 calling capabilities, AT&T does not warrant or represent that the equipment and software (e.g., IP PBX) documented in these Application Notes will properly operate with AT&T IP Flexible Reach to complete 911/E911 calls; therefore, it is the customer's responsibility to ensure proper operation with the equipment/software vendor.
   While AT&T IP Flexible Reach services support E911/911 calling capabilities under certain Calling Plans, there are circumstances when the E911/911 service may not be available, as stated in the Service Guide for AT&T IP Flexible Reach found at http://new.serviceguide.att.com. Such circumstances include, but are not limited to, relocation of the end user's CPE, use of a non-native or virtual telephone number, failure in the broadband connection, loss of electrical power, and delays that may occur in updating the Customer's location in the automatic location information database. Please review the AT&T IP Flexible Reach Service Guide in detail to understand the limitations and restrictions**.**

2.3. Support

For more information on the AT&T IP Flexible Reach service visit:
http://www.business.att.com/enterprise/Service/voice-services/null/sip-trunking/
AT&T customers may obtain support for the AT&T IP Flexible Reach service by calling (877) 288-8362.

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.

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** below 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..
- Avaya “desk” telephones are represented with an Avaya 1608 H.323 set, an Avaya 6211 Analog set, an Avaya 1120E SIP set, as well as Avaya Communicator 2.0 (SIP). Fax endpoints are represented by PCs running Ventafax emulation software connected by modem to an Avaya IP Office analog port.
- Avaya IP Office embedded voicemail provided the voice messaging capabilities in the reference configuration.
- In the reference configuration, Avaya IP Office interface “LAN 1” (labeled LAN) is connected to the private CPE, and interface “LAN 2” (labeled WAN) is connected to the public network and AT&T.
- TCP transport via port 5060, was used on the Avaya IP Office LAN1 connection to the CPE.
- The AT&T IPFR-EF service requires the following SIP trunk network settings between the Avaya IP Office LAN 2 (SIP Trunk) interface and the IPFR-EF 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 IPFR-EF service may deliver 10 or 7 digits in the SIP Invite R-URI depending on the circuit order provisioning. In the reference configuration the IPFR-EF service delivered 7 digits.

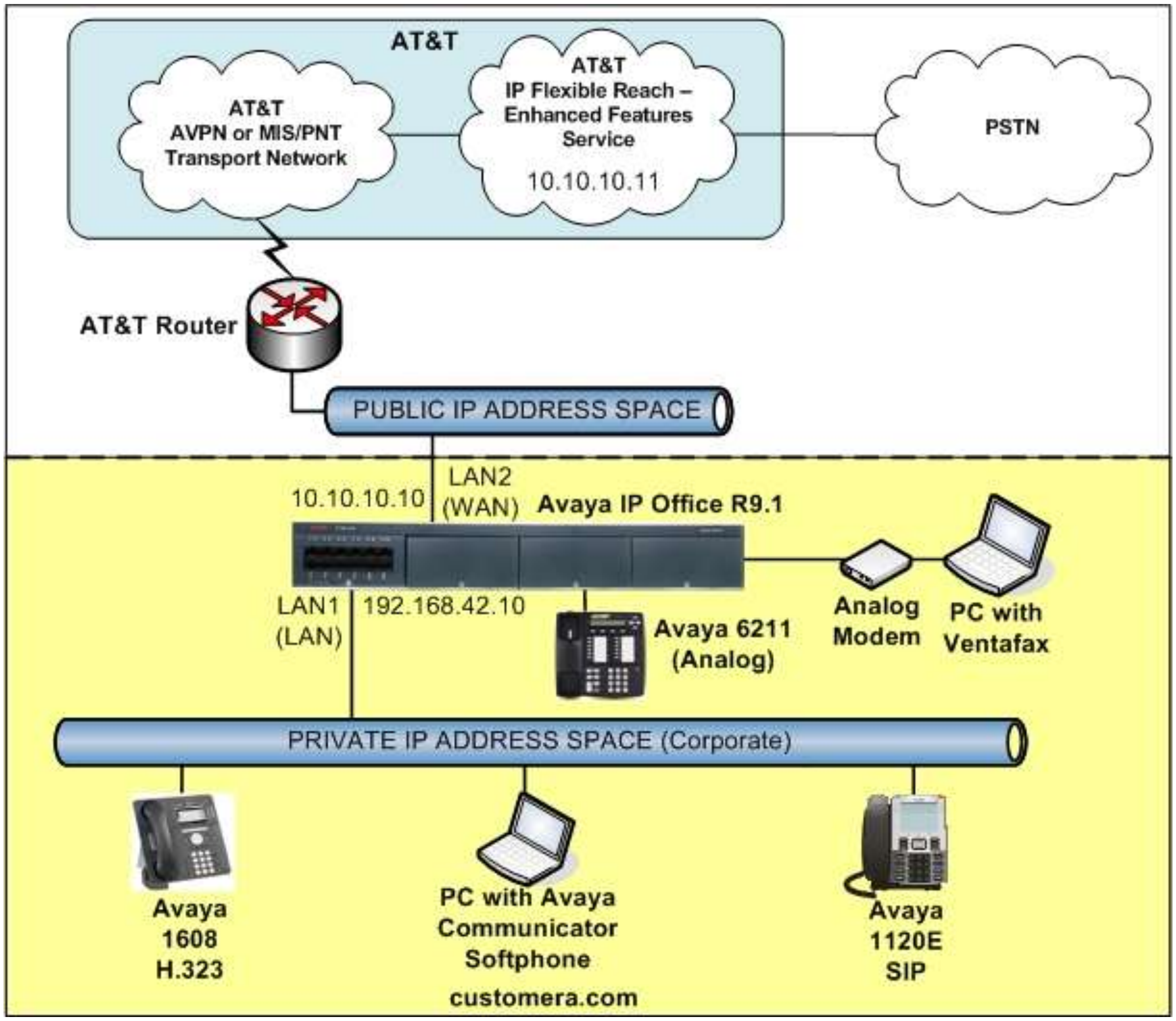


Figure 1: Reference Configuration

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 IPFR-EF 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 **10.10.10.10** (Avaya IP Office LAN 2 address), and **10.10.10.11/10.10.10.12** (AT&T BE IP addresses), 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 IPFR-EF provisioning process.

Component	Illustrative Value in these Application Notes
Avaya IP Office 500 V2 Platform	
Private network LAN1 interface, (labeled “LAN” on the chassis)	192.168.42.10
LAN2 interface, (labeled “WAN” on the chassis), for management access.	10.10.10.10
AT&T IPFR-EF Service	
Border Element IP Address	10.10.10.11 & 10.10.10.12

Table 1: Illustrative Values Used in these Application Notes

3.2. Call Flows

To understand how inbound and outbound AT&T IPFR-EF service calls are handled by Avaya IP Office, four 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 IPFR-EF 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 IPFR-EF service number.
2. The PSTN routes the call to the AT&T IPFR-EF service network.
3. The AT&T IPFR-EF 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

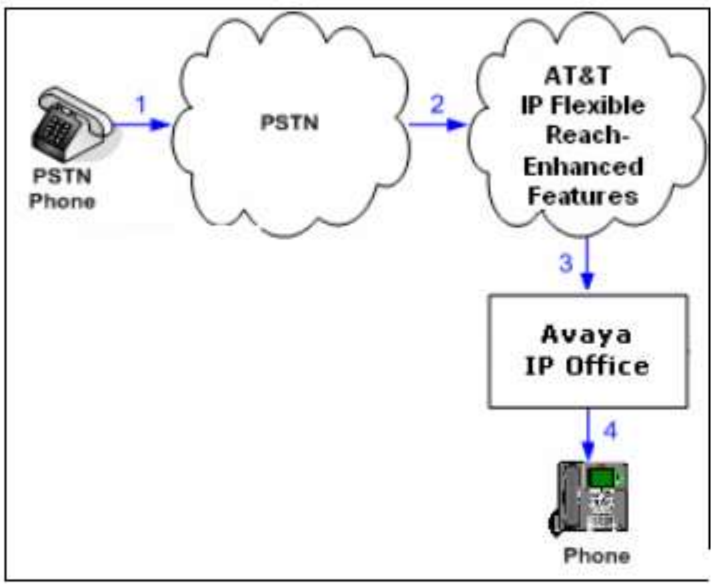


Figure 2: Inbound AT&T IPFR-EF Call

3.2.2. Outbound

The second call scenario illustrated in the figure below is an outbound call initiated on Avaya IP Office for delivery to AT&T IPFR-EF service.

1. An Avaya IP Office phone or fax endpoint originates a call to an AT&T IPFR-EF service number for delivery to PSTN.
2. Avaya IP Office applies any necessary origination treatment (verifying permissions, determining the proper route, selecting the outgoing trunk, etc.) and sends the call to the AT&T IPFR-EF service.
3. The AT&T IPFR-EF service delivers the call to PSTN.
4. PSTN delivers the call to a phone or fax endpoint.

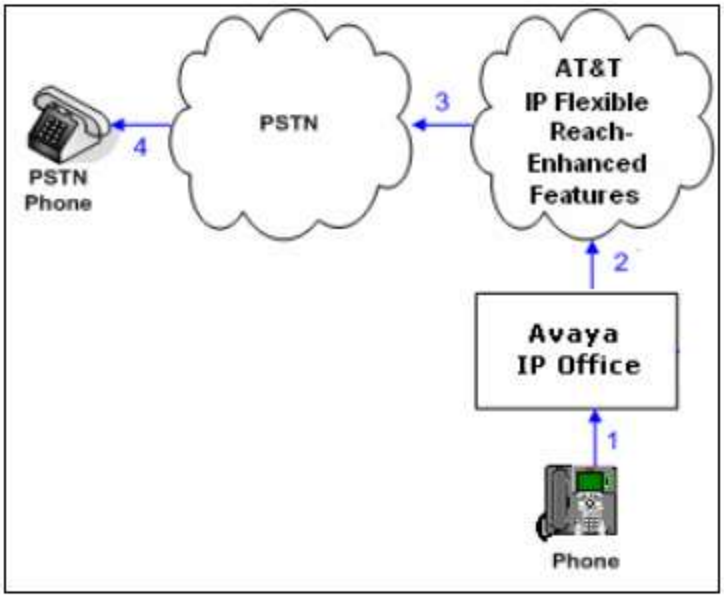


Figure 3: Outbound Call to AT&T IPFR-EF

3.2.3. Call Forward

The third call scenario illustrated in the figure below is an inbound AT&T IPFR-EF service call destined for an Avaya IP Office station that has set Call Forwarding to an alternate destination. Without answering the call, Avaya IP Office redirects the call back to the AT&T IPFR-EF service for routing to the alternate destination.

Note – AT&T requires the Diversion header be used when a call is redirected to AT&T IPFR-EF service (See **Section 5.4.3**, and **Section 2.2**.

1. Same as the first call scenario in **Section 3.2.1**.
2. The Avaya IP Office phone has set Call Forward to another AT&T IPFR-EF service number, therefore, Avaya IP Office initiates a new call back out to the AT&T IPFR-EF service network. This new SIP INVITE will contain a Diversion Header.
3. The AT&T IPFR-EF service places a call to the alternate destination.

Note – The IPFR-EF service offers similar Call Forwarding features that allow users to predefine alternate call destinations based on Ring-No-Answer, Busy, Not Reachable, or Unconditional criteria.

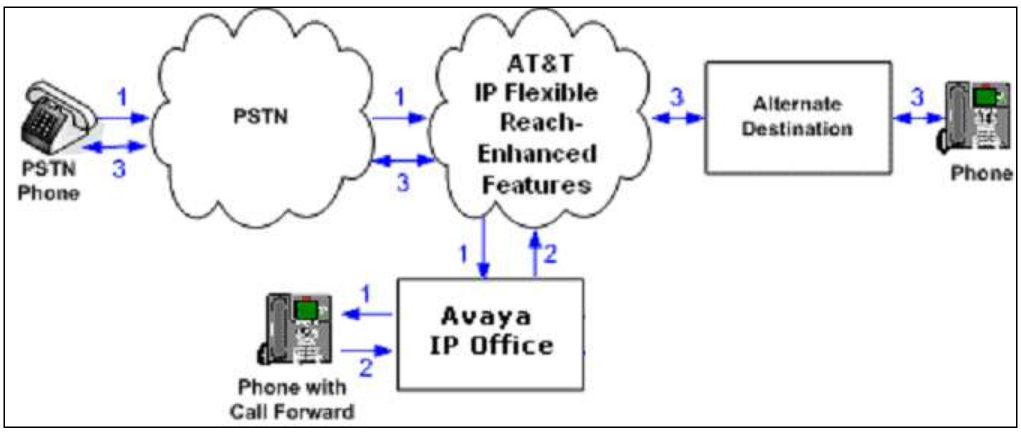


Figure 4: Call Forward

3.2.4. 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 the embedded Avaya IP Office Voicemail.

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 the embedded Avaya IP Office Voicemail.

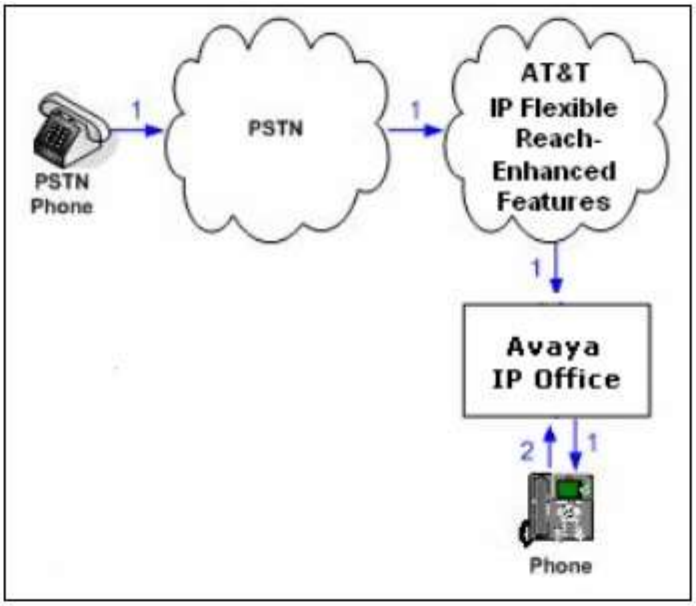


Figure 5: Coverage to Avaya IP Office Voicemail

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	R9.1 (437) (see Section 2.2, Item 5)
Avaya 1608 (H.323) Telephone	Ha1608ua1_350B.bin
Avaya 1120E (SIP) Telephone	04.04.10.00
Avaya Communicator for Windows	2.0.3.30
Avaya 6211 Analog Telephone	-
Fax device	Ventafax 6.3

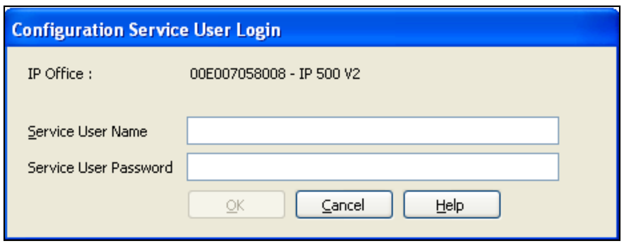
Table 2: Equipment and Software Versions

Note - Compliance Testing is applicable when the tested solution is deployed with a standalone IP Office 500 V2 and also when deployed with all configurations of IP Office Server Edition without T.38 Fax Service...

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.

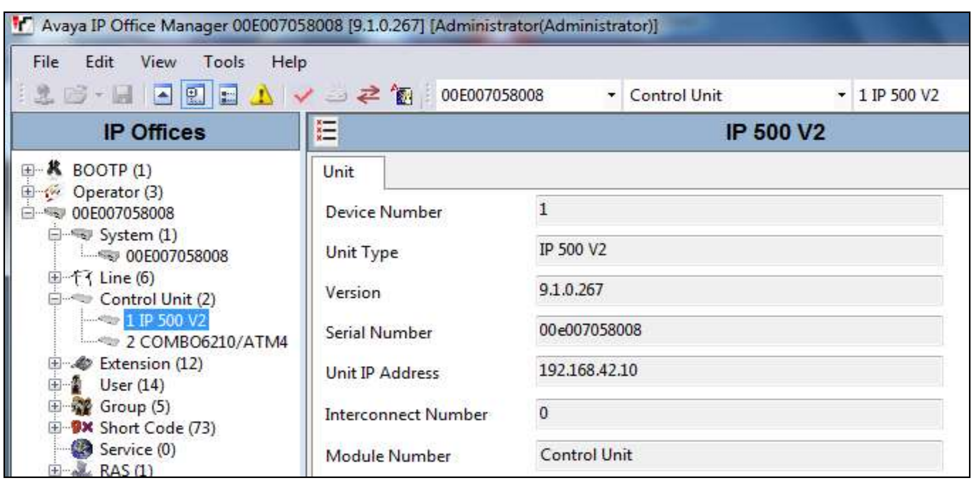
Avaya IP Office is configured via the Avaya IP Office Manager program. For more information on provisioning Avaya IP Office Manager, consult reference **[1]**. From the Avaya IP Office Manager PC, select **Start** → **Programs** → **Avaya IP Office** → **Manager** to launch the Manager application. Enter the appropriate credentials.



5.1. Platform Information

Note - In the following sections, the left hand Navigation pane will be used to select Avaya IP Office provisioning options.

This section describes attributes of the reference configuration. The following screen shows the Avaya IP Office module configuration used in the reference configuration. In the screen below, the **IP 500 V2** platform is displayed along with the COMBO6210/ATM4 module.



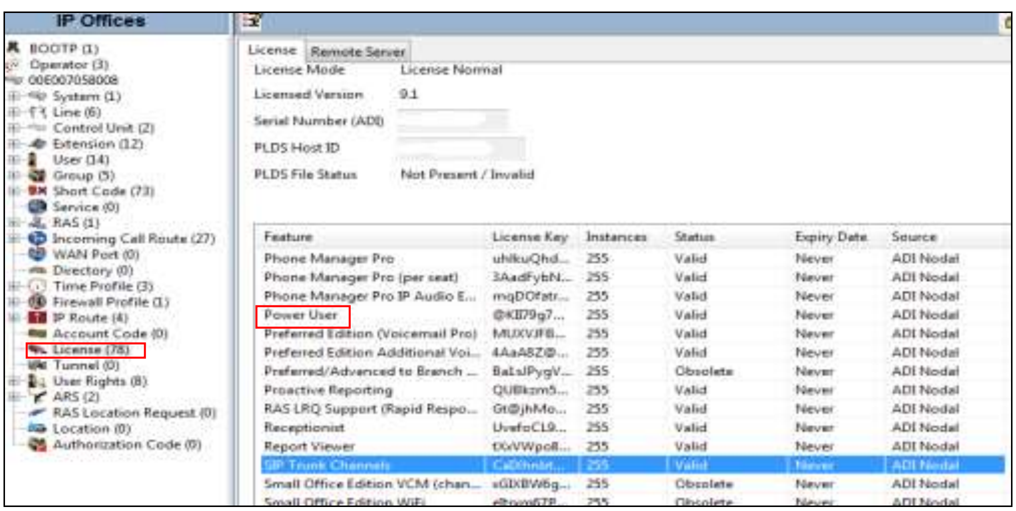
The Avaya IP Office 500 V2 has two Ethernet ports on the back of the chassis, labeled **WAN** and **LAN**. In the reference configuration, the LAN port (LAN1) is connected to the private CPE network. The Avaya H.323 and SIP telephones, and the Avaya IP Office management/Softphone PC, are also connected to the private CPE network. The WAN port (LAN2) is the SIP trunk connected to the public network, accessing the AT&T network. Provisioning for these interfaces is described in **Section 5.3.2** and **5.3.3**.

In order for the Avaya IP Office system to be able to route data to/from the AT&T network, a default route must be added. To add an IP Route in Avaya IP Office, right-click **IP Route** from the left hand 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 relevant default route using **Destination → LAN2** (to AT&T).

5.2. Licensing

The configuration and features described in these Application Notes require the Avaya 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 and verify that **SIP Trunk Channels** has sufficient "Instances" (trunk channels). If any of those endpoints are to be defined as a **Power User**, then that must be licensed as well.



5.3. System Settings

This section illustrates the configuration of system settings. Select **System** in the left hand Navigation pane to configure these settings.

5.3.1. System Tab

With the proper system name selected in the Group pane, select the **System** tab in the Details pane. The following screen shows a section of the **System** tab. The **Name** field can be used for a descriptive name of the system.

5.3.2. LAN 1 Tab

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

5.3.2.1 LAN 1 – LAN Settings Tab

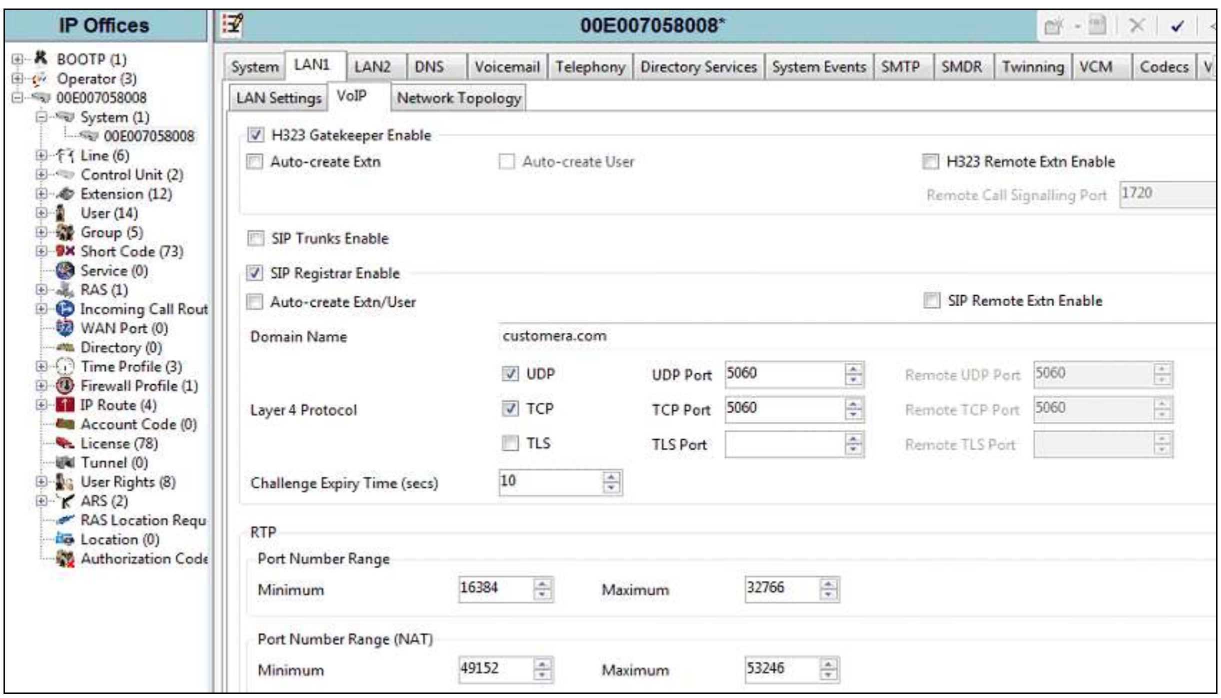
To view or configure the LAN 1 IP address, select the **LAN 1 → LAN Settings** tab, and enter the following:

- **IP Address:** Set to **192.168.42.10** as used in the reference configuration.
- **DHCP Mode** is also set to **Server** so that IP phones will get an IP Address from the Avaya IP Office Server. Other parameters on this screen may be set according to customer requirements.
- Click the **OK** button (not shown).

5.3.2.2 LAN 1 - VoIP Tab

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

- The **H323 Gatekeeper Enable** box is checked to allow the use of Avaya IP Telephones using the H.323 protocol, such as the Avaya 1600-Series Telephones used in the reference configuration.
- The **SIP Registrar Enable** box is checked to allow Avaya 11xx (SIP) and Avaya IP Office Softphone (SIP) usage.
- The **Domain Name** used in the reference configuration is **customera.com**.
- In the **Layer 4 Protocol** section, select **UDP/5060** and **TCP/5060**.
- **RTP Port Number Range:** The AT&T IPTF service requires that the RTP use the port range 16384 to 32767.
    - o **16384** entered in the **Port Range (Minimum)** field.
    - o **32766** entered in the **Port Range (Maximum)** field, as this field requires even numbers.
- Let all other values default.
- Click the **OK** button (not shown).

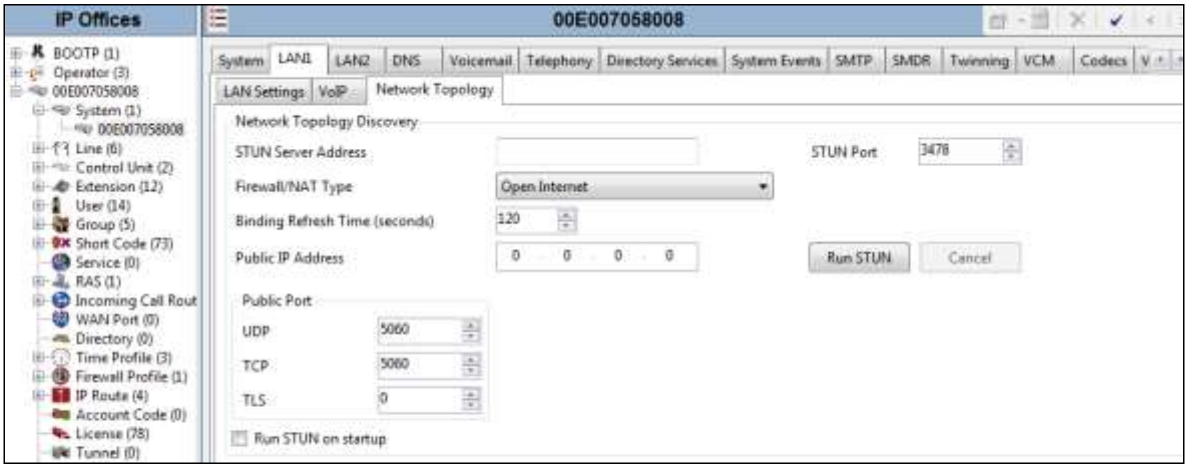


5.3.2.3 LAN 1 - Network Topology Tab

Select the **LAN1 → Network Topology** tab as shown in the following screen, and enter the following:

- **Public IP Address**: The **0.0.0.0** default value is used. This means Avaya IP Office will use the LAN1 IP address specified on the LAN1 **LAN Settings** tab described above (192.168.42.10).

- **Public Port:** Enter **UDP/5060** and **TCP/5060.**
- **Firewall/NAT Type** is set to **Open Internet** in the reference configuration. Note that the **Firewall/NAT Type** parameter may need to be set differently, depending on the type of firewall or Network Address Translation device used at the customer premise.
- Click the **OK** button (not shown).

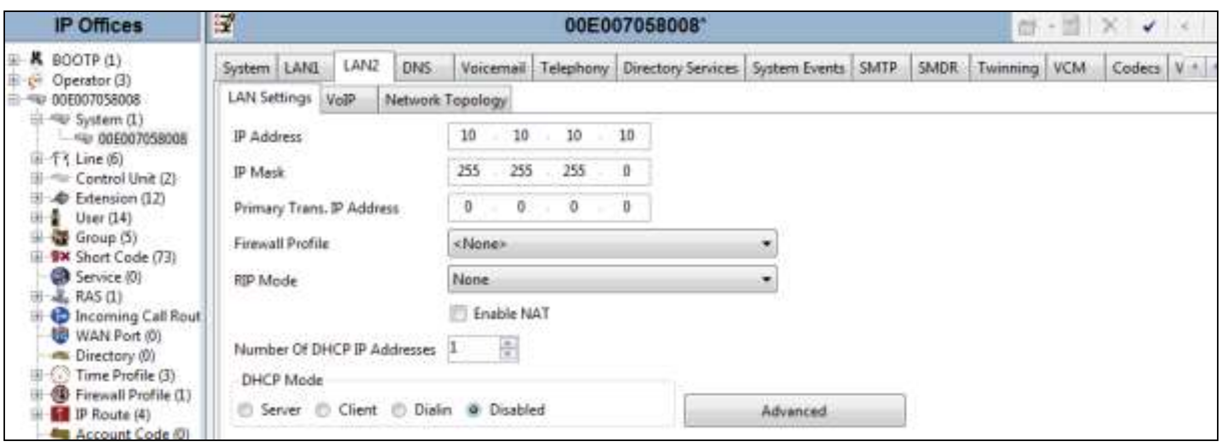


5.3.3. LAN 2 Tab

The LAN 2 interface is used for the SIP trunk connection to AT&T.

5.3.3.1 LAN 2 - LAN Settings Tab

- **IP Address:** In the reference configuration the IP Office public address is **10.10.10.10**.
- Other parameters on this screen are set to the defaults.

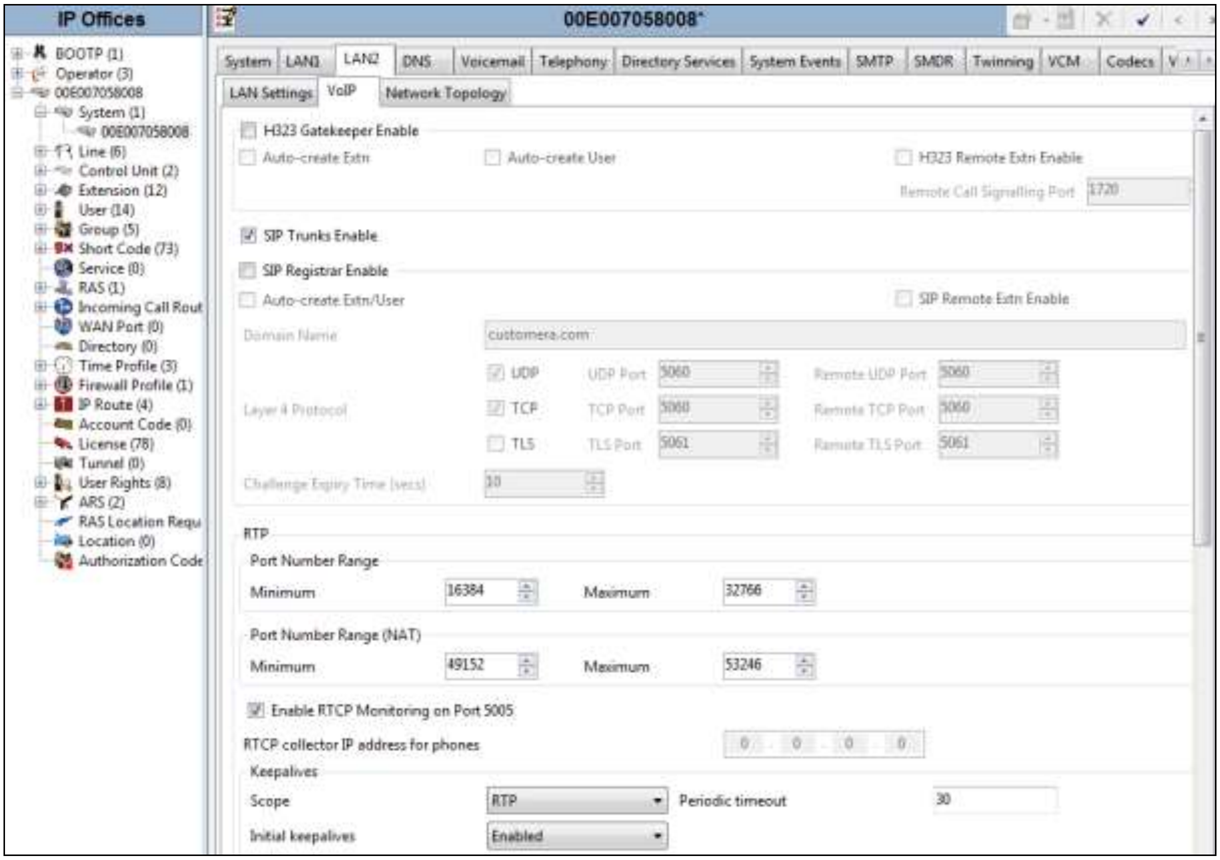


5.3.3.2 LAN 2 - VoIP Tab

- Select the **SIP Trunks Enabled** option.
- 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:

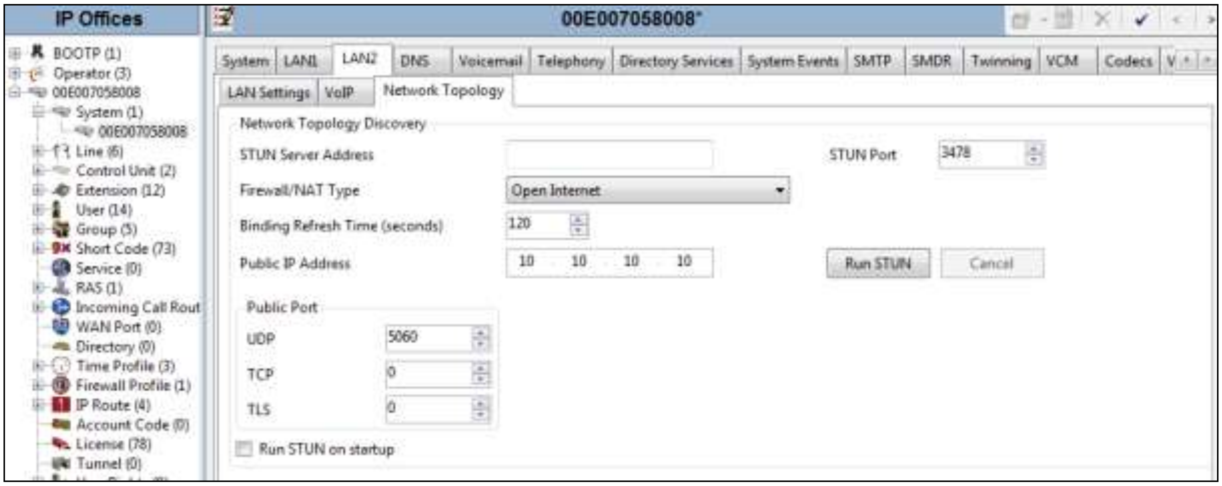
- **Scope**: Select **RTP**
- **Periodic Timeout**: Enter **30**
- **Initial keepalives**: Select **Enabled**

- Other parameters on this screen are set to the defaults.
- Click the **OK** button (not shown).



5.3.3.3 LAN 2 - Network Topology Tab

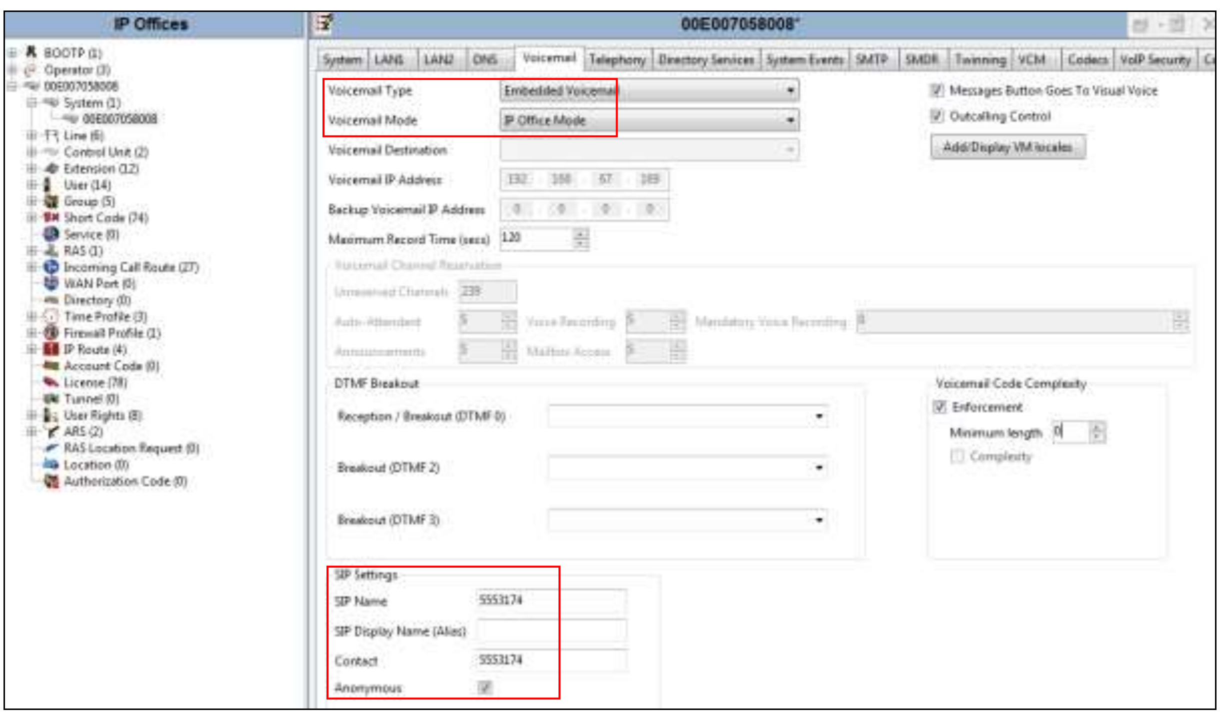
- Set the **Firewall/NAT Type** is set to **Open Internet** in the reference configuration. Note that the **Firewall/NAT Type** parameter may need to be set differently, depending on the type of firewall or Network Address Translation device used at the customer premise.
- **Binding Refresh Time:** This 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 (see **Section 2.2**).
- **Public IP Address**: In the reference configuration the IP Office public address is **10.10.10.10**.
- Set the **Public Port** to **UDP/5060**.
- Click the **OK** button (not shown).



5.3.4. Voicemail Tab

As described in **Section 3**, the embedded Avaya IP Office Voicemail system was used in the reference configuration.

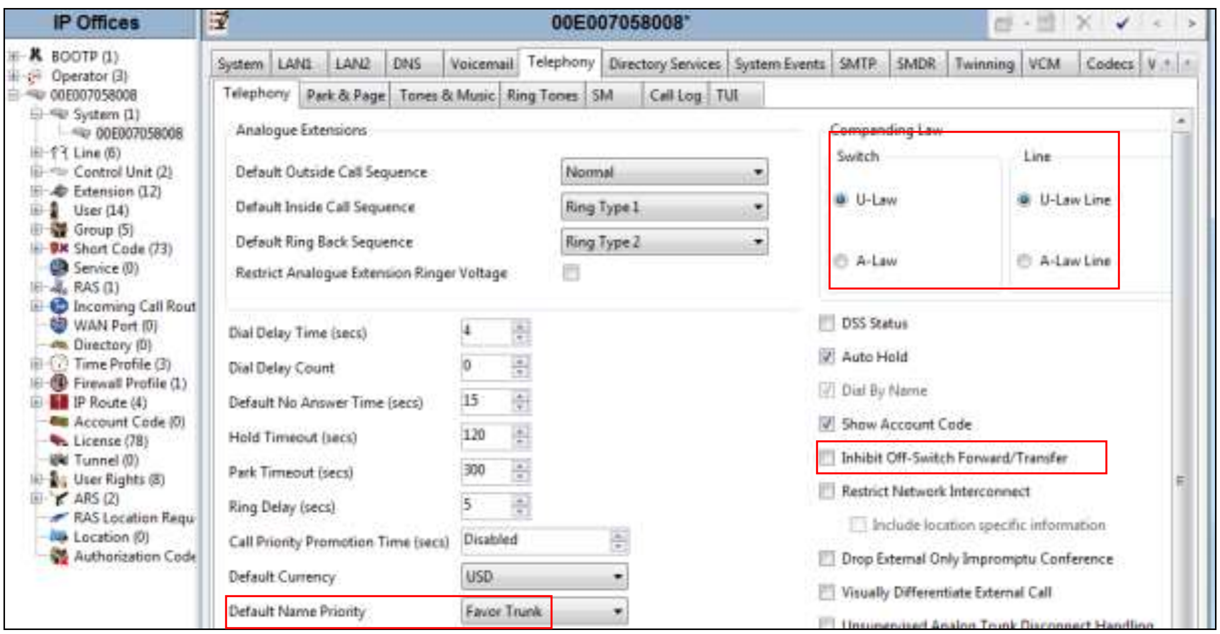
- Set **Voicemail Type** to **Embedded Voicemail**.
- Set **Voicemail Mode** to **IP Office Mode**.
- In the **SIP Settings** section, set the **SIP Name** and **Contact** fields to the AT&T DNIS digits used to call directly to Voicemail (e.g., **5553174** for message retrieval, see **Sections 5.6.4.1** and **5.7**). Note that the **Anonymous** box is checked by default, so no entry is needed in the **SIP Display Name (Alias)** field.
- Other parameters on this screen are default. Click the **OK** button (not shown).



5.3.5. Telephony Tab

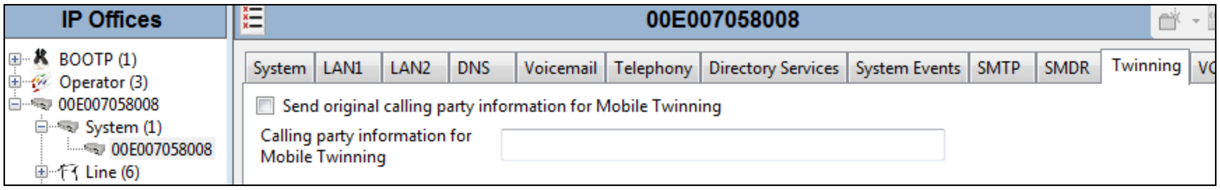
To view or change telephony settings, select the **Telephony** tab and **Telephony** sub-tab as shown in the following screen. The settings presented here simply illustrate the values used in the reference configuration and are not intended to be prescriptive.

- *Uncheck* the **Inhibit Off-Switch Forward/Transfer** box. This is so that call forwarding and call transfer to PSTN destinations via the AT&T IPFR-EF service can be tested.
- Set the **Companding Law** parameters are set to **U-LAW** as is typical in North America.
- In the reference configuration, **Default Name Priority** is set to **Favor Trunk**. With the option set to **Favor Directory**, Avaya IP Office will prefer to display names found in a personal or system directory over those arriving from the far-end, if there is a directory match to the caller ID. This capability is also defined in the **SIP Line** tab in **Section 5.4.3**. A user's personal directory example is shown in **Section 5.5.2**.
- Default values are used in the other fields.
- Click the **OK** button (not shown).



5.3.6. Twinning Tab

In the reference configuration, a PSTN number was defined. On this form, default values were used. With this configuration, and related configuration of Diversion Header on the SIP Line (**Section 5.4.3**), the true identity of a PSTN caller can be presented to the twinning destination (e.g., a user's mobile phone) when a call is twinned out via the AT&T IPFR-EF service.



5.3.7. Codecs 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 line or extension.

- Populate the **Selected** column with **G.711 ULAW 64K** as the first codec and **G.729(a) 8K CS-ACELP** as the second codec.
- In the **RFC2833 Default Payload** setting field, specify **100**, which is the recommended value for AT&T (see **Section 2.2**).
- Click the **OK** button (not shown).

Note - In the reference configuration, the Extension codec lists (see **Section 5.5.2**) also specify G.711mu and G.729(a) (in that order), and the SIP Line (see **Section 5.4.6)** offers G.729(a) and G.711U (in that order). In this manner local Avaya IP Office calls will offer G.711U first, and SIP trunk calls will offer G.729 first.
However, the AT&T IPFR-EF service uses G.729 with Silence Suppression (G.729B) by default. See **Sections 5.4.6** and **5.5.2** for methods for enabling Silence Suppression, so that G.729B will be offered to AT&T.

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 IPFR_EF service.

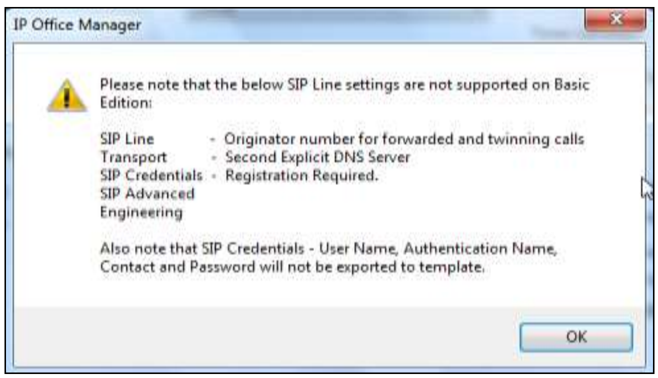
The recommended method for creating/configuring a SIP Line is to use the template associated with the provisioning described in these Application Notes. The template is an .xml file that can be used by Avaya IP Office Manager to create a new SIP Line for SIP trunking with the AT&T IPFR-EF service. Follow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



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**.

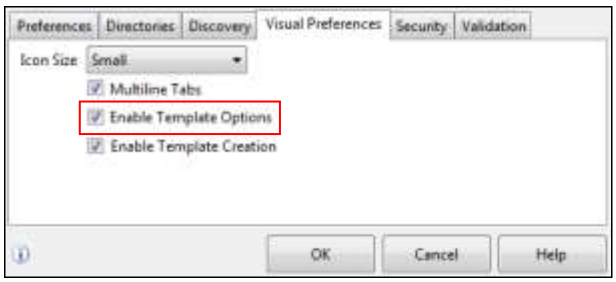
5.4.1. Importing a SIP Line Template

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 (500V2) 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.

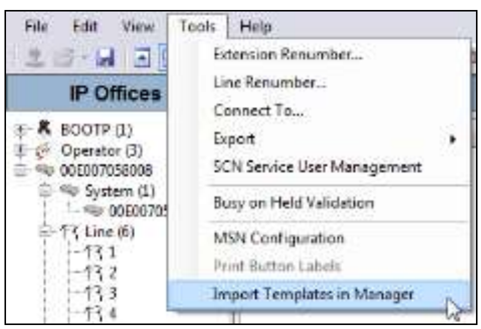
1. Copy a previously created template file to a location (e.g., *temp*) on the same computer where IP Office Manager is installed. By default, the template file name will have the format **AF_*<user supplied text>*_SIPTrunk.xml**, where the ***<user supplied text>*** portion is entered during template file creation.

Note – If necessary, the ***<user supplied text>*** portion of the template file name may be modified, however the **AF_*<user supplied text>*_SIPTrunk.xml** format of the file name must be maintained. For example, an original template file **AF_*TEST* _SIPTrunk.xml** could be changed to **AF_*Test1*_SIPTrunk.xml**. The template file name is selected in **Section 5.4.2** to create a new SIP Line.

2. Verify that Template Options are enabled in IP Office Manager. In IP Office Manager, navigate to **File → Preferences**. In the IP Office Manager Preferences window that appears, select the **Visual Preferences** tab. Check the box next to **Enable Template Options**. Click **OK**.

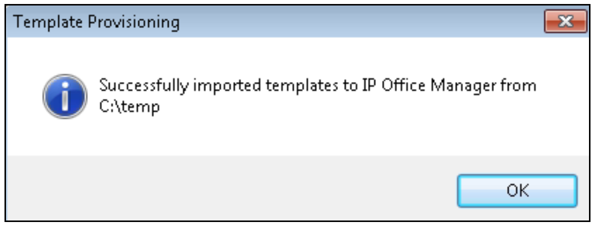


3. Import the template into IP Office Manager. From IP Office Manager, select **Tools → Import Templates in Manager**.

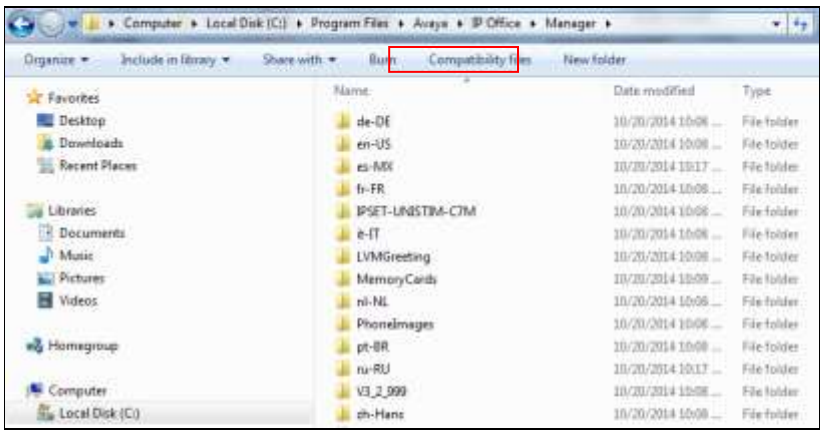


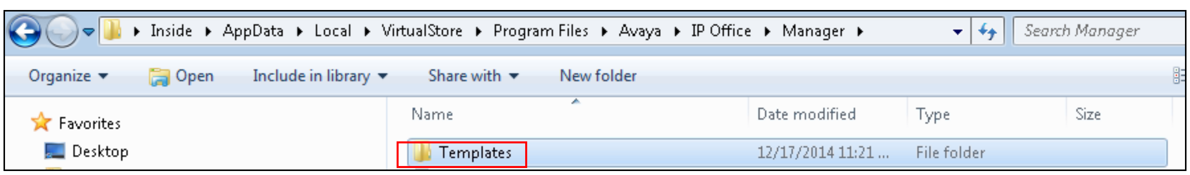
4. A folder browser will open (not shown). Select the directory used in **step 1** to store the template(s) (e.g., *temp*). In the reference configuration, template files **AF_no_SBCE_ SIPTrunk.xml** and **AF_with_SBCE_ SIPTrunk.xml** were imported. The template files are automatically copied into the IP Office default template location, **C:\Program Files\Avaya\IP Office\Manager\Templates**.

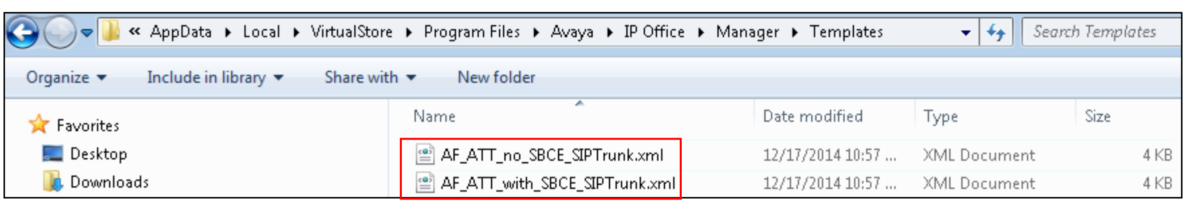
5. After the import is complete, a final import status pop-up window will open stating success or failure. Click **OK**.



Note –Windows 7 (and later) locks the Avaya IP Office 9.1 **\Templates** directory, and it cannot be viewed. To enable browsing of the **\Templates** directory, open Windows Explorer, navigate to **C:\Program Files\Avaya\IP Office\Manager** (or C:*Program Files (x86)*\Avaya\IP Office\Manager), and then click on the **Compatibility files** option shown below. The **\Templates** directory and its contents can then be viewed.

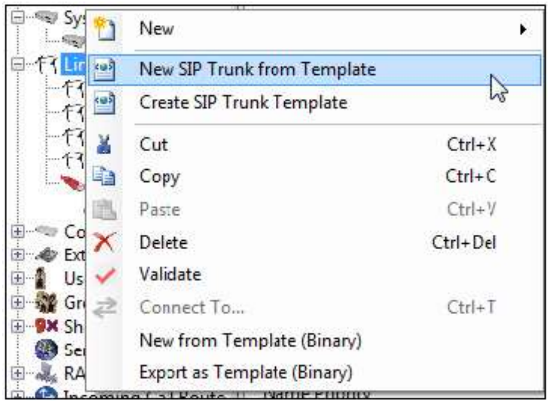






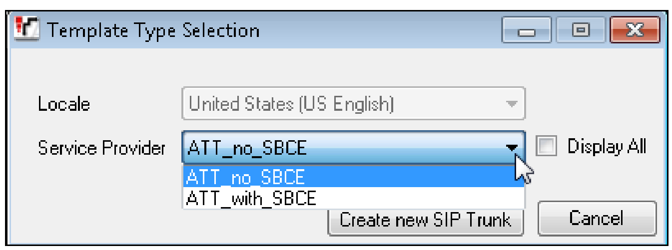
5.4.2. Creating a SIP Trunk from an XML Template

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

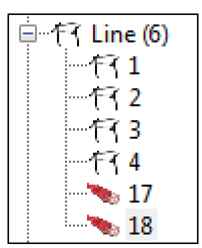


2. In the subsequent **Template Type Selection** pop-up window, from the **Service Provider** pull-down menu, select the XML template name from **Section 5.4.1**. Click **Create new SIP Trunk**.

Note – The drop down menu will display the ***<user supplied text>*** part of the template file name (see **Section 5.4.1**). If you check the **Display All** box, then the full template file name is displayed.



The newly created SIP Line will appear in the Navigation pane (e.g., SIP Line **18**).

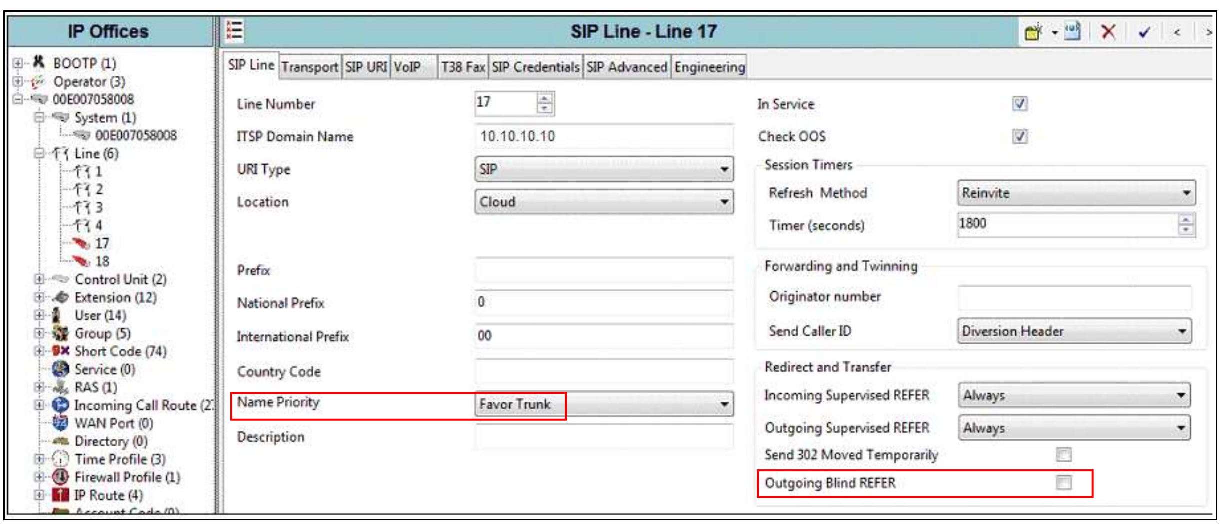


5.4.3. SIP Line – SIP Line tab

The **SIP Line** tab in the Details pane is shown below for **Line Number 17**, used for the SIP Trunk to the Avaya SBCE, and ultimately 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 17 was created. The SIP Line form is completed as follows:

- **ITSP Domain Name**: Set to the IP address of the Avaya IP Office LAN2 SIP trunking interface (e.g., **10.10.10.10**). See **Section 2.2**.
- **In Service** and **Check OOS**: These boxes are checked (default).
  - Note that the Out Of Service (OOS) option is used in conjunction with SIP OPTIONS (see **Section 5.10**).
- **Country Code**: Use the default <blank>.
- **Send Caller ID**: Set to **Diversion Header**. This is required by the AT&T IPFR-EF service for call redirection scenarios (e.g., Call Forward, Mobile Twinning).
- **Incoming Supervised Refer**: Set this field to **Always** to enable Avaya IP Office to accept REFER sent by the network during a transfer scenario.
- **Outgoing Supervised Refer**: Set this field to **Always** to enable Avaya IP Office to use REFER (with Replaces) for station initiated call transfer scenarios back to PSTN.
- **Outgoing Blind Refer**: Optional. Enable this option to support Refer (without Replaces) for "Blind" (unattended) transfers (e.g., transfer-to party is still ringing when the transfer operation is completed). If this feature is not enabled then Refer (with Replaces) will be used. **Note – This feature is only supported with SIP telephones.**
- Use the default values for the other fields.
- Click **OK** (not shown).

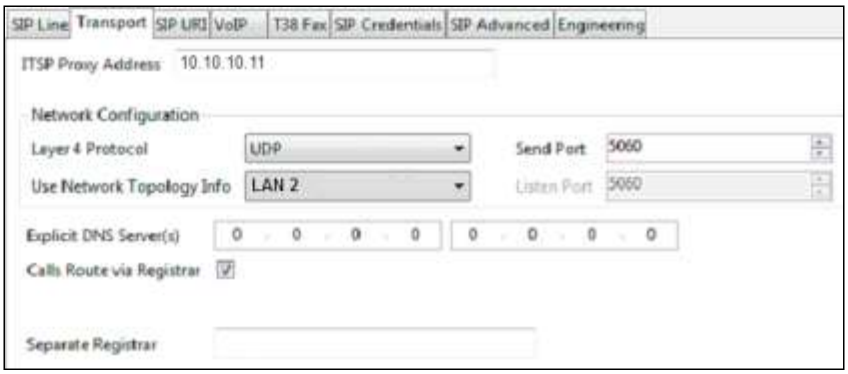
As described in **Section 5.3.5**, the **Name Priority** parameter may retain the default **Favor Trunk** setting, or can be configured to **Favor Directory**. As shown below, the default **Favor Trunk** setting was used in the reference configuration.



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., **10.10.10.11**).
- **Network Configuration → Layer 4 Protocol**: Set to **UDP**.
- **Network Configuration → Send Port**: Set to **5060** (default).
- **Network Configuration → Use Network Topology Info**: Set to **LAN 2**.
- Verify **Calls Route via Registrar:** Enabled (default).
- **Click OK** (not shown).



5.4.5. SIP Line - SIP URI tab

A SIP URI entry needs to be created to match each number that Avaya IP Office and the service provider will accept on this line. Select the **SIP Line → SIP URI** tab. On this form a list of the DNIS digits delivered by AT&T is created.

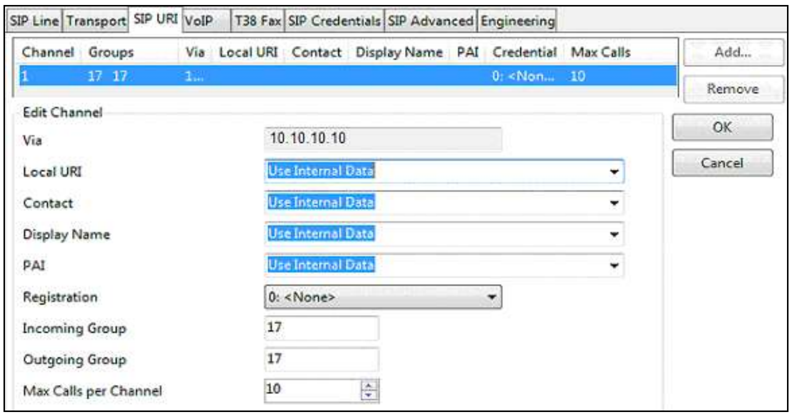
Note – In the reference configuration the AT&T IPFR-E service delivered seven DNIS digits in the R-URI. The entries below match on these DNIS digits, not the dialed DID number.

To add a new SIP URI, click the **Add…** button. At the bottom of the screen, a **New Channel** area will be opened. Two types of entries are used:

1. **<u>Type 1</u>**: A “global” entry that will use the contents of SIP headers containing “called party info” information. This type of entry is used for inbound calls to Avaya IP Office Users, Hunt Groups, or Voicemail access where the matching AT&T DNIS digits are specified on their corresponding **SIP Settings** (**Section 5.3.4**) or **SIP** tabs (see **Section 5.5**). Otherwise, the call will be denied. In this method the following information is specified:
    - The **Via** field will automatically be populated with the IP address of the LAN 2 interface with which the SIP trunk is associated (see **Section 5.3.3**).
    - **Local URI**, **Contact**, **Display Name**, **and PAI** fields: Set these fields to **Use Internal Data**.

    Note – Setting the PAI field to Use Internal Data causes Avaya IP Office to use PAI for privacy (see **Sections 5.4.8** and **5.9**).

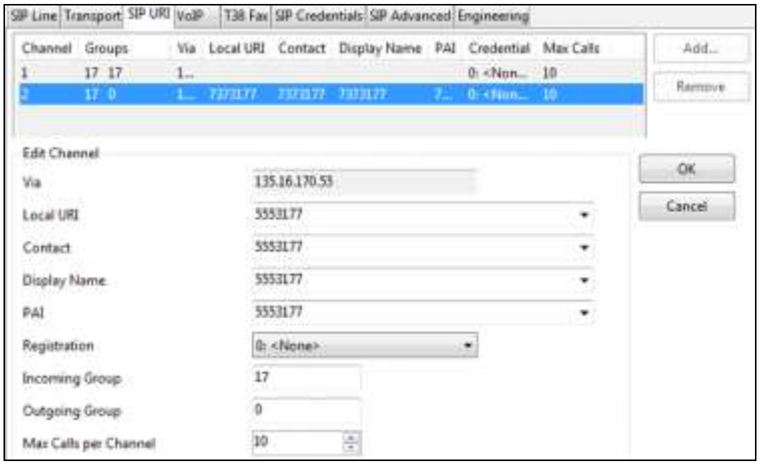
- Verify **Registration:** Set to the default **0: <None>**.
- **Incoming Group:** Set to **17** (SIP Line 17). This value references the table created with **Incoming Call Routes** in **Section 5.7**.
- **Outgoing Group:** Set to **17** (SIP Line 17). This will be used for routing outbound calls to AT&T via the **Short Codes** (**Section 5.6**) or **ARS** configuration (**Section 5.8**).
- **Max Calls per Channel:** 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**.



2. **Type 2**: This is an explicit entry matching inbound DNIS digits from AT&T. This method must be used for Avaya IP Office call destinations that cannot specify matching DNIS digits from AT&T. These call destinations may be Short Codes (e.g., Auto Attendant and Meet-Me conference), or other inbound destinations that do not have a SIP tab. For this method the following information is specified:
   - **Local URI**, **Contact**, PAI, and **Display Name**: Set to an AT&T DNIS number (e.g., **5553177).**

Note – Setting the PAI field to a number causes Avaya IP Office to use PAI for privacy (see **Sections 5.4.8** and **5.9**).

   - Verify **Registration:** Set to the default **0: <None>**.
   - **Incoming Group:** Set here to **17** (SIP Line 17). This value references the **Incoming Call Routes** in **Section 5.7**.
   - **Outgoing Group:** For destinations such as Auto Attendant and Meet-Me conference, an outbound group is not required. In these cases enter **0**. Otherwise specify 17.
   - **Max Calls per Channel:** 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.
   - Repeat these steps as required, and click **OK** to save the information.



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

5.4.6. SIP Line - VoIP tab

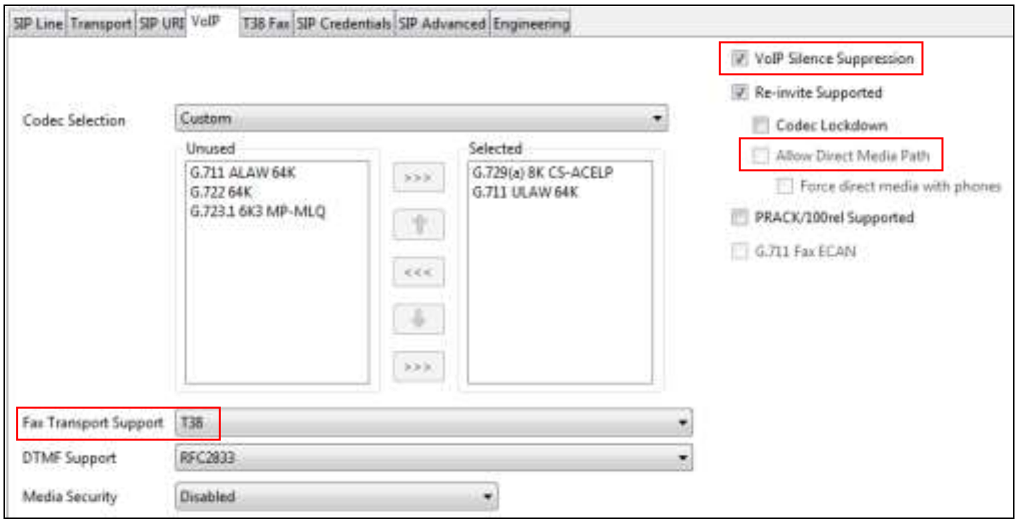
Select the **SIP Line → VoIP** tab.

- 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 IPFR-EF network.
- T.38 fax was used in the reference configuration. Set the **Fax Transport Support** drop-down menu to **T.38**. Note that Error Correction Mode (ECM) is enabled by default on the **T.38 Fax** tab (**Section 5.4.7**). ECM is supported by the AT&T IPFR-EF service. G.711 fax also worked in the reference configuration (T.38 option disabled); however T.38 is the preferred method.

Note – With T.38 specified, the Avaya IP Office Direct Media feature cannot be selected. However if G.711 fax is selected, Direct Media should still not be used. See **Section 2.2**. Also, see **Section 2.2** relating to an issue with Super G3 (SG3) fax.

- The **DTMF Support** parameter can remain set to the default value **RFC2833**.
- 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.
- Click **OK** (not shown).

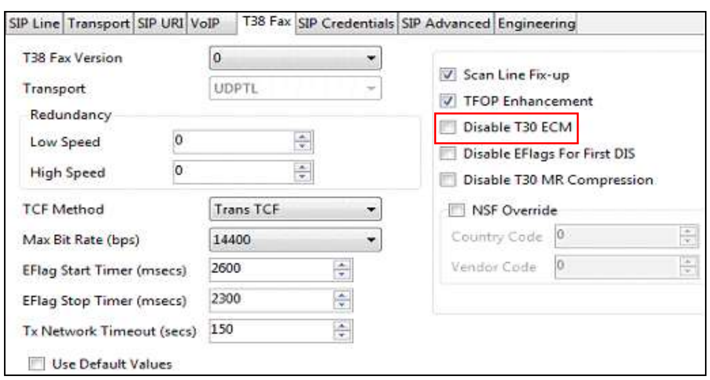
Note - By default the VoIP Silence Suppression box is not checked, disabling the use of the G.729B codec. The AT&T IPFR_EF service specifies G.729B by default. Therefore G.729B was specified in the reference configuration. However G.729A (no silence suppression) is acceptable as well. If silence suppression is desired, check this box, and enable the **VoIP Silence Suppression** option on the **Extension** form **VoIP** tab for the various IP endpoints (see **Section 5.5**).



5.4.7. SIP Line - T38 Fax Tab

Note - 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**).

- Unselect the **Use Default Values** option.
- Set the **T38 Fax Version** option to **0** (zero).
- Verify that **Disable T30 ECM** is *not* checked, and select **Ok** (not shown).



5.4.8. SIP Line – SIP Advanced Tab

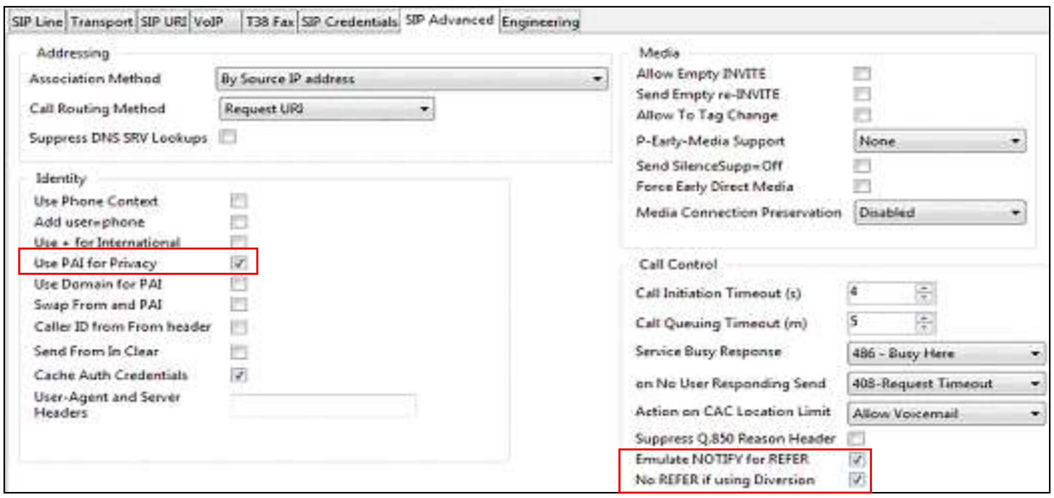
By default, Avaya IP Office will use the PPI (P-Preferred-Identity) header for signaling user information when privacy is invoked. However, AT&T utilizes the PAI (P-Asserted-Identity) header for privacy. Therefore Avaya IP Office is configured to use the PAI header to pass the calling party information for authentication and billing when privacy is used (see **Sections 5.4.5** and **5.9**).

- Select **Emulate NOTIFY for Refer**.

Note – The AT&T IPFR-EF service does not support NOTIFY. Some Avaya endpoints (e.g., Avaya Communicator for Windows) require receipt of a NOTIFY when Refer based call transfers are performed. This option will send a NOTIFY to these endpoints.

- Select the **Use PAI for Privacy** option, and click **Ok** (not shown).

Note – By default, Avaya IP Office sends Refer in addition to Diversion header, for call forward scenarios. However AT&T only requires Diversion header. Therefore in the reference configuration the **No Refer if using Diversion** was selected.



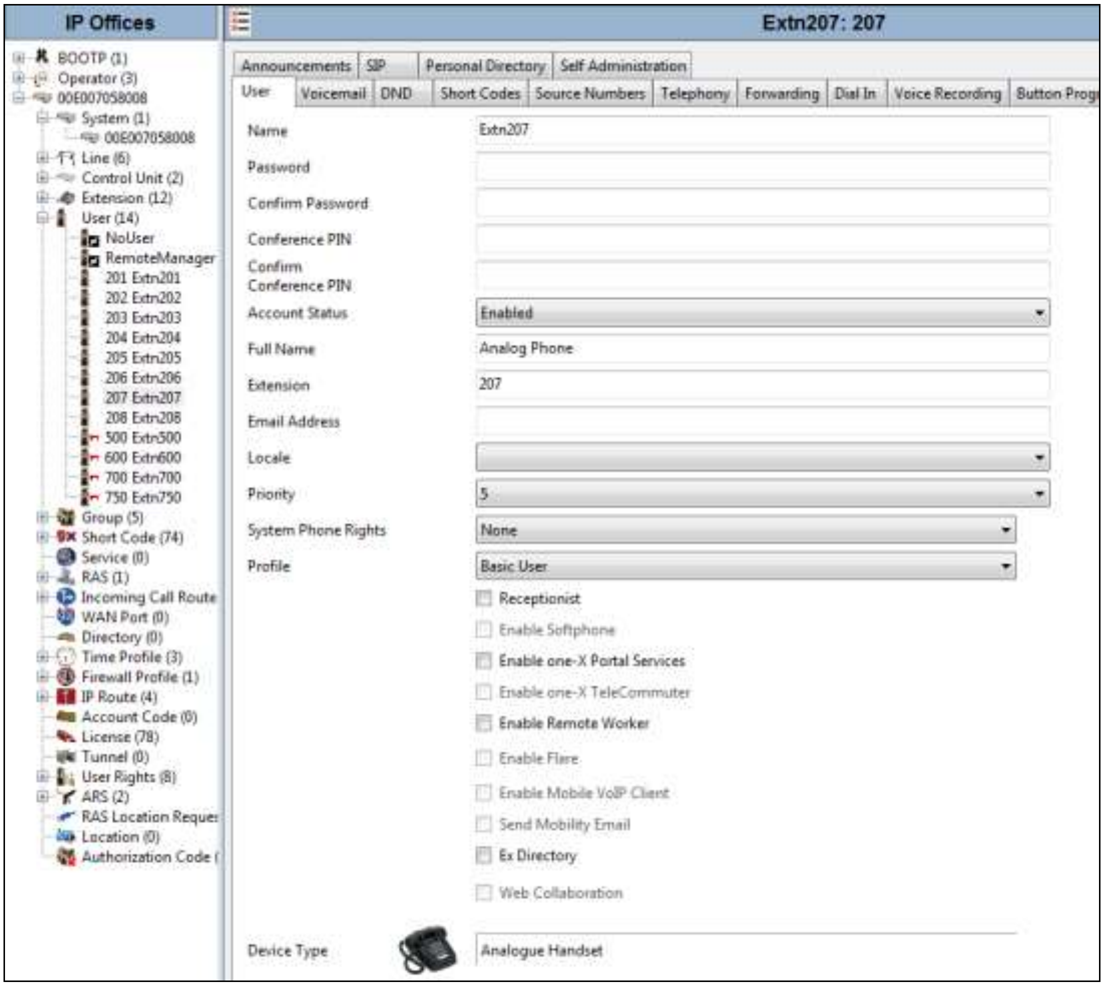
5.5. Users, Extensions, and Hunt Groups

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

5.5.1. Analog User 207

The following screen shows the **User** tab for analog phone User **207**. This user corresponds to the Avaya Analog 6211 set.

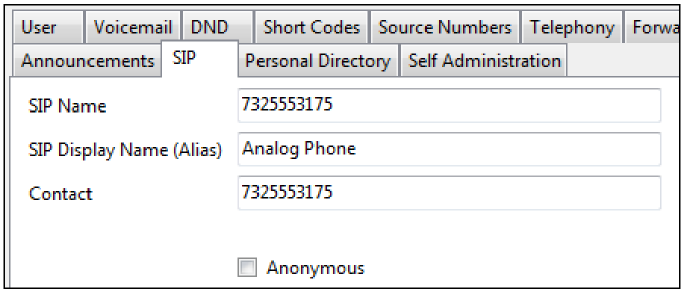
1. To add a User, right click on **User** in the Navigation pane, and select **New** (not shown). To edit an existing User, select **User** in the Navigation pane, and select the appropriate user to be configured.



The following screen shows the **SIP** tab for User **207**.

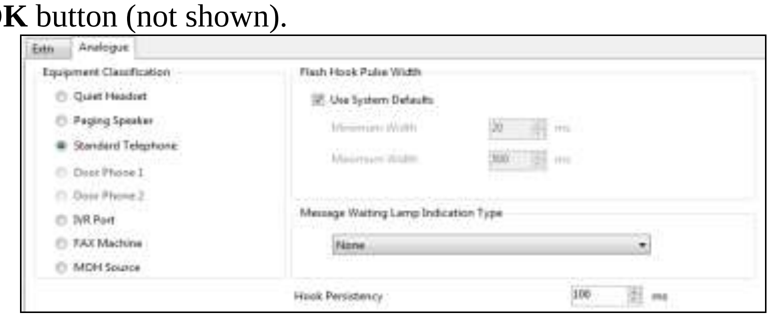
- The **SIP Name** and **Contact** parameters are configured with the associated AT&T DNIS number of the user, (e.g., **7325553175**). These parameters configure the user part of the SIP URI in the From and Contact headers for outgoing SIP trunk calls, and allow matching of the SIP URI for incoming calls, without having to enter this number as an explicit SIP URI for the SIP Line (see **Section 5.4.5**).
- 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 5.9**). See **Section 5.6.3** for a method of using a Short Code (rather than static user provisioning) to place an anonymous call.

Note – See **Section 5.9** regarding the use of privacy with Avaya IP Office and AT&T.



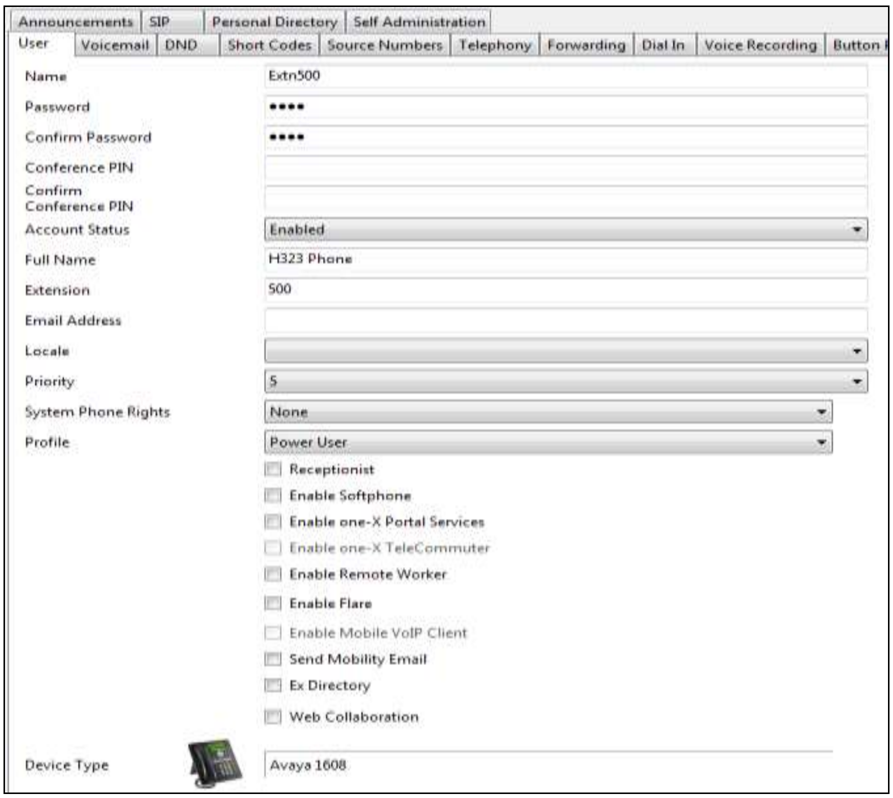
2. Analog (or digital) phone extension ports are either integral to the control unit or added by the installation of an analog or digital phone expansion module. Analog (or digital) extension records are automatically created for each physical extension port within the system. These ports cannot be added or deleted manually. For Server Edition, non-IP extensions are only supported on Expansion System (V2) units. Based on the hardware configuration used in the reference configuration, analog ports 207 and 208 are automatically defined by the system.
   - To edit an existing analog extension, select the appropriate extension to be configured (e.g., **207**).

   - Select the Analogue tab and verify that **Standard Telephone** is selected, and click the **OK** button (not shown).

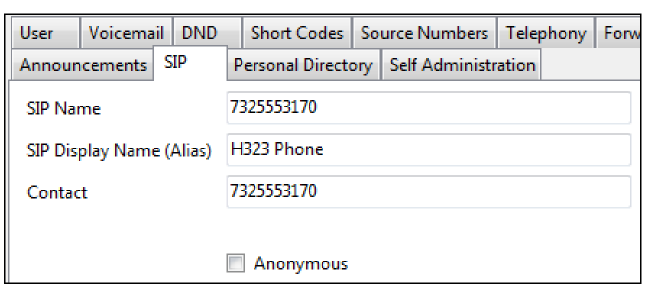


5.5.2. IP Phone User 500

1. Following the steps shown in **Section 5.5.1**, create a 1608 H.323 IP phone user (e.g., **500**). Note that this user will be granted "Power User" features.
   - **Password**: This password is used by user applications such as SoftConsole, Phone Manager and TAPI, or users with Dial In access. Note that this is ***not*** the user's phone log in code (see the information on the **Telephony → Supervisor Settings** tab below), or their Voicemail mailbox password (see information on the **Voicemail** tab below).
   - The **Profile** parameter is set to **Power User**. This gives this user access to additional Avaya P Office features. See **[1]** for more information.

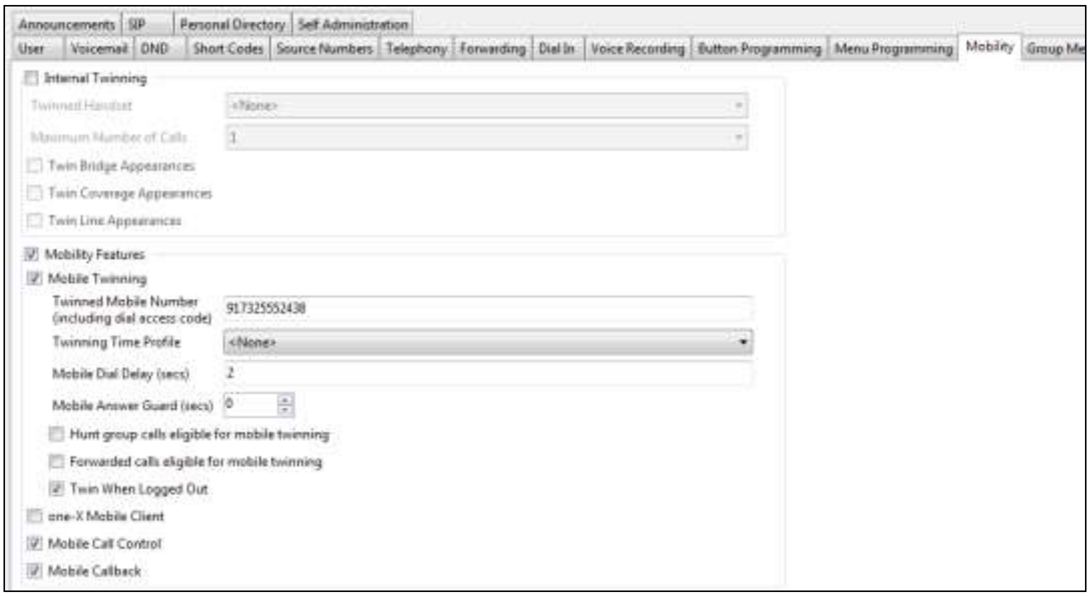


Like the analog user 207, the **SIP** tab for 500 is configured with a **SIP Name** and **Contact** specifying the user's associated AT&T number (e.g., **7325553170**).



User **500** will use the Avaya IP Office Mobile Twinning feature.   The following screen shows the **Mobility** tab for User Extn500.

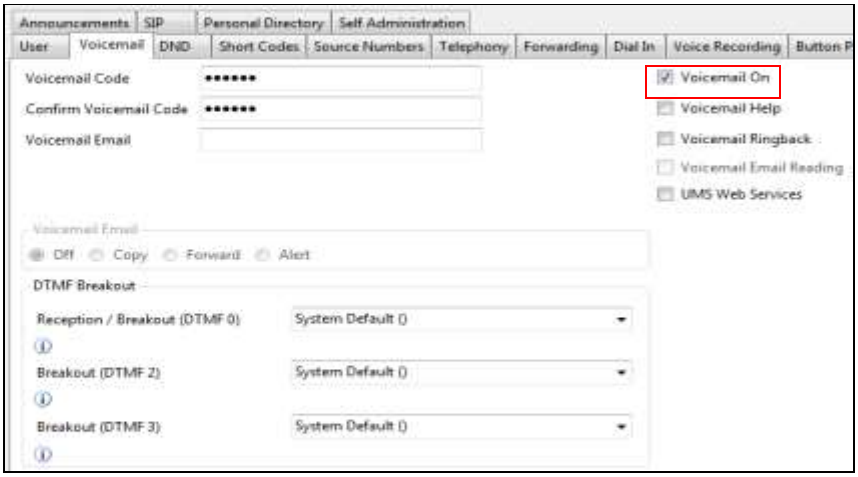
- The **Mobility Features**, **Mobile Twinning**, **Twin When Logged Out**, **Mobile Call Control**, and **Mobile Callback** boxes are checked**.**
- The **Twinned Mobile Number** field is configured with the number to dial to reach the twinned mobile telephone, in this case **917325552438** (note that the outbound call access code **9** to ARS is specified, as described for the Short Code **9N;** in **Section 5.6.2**).
- The **Mobile Call Control** and **Mobile Callback** features are accessed via Short Codes (as shown in **Section 5.6.4**), and **Incoming Call Routes** (as shown in **Section 5.7**).



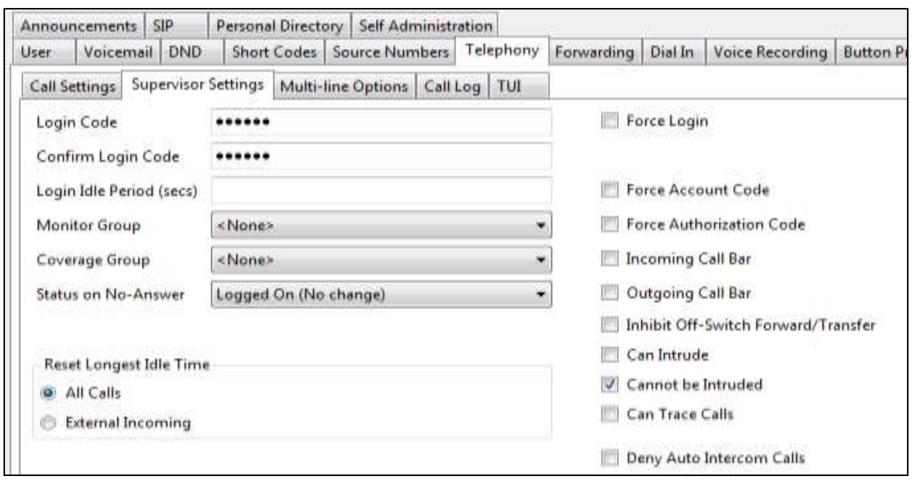
Avaya IP Office offers a feature where users can define names in a Personal Directory, and display these names, based on the inbound calling number. The following screen shows the **Personal Directory** tab for User **500**. With the configuration shown below, if user 500 receives an inbound AT&T call from the telephone number **7325552438**, the phone will display the name "PSTN Phone" (along with the number), even if AT&T provided a different name in the SIP INVITE message sent to Avaya IP Office.

Note – In the reference configuration, the SIP Line is configured with **Name Priority → Favor Trunk** (see **Section 5.4.3**). This setting needs to be changed to **Name Priority → Favor Directory**, to enable this feature.

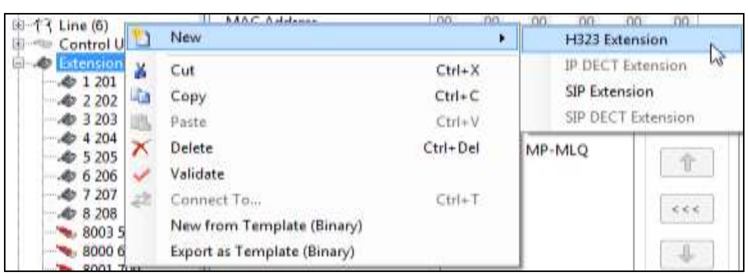
The following screen shows the **Voicemail** tab for user 500. The **Voicemail On** box is checked and a Voicemail password can be configured using the **Voicemail Code** and **Confirm Voicemail Code** parameters.



Select the **Telephony → Supervisor Settings** tab as shown below. The **Login Code** will be used by the telephone user as the phone login password.



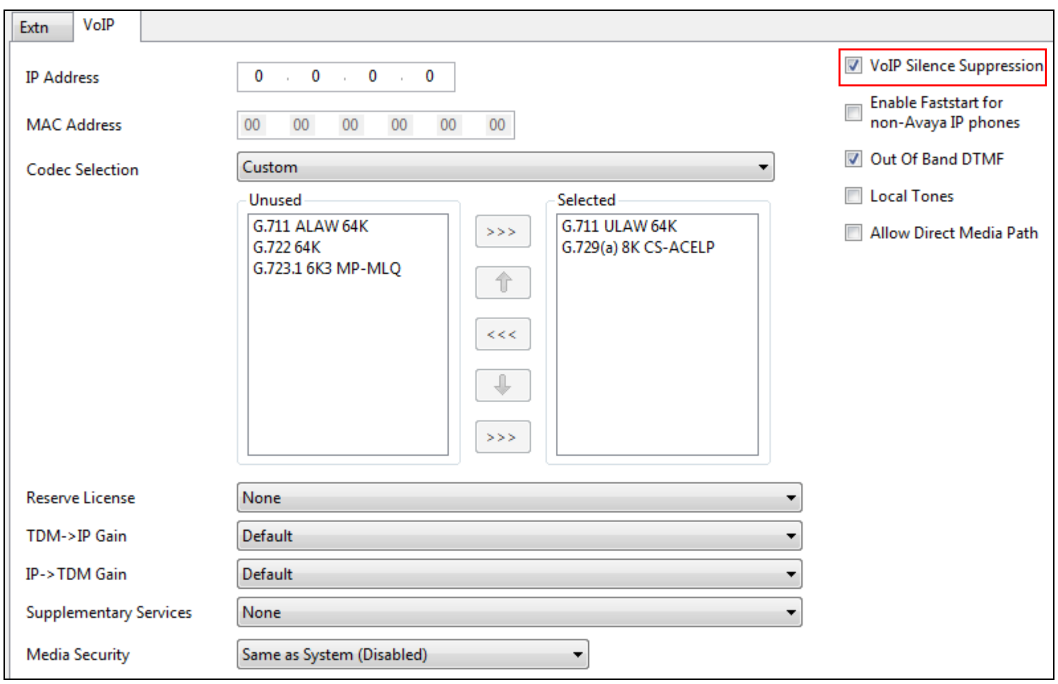
2. To create an associated extension, right click on **Extension** in the Navigation Pane, and select **H323 Extension**.



On the **Extn** tab, enter the **Base Extension** (e.g., **500**). Note that the **Extension ID** field will auto populate.

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

- Keep the **IP Address** field as the default value (**0.0.0.0**).
- As described in **Section 5.3.7**, a custom codec list is used. Populate the **Selected** column with **G.711 ULAW 64K** as the first codec and **G.729(a) 8K CS-ACELP** as the second codec.
- Select **VoIP Silence Suppression** (see **Section 5.3.7**).
- Click the **OK** button (not shown).



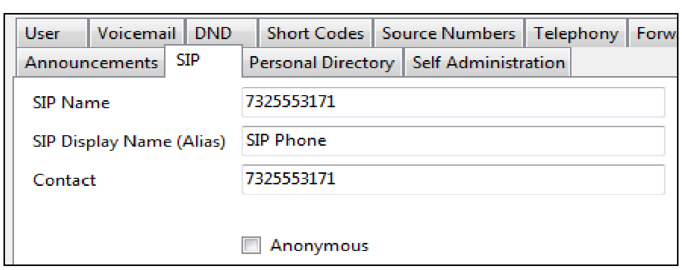
5.5.3. SIP Telephone Users (Avaya 1120E and Avaya Communicator)

In the reference configuration, an Avaya 1120E SIP telephone and Avaya Communicator softphone were provisioned as SIP users.

5.5.3.1 SIP Avaya 1120E

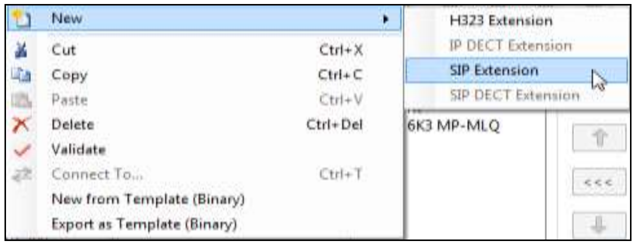
1. The following screen shows an 1120E Telephone provisioned in the **User** tab for User **600**. The provisioning of this user is the same as for the H.323 station in **Section 5.5.2**. Note that this station is set as a Basic User.

Like the H.323 500 user, the **SIP** tab for 600 is configured with a **SIP Name** and **Contact** specifying the user's associated AT&T number (e.g., **7325553171**).

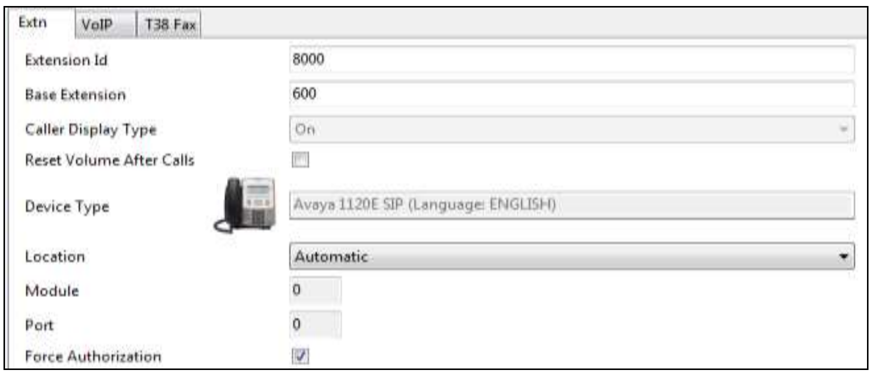


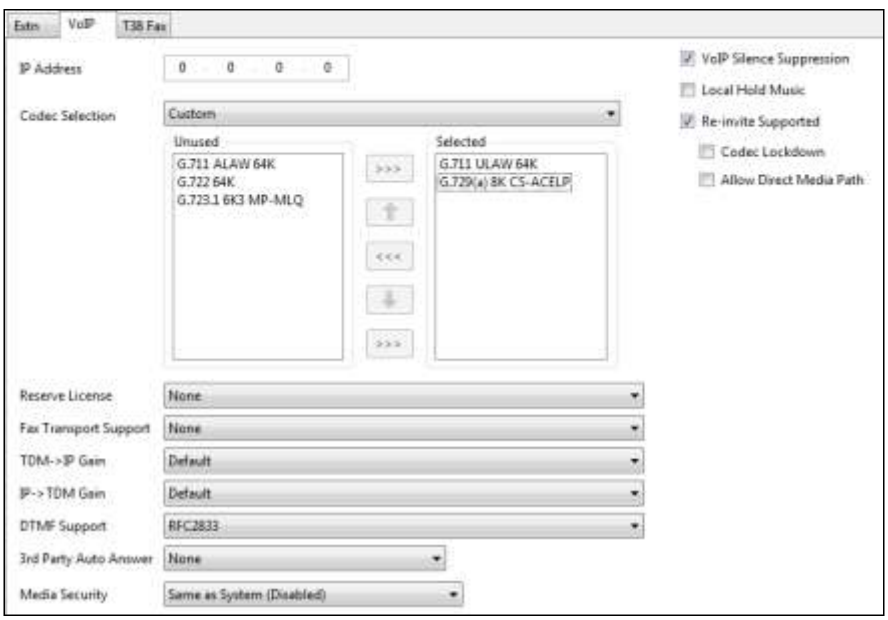
Voicemail and a **Login Code** are also defined following the examples shown for the H.323 User 500 in **Section 5.5.2**.

2. Following the steps shown in **Section 5.5.2** for the H.323 phone, create a corresponding **SIP Extension** for the 1120E SIP telephone (e.g., **600**).



The following screens show the **Extn** and **VoIP** tabs for the Avaya 1120E extension 600 (the **T.38 Fax** tab is not used). The **Extension ID** on the Extn tab is auto populated by the system.

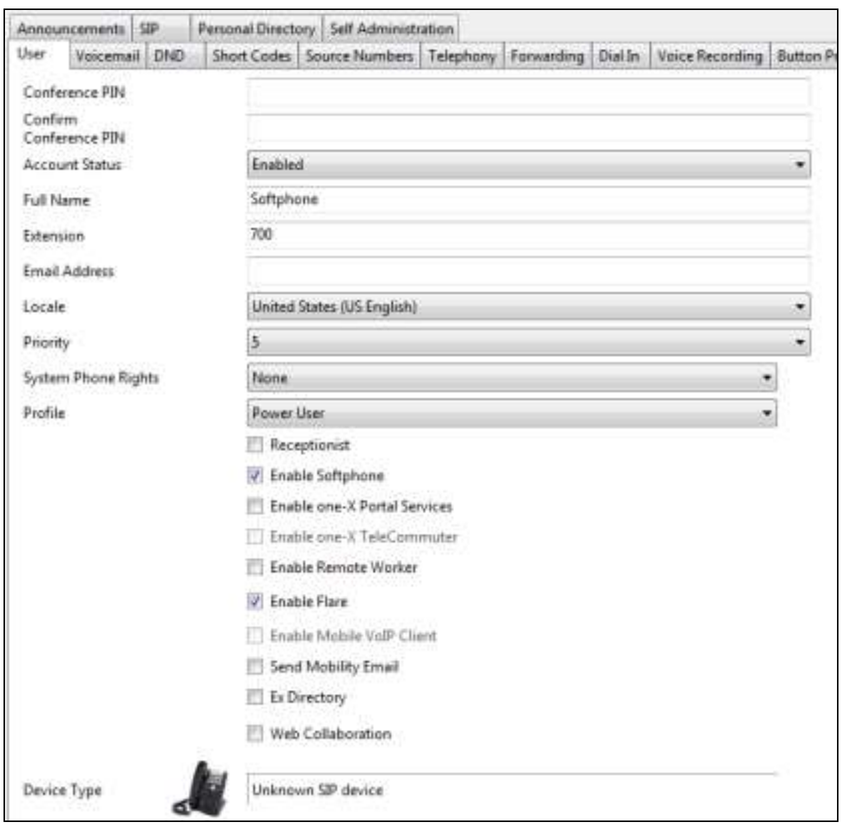


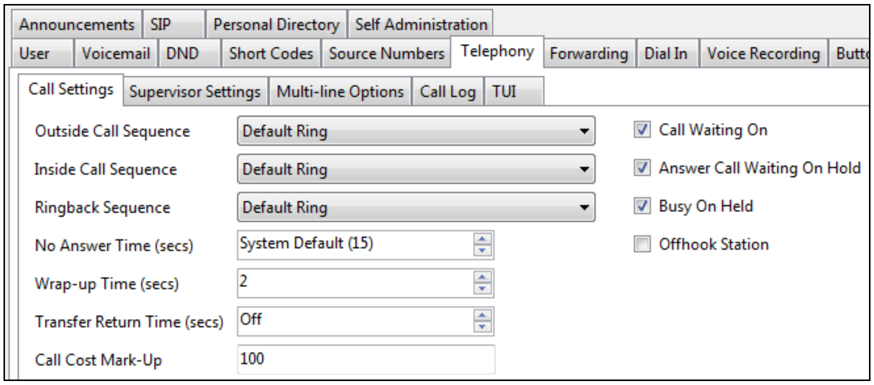


5.5.3.2 SIP Avaya Communicator Softphone

Repeat the steps shown in **Section 5.5.3.1** to create user **700** with the following settings.

1. Defining a **User**
   - **User** tab (shown below).
     - **Extension** = **700**
     - The **Enable Softphone** box is checked.
     - The **Enable Flare** box is checked.
   - **SIP** tab (not shown).
     - **SIP Name** and **Contact** specifying the user's associated AT&T DNIS number (e.g., **7325553172**).
   - **Voicemail** tab (not shown).
     - The **Voicemail On** box is checked.
   - **Telephony→ Call Settings** tab (shown below).
     - In the reference configuration, the **Call Waiting On** box, to allow multiple call appearances and transfer operations, was enabled. However depending on the desired call behavior, this setting may be mutually exclusive with the default **Busy On Held** setting. Combinations of these options should be attempted to achieve the desired effect.
   - **Telephony→ Supervisor Settings** tab (not shown)
     - The **Login Code** is specified.



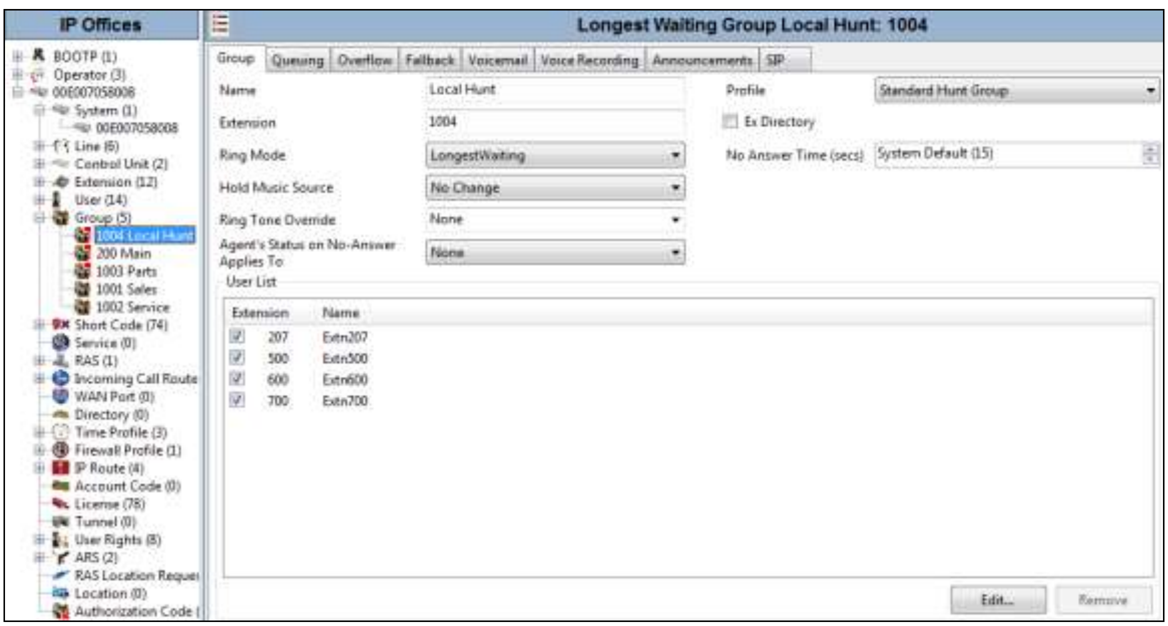


2. Define an **Extension** for the Avaya Communicator Softphone (e.g., **700**).
   - The **Extn** and **VoIP** tabs (not shown), are similar to those shown for extension 600 in **Section 5.5.3.1**.

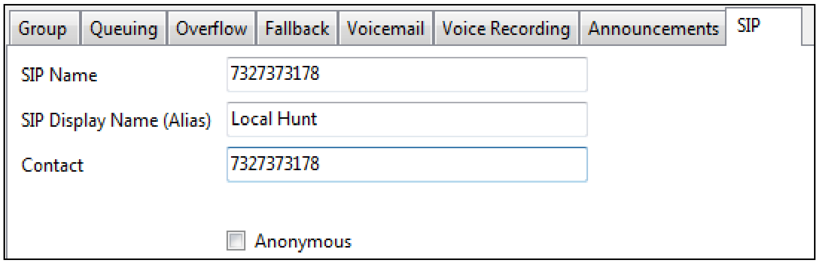
5.5.4. Hunt Groups

Users may 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** (not shown). To view or edit an existing hunt group, select **Group** from the Navigation pane, and the appropriate hunt group from the Group pane.

The following screen shows the **Group** tab for hunt group **1004**. This hunt group was configured to contain the Analog telephone (207), the H.323 telephone (500), SIP telephone (600), and the SIP Softphone (700). In the reference configuration, these telephone extensions are rung based on idle time, due to the **Ring Mode** setting **LongestWaiting**. Click the **Edit** button to change the **User List** included in the Hunt Group from the list of available users.



The following screen shows the **SIP** tab for hunt group **1004**. The **SIP Name** and **Contact** are configured with the AT&T DNIS number **7325553178**. In **Section 5.7**, an **Incoming Call Route** will map **5553178** to this hunt group.



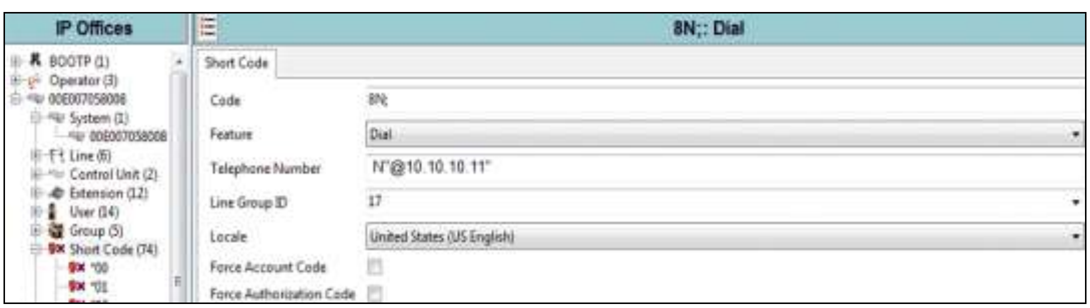
5.6. Short Codes

Avaya IP Office provides predefined Short Codes, however new Short Codes may be defined to match number strings to an action. To add a Short Code, right click on **Short Code** in the Navigation pane, and select **New** (not shown). To edit an existing Short Code, click **Short Code** in the Navigation pane, and the Short Code to be configured in the Group pane.

5.6.1. Short Code for Direct Dialing (no ARS access)

In the screen shown below, the Short Code **8N;** is illustrated (note the semicolon at the end of the string). This Short Code will allow an Avaya IP Office user to dial the digit 8 followed by any telephone number, symbolized by the letter **N**, to reach the SIP Line to AT&T. The variable **N** can be any number string. Note that when users dial 8 plus the number, Avaya IP Office will wait for an end of dialing timeout, then send a SIP INVITE.

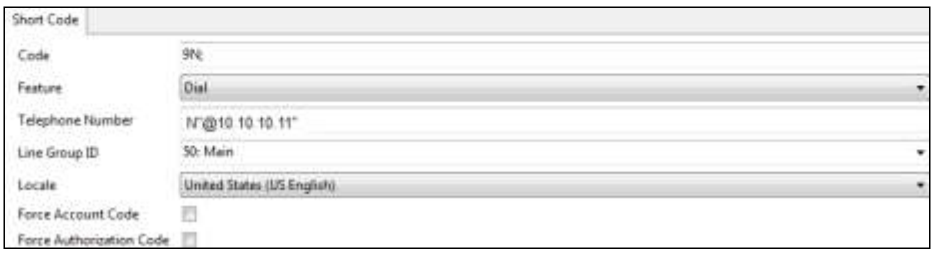
- The **Code** parameter is set to **8N;**
- The **Feature** parameter is set to **Dial**
- The **Telephone Number** parameter is set to **N"@10.10.10.11"** with the text string beginning with @ in quotes. Address 10.10.10.11 is the AT&T Border Element IP address**.**
- The **Line Group ID** parameter is set to **17**, matching the number of the **Outgoing Group** configured on the **SIP URI** tab of **SIP Line 17** (see **Section 5.4.5**).
- Click the **OK** button (not shown).



5.6.2. Short Code Dialing via Automatic Route Selection (ARS access)

In the screen shown below, the Short Code **9N;** is illustrated (note the semicolon at the end of the string). This simple Short Code will allow an Avaya IP Office user to dial the digit 9 followed by any telephone number, symbolized by the letter **N**, to reach the SIP Line to AT&T. However, Avaya IP Office will first consult the ARS table defined in **Section 5.8**. The variable **N** can be any number string.

- The **Code** parameter is set to **9N;**
- The **Feature** parameter is set to **Dial**
- The **Telephone Number** parameter is set to **N"@10.10.10.11"**
- The **Line Group ID** parameter is set to **50:Main** (default value provided by Avaya IP Office), which directs the call to ARS (see **Section 5.8**).
- Click the **OK** button (not shown).



Other short codes were also defined in the reference configuration:

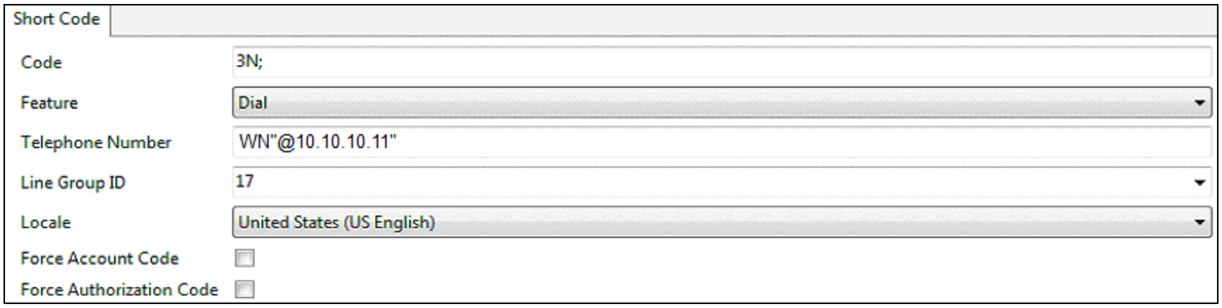
- **3N;** = Outbound dialing with privacy (see **Section 5.6.3**).
- ***17** = Access Voicemail (see **Section 5.6.4.1**).
- ***97** = Mobile Call Back (see **Section 5.6.4.2.1**).
- ***98** = Mobile Call Control (see **Section 5.6.4.2.2**).

5.6.3. Privacy Dialing

Optionally, a Short Code may be added to access the SIP Line anonymously. In the screen shown below, the Short Code **3N;** is illustrated. This code is similar to the **8N;** code. The **Telephone Number** field begins with the letter **W,** which means "withhold the outgoing calling line identification".

Note - When a user dials 3 plus the number, Avaya IP Office will include the user's telephone number in the **P-Asserted-Identity** (PAI) header along with a **Privacy** header of **Id**. The **From** and **Contact** headers will contain **Anonymous**.

- The **Code** parameter is set to **3N;**
- The **Feature** parameter is set to **Dial**
- The **Telephone Number** parameter is set to **WN"@10.10.10.11"**.
- The **Line Group ID** parameter is set to **17**, matching the number of the **Outgoing Group** configured on the **SIP URI** tab of **SIP Line 17** (see **Section 5.4.5**).
- Click the **OK** button (not shown).

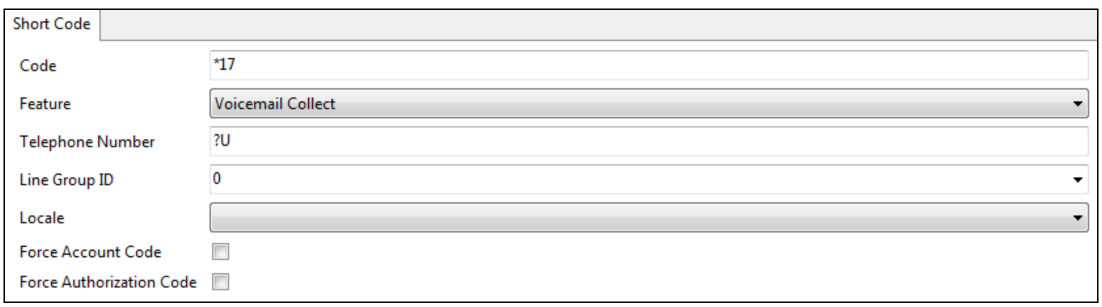


5.6.4. Feature Dialing

Optionally, add a Short Code that can be used to access Avaya IP Office features directly.

5.6.4.1 Voicemail Access

- To access the Voicemail system directly, the code ***17** is defined. This Short Code will be used as one means to allow an AT&T DNIS number to be programmed to route directly to voice messaging (via inclusion of this Short Code as the destination of an **Incoming Call Route** in **Section 5.7**).
- **Feature** = **Voicemail Collect**
- **Telephone Number** = **?U**
- **Line Group** = **0**



5.6.4.2 Feature Name Extension (FNE) Access

Two Avaya IP Office Mobility features, **Mobile Call Control** and **Mobile Callback**, are shown as examples of Feature Name Extension (FNE) Access.

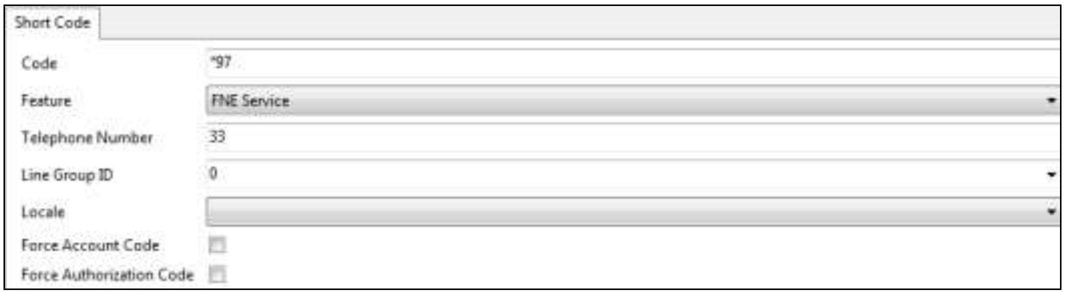
5.6.4.2.1 Mobile Callback

The following screen illustrates the **Code *97** which is defined for **Feature** = **FNE Service** and **Telephone Number** = **33** for the Avaya IP Office Mobility feature **Mobile Callback** (note that 33 is predefined in Avaya IP Office for this feature). Code, ***97** was used as the destination of an Incoming Call Route for an AT&T DNIS number.

In a Mobile Call Back scenario, the PSTN (mobile) number **17325552438**, defined in the **User → Mobility → Twinned Mobile Number** field of associated Avaya IP Office station 500 (see **Section 5.5.2**), calls the DID associated with the Mobile Call Back Short Code *97 (e.g., **17325553177**, see **Section 5.7**), and then hangs up while hearing Avaya IP Office ring back. Avaya IP Office will then call the PSTN caller back at the **17325552438** number.

Note – For this feature to work, the inbound calling number information must match the number provisioned in the associated **User → Mobility → Twinned Mobile Number**. For example, in **Section 5.5.2** the number **917325552438** was defined as the Twinning number for the H.323 set. Therefore, the inbound calling number must match **7325552438.**

- **Feature** = **FNE Service**
- **Telephone Number** = **33**
- **Line Group** = **0**

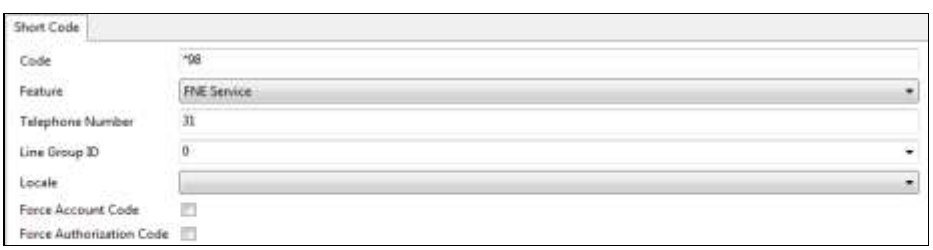


5.6.4.2.2 Mobile Call Control

The following screen illustrates another Mobility Short Code. In this case, the **Code *98** is defined for **Feature** = **FNE Service** and **Telephone Number** = **31** for the Avaya IP Office Mobility feature **Mobile Call Control** (note that 31 is predefined in Avaya IP Office for this feature). Code ***98** was used as the destination of an Incoming Call Route for an AT&T DNIS number.

In a Mobile Call Control scenario the PSTN (mobile) number **17325552438**, defined in the **User → Mobility → Twinned Mobile Number** field of associated station 500 (see **Section 5.5.2**), calls the DID associated with the Mobile Call Control Short Code *98 (e.g., **17325553179**, see **Section 5.7**). Avaya IP Office will return dial tone, allowing the mobile user to make calls as if the calls were made locally from the caller's associated Avaya IP Office extension in the office (e.g., extension 500).

- **Feature** = **FNE Service**
- **Telephone Number** = **31**
- **Line Group** = **0**

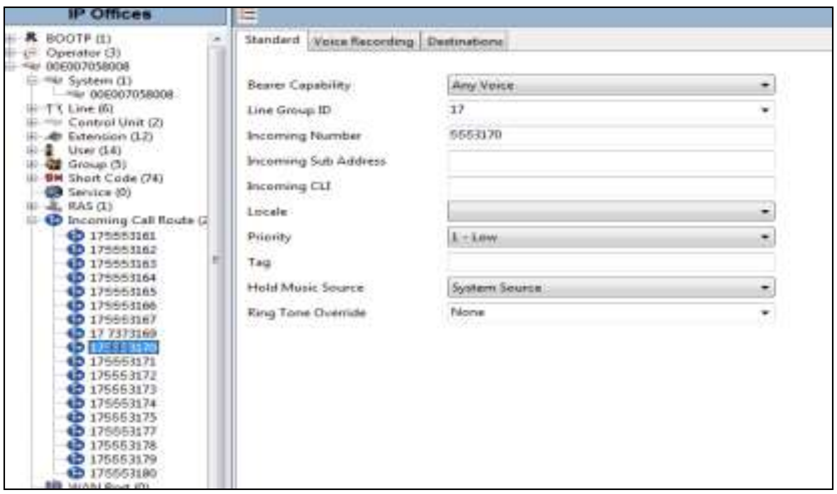


5.7. Incoming Call Routes

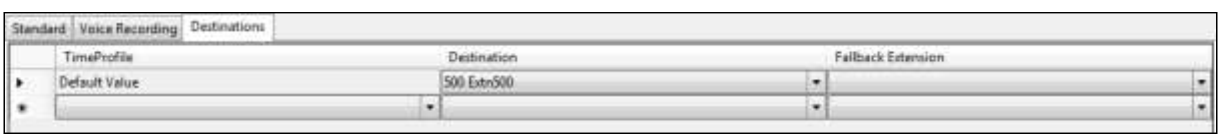
Each Incoming Call Route will map a specific AT&T DNIS number to a destination User, Hunt Group, or Short Code, on Avaya IP Office. To add an incoming call route, right click on **Incoming Call Route** in the Navigation pane, and select **New** (not shown). To edit an existing incoming call route, select **Incoming Call Route** in the Navigation pane, and the appropriate incoming call route to be configured in the Group pane.

Note – In the reference configuration, although a ten digit AT&T access number is dialed in the PSTN (e.g., **7325553170**), the AT&T IPFR-E service delivered seven DNIS digits in the SIP Invite R-URI. Therefore incoming calls to Avaya IP Office will match on the seven digit inbound AT&T DNIS string (e.g., **5553170**). Verify the digits being delivered by AT&T.

In the screen shown below, the incoming call route for **Incoming Number → 5553170** is illustrated. The **Line Group ID** is set to **17**, matching the **Incoming Group** field configured in the **SIP URI** tab for the SIP Trunk to AT&T in **Section 5.4.5**.



Select the **Destinations** tab. From the **Destination** drop-down menu, select the extension to receive the call when AT&T delivers DNIS digits **5553170**. In the reference configuration DNIS digits **5553170** is associated with user **500** (the 1608 H.323 telephone).



Repeat this process to route all AT&T DNIS numbers to additional telephone, as well as other Avaya IP Office destinations (Hunt Group (**5553178**), Voicemail (**5553174**), Short Codes (**5553179**), etc). For example:

Note - The **Destination** menu may not contain all desired destinations (e.g., Short Codes). In these cases the desired destination may be manually typed into the **Destination** field.

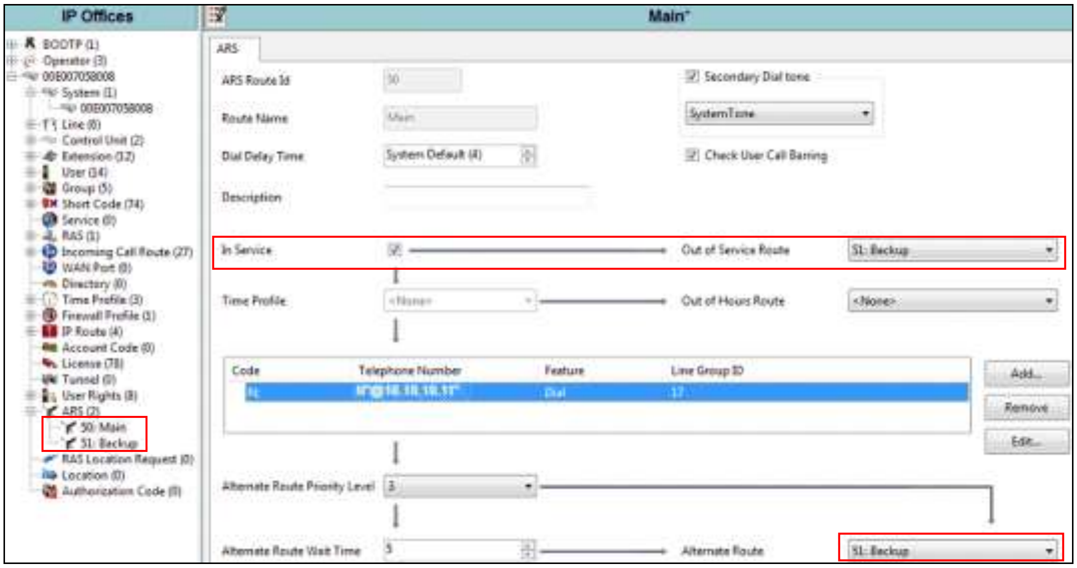
5.8. Automatic Route Selection (ARS) and Alternate Routing

While detailed coverage of ARS is beyond the scope of these Application Notes, this section includes basic ARS screen illustrations and considerations, including alternate routing should the SIP Line be out of service or temporarily not responding. As described in **Section 5.6.2**, Short Code **9N;** was defined for ARS access. Therefore outbound calls via ARS are dialed as 9 plus the number. ARS will strip off the 9 and process the call based on the remaining digits.

- To add a new ARS route, right-click **ARS** in the Navigation pane, and select **New** (not shown). To view or edit an existing ARS route, select **ARS** in the Navigation pane, and select the appropriate route name in the Group pane (e.g., **50:Main**).
- To add a new ARS table entry, click on the **Add** button. To change an existing entry, click on the **Edit** button (note that the Edit button is grayed out until an entry is selected).

The following screen shows an example ARS configuration for the route **Main** (ARS Route ID 50). Note that the **In Service** parameter refers to the ARS form itself, not the SIP Line Group(s) that may be referenced in this form. Also note that if the **In Service** box is *un-checked*, calls are routed to the ARS route specified in the **Out of Service Route** parameter (in the reference configuration, a second ARS route, **51:Backup,** was defined).

- **Code** = **N;** This means any dialed string will be routed to the specified Line Group.
- **Telephone Number** = **N"@10.10.10.11"**, where 10.10.10.11 is the AT&T Border Element IP address.
- **Feature** = **Dial**
- **Line Group ID = 17** (SIP Line 17).



Assuming the primary route is in-service, the number passed from the Short Code used to access ARS (e.g., **9N;** in **Section 5.6.2**) will direct the call to a specific Line Group ID (**17**). If Line Group 17 cannot be used, the call can automatically route to the route name configured in the **Alternate Route** parameter in the lower right of the screen above (e.g., **51:Backup**).

If a primary route experiences a network outage such that no response is received to an outbound INVITE, Avaya IP Office routes the call via the alternate route. The user receives an audible tone when the re-routing occurs and may briefly see "Waiting for Line" on the display. The redirection will occur if a call is made right after a failure of the primary route, as Avaya IP Office has not yet marked the SIP Line out of service as a result of no response to SIP OPTIONS (see **Section 5.10**). Alternatively calls can be delivered via the alternate route when the primary route is manually marked out-of-service, or known to be out-of-service due to prior failure of SIP OPTIONS.

5.9. Placing Privacy / Anonymous Calls

Note - By default, Avaya IP Office will use the PPI (Per-Packet-Information) header for signaling user information when privacy is invoked. However, AT&T utilizes the PAI (P-Asserted-Identity) header for privacy. Therefore, for all the privacy methods described below, Avaya IP Office is configured to use the PAI header to pass the calling party information for authentication and billing. See **Sections 5.4.5**, **5.4.8**, and **5.5** regarding settings for privacy.

As described in **Section 5.6.3**, an Avaya IP Office user can request privacy by dialing the Short Code **3N;** to access the SIP Line.

The Avaya 1600-Series IP Telephones can also request privacy for a specific call, without dialing a unique Short Code, using **Features → Call Settings → Withhold Number**, on the phone itself.

Alternatively, specific users may be configured to always withhold calling line identification by checking the **Anonymous** field in the **SIP** tab for the user (see **Section 5.5**).

For outbound calls with privacy (anonymous) enabled, Avaya IP Office will replace the calling party number in the From and Contact headers of the SIP INVITE message with "anonymous", populate the PAI header with the user information, and insert a Privacy-ID header.

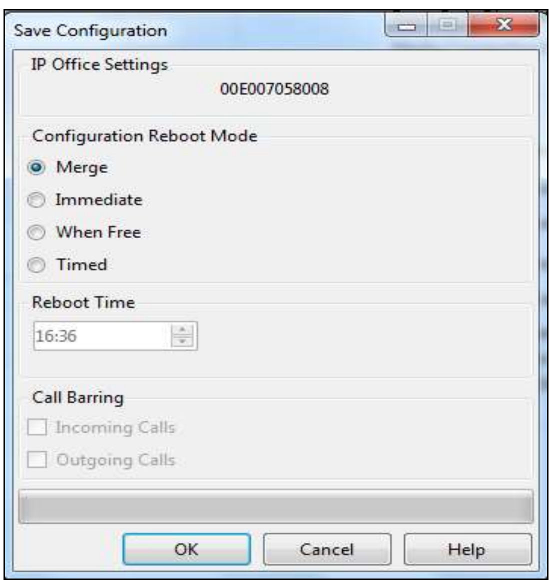
5.10. SIP Options

Avaya IP Office periodically checks the health of the SIP Line by sending a SIP OPTIONS message, based on the **Binding Refresh Time** (see **Section 5.3.3**). In the reference configuration, the Binding Refresh Time is set to 120 seconds (see **Section 2.2**).

5.11. 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. At the top of the Avaya IP Office Manager page click **File → Save Configuration** (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 Flexible Reach – Enhanced Features Configuration

AT&T provides the IPFR-EF service border element IP address, the access DID numbers, and the associated DNIS digits used in the reference configuration. In addition, the AT&T EIPFR features, and their associated access numbers, are also assigned by AT&T.

7. Verification Steps

The following procedures may be used to verify the Avaya IP Office R9.1 with the AT&T IP Flexible Reach - Enhanced Features service configuration.

7.1. AT&T IP Flexible Reach – Enhanced Features

The following scenarios may be executed to verify Avaya IP Office and the AT&T IPFR-EF service interoperability:

- Place inbound and outbound calls, answer the calls, and verify that two-way talk path exists. Verify that the calls remain stable for several minutes and disconnects properly.
- Verify basic call functions such as hold, transfer, and conference.
- Verify the use of DTMF signaling.
- Place an inbound call to a telephone, but do not answer the call. Verify that the call covers to voicemail (e.g., Avaya IP Office imbedded voicemail). Retrieve the message either locally or from PSTN.
- Using the appropriate IPFR-EF access numbers and codes, verify that the following features are successful:
  - Network based Simultaneous Ring – The “primary” and “secondary” endpoints ring, and either may be answered.
  - Network based Sequential Ring (Locate Me) – Verify that after the “primary” endpoint rings for the designated time, the “secondary” endpoint rings and may be answered.
  - Network based Call Forwarding Always (CFA/CFU), Network based Call Forwarding Ring No Answer (CF-RNA), Network based Call Forwarding Busy (CF-Busy), Network based Call Forwarding Not Reachable (CF-NR) – Verify that based on each feature criteria, calls are successfully redirected and may be answered.
- Inbound / Outbound fax using T38 or G.711.
- SIP OPTIONS monitoring of the health of the SIP trunk.
- Incoming and outgoing calls using the G.729 (A or B) and G.711 ULAW codecs.
- Avaya IP Office Mobile twinning to a mobile phone when the associated Avaya IP Office extension is called, as well as Mobility features such as Mobile Callback and Mobile Call Control may also be verified.

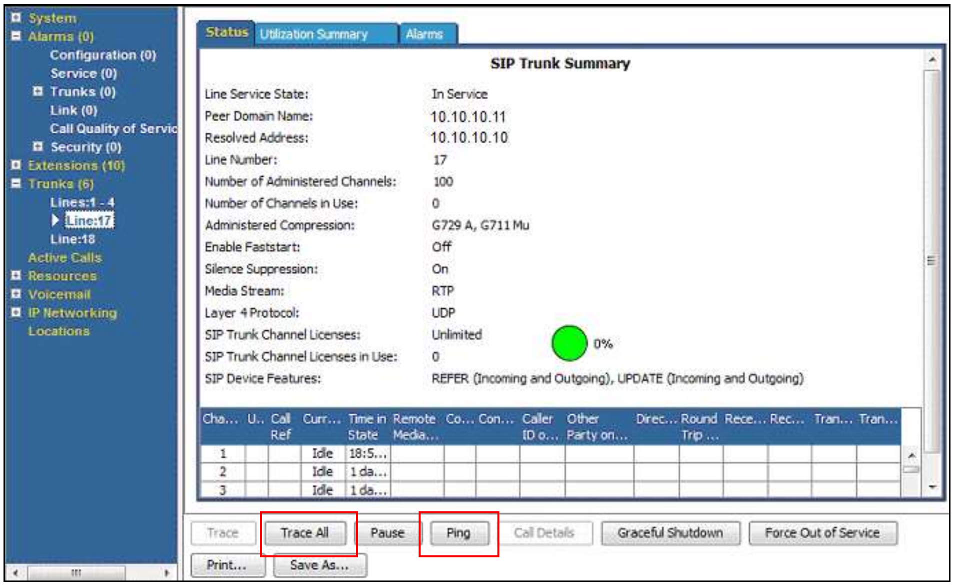
7.2. Avaya IP Office 9.1

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

7.2.1. System Status Application

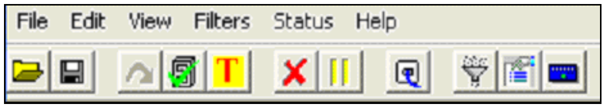
The System Status application can be used to monitor or troubleshoot Avaya IP Office. The System Status application can typically be accessed from **Start → Programs → Avaya IP Office → System Status**. The following screen shows an example **Logon** screen. Enter the Avaya IP Office IP address in the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

After logging in, select **Trunks → Line: 17** from the left navigation menu. (SIP Line 17 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 Controller, however the AT&T Border Controller may not be configured to respond to pings.).

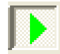


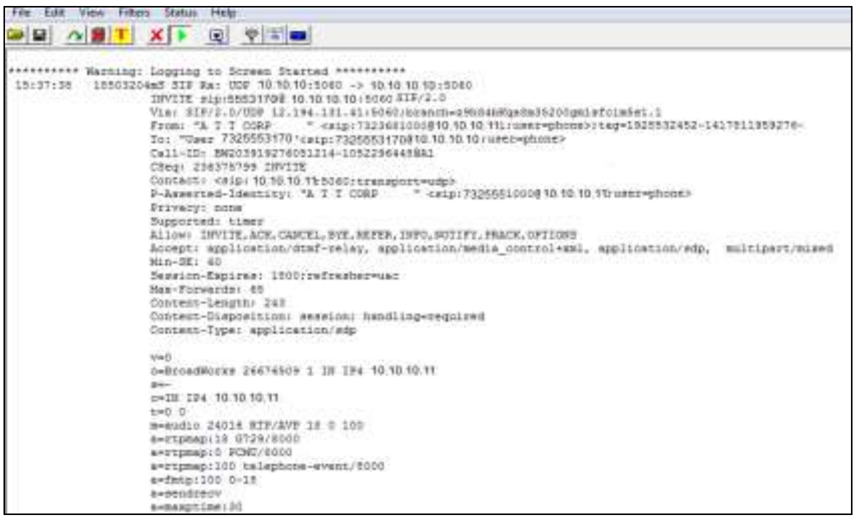
7.2.2. System Monitor Application

The System Monitor application can also be used to monitor or troubleshoot Avaya IP Office functionality (see reference **[1]**. The System Monitor application can typically be accessed from **Start → Programs → Avaya IP Office → Monitor**.

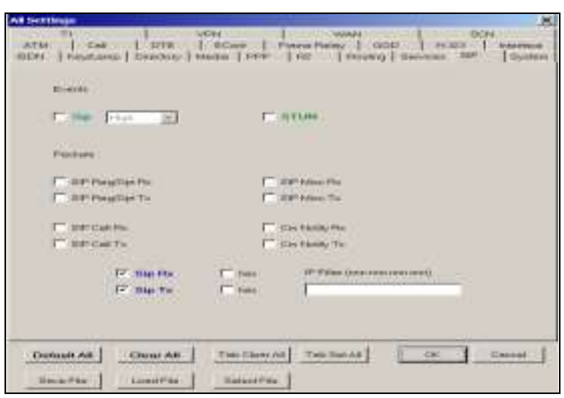


The Monitor will be active at startup. To pause the Monitor, press the Pause  button.

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 to Avaya IP Office SIP telephone 500.



To customize what data is displayed, select the **Options** button  that is third from the right, 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 R9.1 can be configured to interoperate successfully with the AT&T IP Flexible Reach - Enhanced Features service using **AVPN** or **MIS/PNT** transport connections, within the limitations described in **Section 2.2**.

This solution provides users of Avaya IP Office R9.1 the ability to support inbound and outbound calls utilizing an AT&T IPFR-EF SIP trunk service connection, via AVPN or MIS/PNT transport, using the platform and service features listed in **Section 2.1**.

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] Administering Avaya IP Office™ Platform with Manager, Release 9.1, 10.01, December 2014
[2] Additional Avaya IP Office information can be found at:
http://marketingtools.avaya.com/knowledgebase/

AT&T IPFR-EF Service:

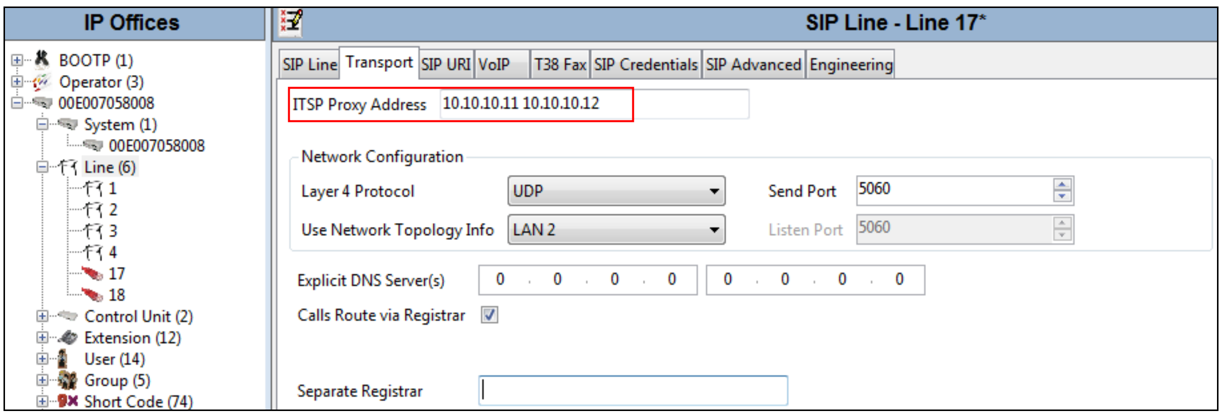
[3] Information on the AT&T IP Flexible Reach service can be found here:
http://www.business.att.com/enterprise/Service/voice-services/null/sip-trunking/

10. Addendum 1 – Multiple AT&T Border Elements

AT&T may provide multiple network border elements for redundancy purposes. Avaya IP Office can be provisioned to monitor both connections.

Given two AT&T border elements **10.10.10.11** (primary) and **10.10.10.12** (secondary), SIP Line (17) defined in **Section 5.4**, can be modified to include the secondary Border Element.

1. Navigate to **Line → 17 → Transport Tab**, and populate the **ITSP Proxy Address** field with the primary address, followed by the secondary address, separated by a space.



2. Click **OK** (not shown), and save the configuration as shown in **Section 5.11**.

When completed, Avaya IP Office will send OPTIONS to both of the AT&T border Elements, as well as responding to AT&T the OPTIONS.

When only the IP addresses are specified, Avaya IP Office will send all outbound calls to the first address specified (10.10.10.10). If the OPTIONS fail for the primary connection, then Avaya IP Office will use the second address (10.10.10.11) for all calls.

However, "weights" may be added to those addresses so that Avaya IP Office can alternate calls between them. The resulting outbound calls will be distributed based on the weight differential. For example, a setting of **10.10.10.11w1 10.10.10.12w1**, will result in a "round-robin" behavior, while a setting of **10.10.10.11w2 10.10.10.12w1**, will result in a "two-to-one" behavior.

Note - Up to four IP addresses may be specified in the **ITSP Proxy Address** field.

Alternatively, the two AT&T Border Elements could be defined to individual SIP Lines (see **Section 5.4**). Then Automatic Route Selection (ARS) would be configured with one SIP Line the primary call path and the other as the backup call path, as mentioned in **Section 5.8**.

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