



Avaya Solution & Interoperability Test Lab

Application Notes for Avaya IP Office Release 10 with AT&T IP Flexible Reach - Enhanced Features – Issue 1.0

Abstract

These Application Notes describe the steps for configuring Avaya IP Office Release 10 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.

These Application Notes complement previously published Application Notes by illustrating the configuration screens and Avaya testing of IP Office Release 10.

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.....	4
2.	General Test Approach and Test Results.....	4
2.1.	Interoperability Compliance Testing.....	5
2.2.	Test Results	5
2.3.	Support	6
3.	Reference Configuration.....	7
3.1.	Illustrative Configuration Information	9
3.2.	Call Flows	10
3.2.1.	Inbound	10
3.2.2.	Outbound.....	11
3.2.3.	Call Forward	12
3.2.4.	Coverage to Voicemail	13
4.	Equipment and Software Validated	14
5.	Avaya IP Office Configuration.....	15
5.1.	Platform Information and Licensing	15
5.2.	System Settings	17
5.2.1.	System Tab.....	17
5.2.2.	LAN 1 Tab	18
5.2.3.	LAN 2 Tab	20
5.2.4.	Voicemail Tab.....	23
5.2.5.	Telephony Tab	23
5.2.6.	VoIP Tab.....	24
5.3.	IP Route.....	25
5.4.	SIP Line.....	26
5.4.1.	Importing a SIP Line Template.....	26
5.4.2.	Creating a SIP Trunk from an XML Template.....	27
5.4.3.	SIP Line – SIP Line tab	28
5.4.4.	SIP Line - Transport tab.....	29
5.4.5.	SIP Line - SIP URI tab.....	30
5.4.6.	SIP Line - VoIP tab.....	31
5.4.7.	SIP Line - T38 Fax Tab	33
5.4.8.	SIP Line – SIP Advanced Tab	33
5.5.	Users, Extensions, and Hunt Groups.....	35
5.5.1.	User 321 (Digital)	35
5.5.2.	User 329 (Avaya Communicator for Windows).....	37
5.5.3.	Hunt Groups.....	41
5.6.	Short Codes	41
5.6.1.	Short Code Dialing via Automatic Route Selection (ARS access).....	42
5.6.2.	Privacy Dialing	42
5.6.3.	Feature Dialing.....	43
5.7.	Incoming Call Routes.....	45
5.8.	Automatic Route Selection (ARS) and Alternate Routing.....	46
5.9.	Placing Privacy / Anonymous Calls.....	48
5.10.	SIP Options.....	48

5.11.	Saving Configuration Changes to Avaya IP Office.....	48
6.	AT&T IP Flexible Reach – Enhanced Features Configuration	49
7.	Verification Steps.....	49
7.1.	AT&T IP Flexible Reach – Enhanced Features	49
7.2.	Avaya IP Office 10.....	50
7.2.1.	System Status Application	50
7.2.2.	System Monitor Application.....	51
8.	Conclusion	53
9.	References	53
10.	Addendum 1 – Multiple AT&T Border Elements	54

1. Introduction

These Application Notes describe the steps for configuring Avaya IP Office Release 10 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 AVPN¹ or MIS/PNT² 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 10 (see **Section 3.2** for call flow examples).

The test environment described in these Application Notes consisted of:

- A simulated enterprise with Avaya IP Office 10, Avaya SIP, H.323 and Analog telephones, as well as a fax machine emulator (Ventafax).
- 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 5.2.6**), it was found that the 1140E 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 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.
3. **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.
4. **Operator calls –** If the SIP Line *National Prefix* field is populated with the default value of “0”, then outbound calls to the operator will fail. The configuration illustrated in these Application Notes includes the National Prefix set to the North American prefix of “1” (**Section 5.4.3**). With this setting, calls to the operator “0” were successful. This anomaly is under investigation by the IP Office product development team.
5. **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 9608 H.323 set, an Avaya 1140E SIP set, as well as Avaya Communicator for Windows (SIP). Fax endpoints are represented by PCs running Ventafax emulation software connected by modem to an Avaya IP Office analog port.
- A single IP Office Application Server was used to provide Voicemail Pro, one-X® Portal, and WebRTC gateway. Voicemail Pro provided the voice messaging capabilities, while one-X® Portal, and WebRTC gateway provided Avaya Communicator for Web capabilities.
- 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.
- 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 10 digits.

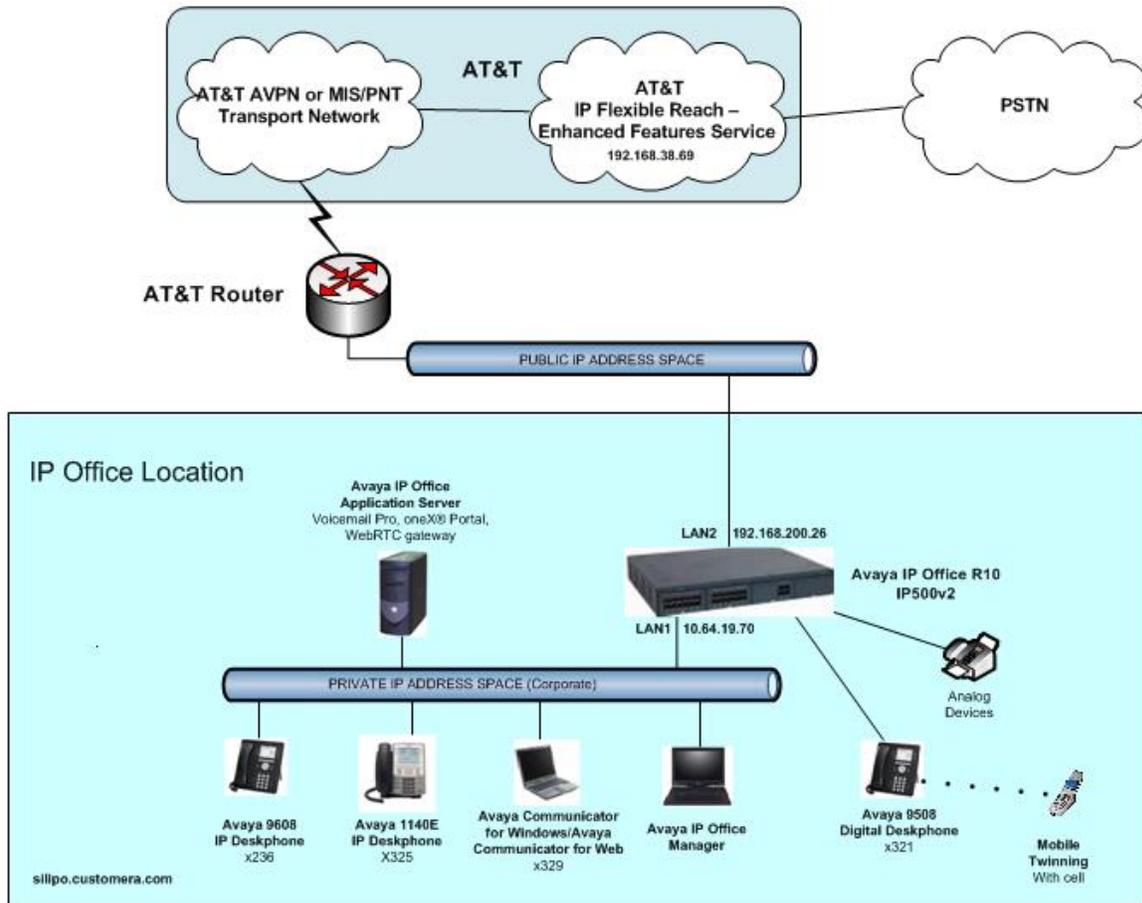


Figure 1: Reference Configuration

Note – For security purposes, any public IP addresses or PSTN routable phone numbers used in the compliance test are not shown in these Application Notes. Instead, public IP addresses have been replaced with private addresses and all phone numbers have been replaced with numbers that cannot be routed.

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 5.2.3**) 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 **192.168.200.26** (Avaya IP Office LAN 2 address), and **192.168.38.69 /192.168.37.149** (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)	10.64.19.70
LAN2 interface, (labeled “WAN” on the chassis), for management access.	192.168.200.26
AT&T IPFR-EF Service	
Border Element IP Address	192.168.38.69 & 192.168.37.149

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

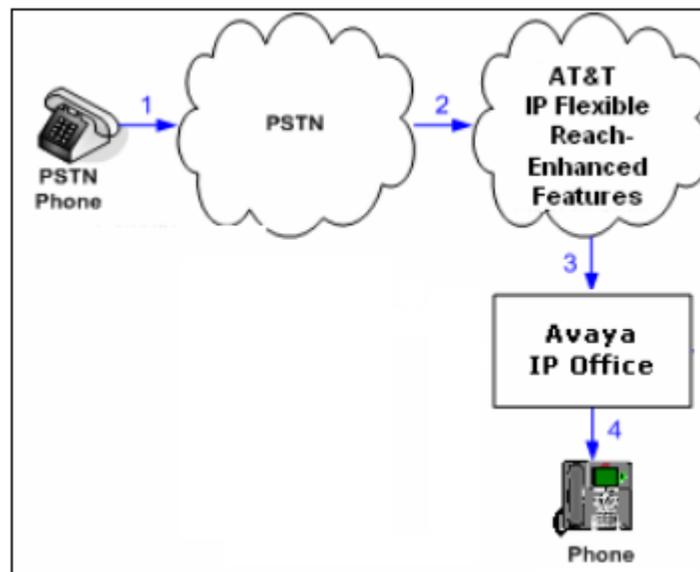


Figure 2: Inbound AT&T IPFR-EF Call

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.

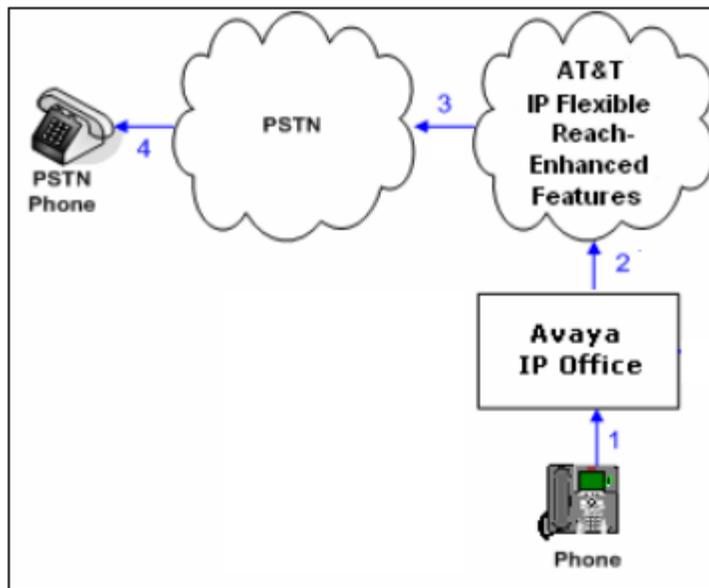


Figure 3: Outbound Call to AT&T IPFR-EF

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 5.4.5**).

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.

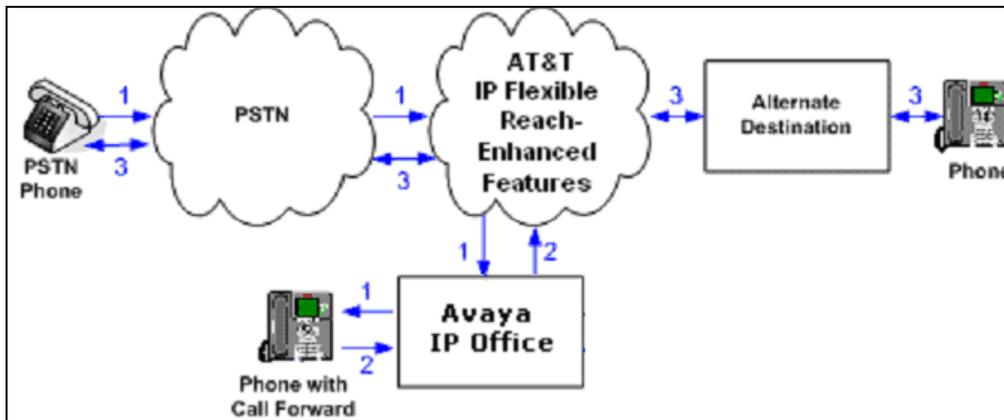


Figure 4: Call Forward

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.

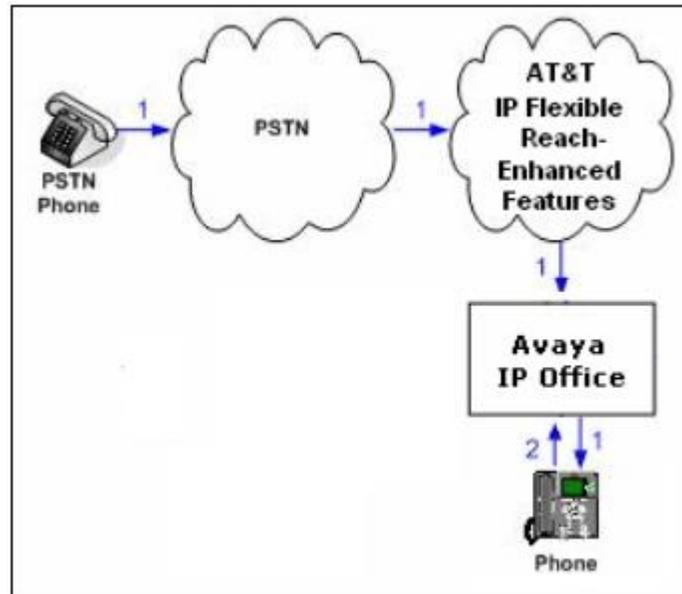


Figure 5: Coverage to Avaya IP Office Voicemail

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	10.0.0.0.0 build 550
Avaya IP Office Application Server <ul style="list-style-type: none">▪ Voicemail Pro▪ Avaya WebRTC Gateway▪ Avaya one-X® Portal for IP Office	10.0.0.0.0 build 550 10.0.0.0.0 build 469 10.0.0.0.0 build 140 10.0.0.0.0 build 980
Avaya 9608 (H.323) Telephone	6.6302
Avaya 1140E (SIP) Telephone	04.04.23
Avaya Communicator for Windows	2.1.2.75
Avaya Communicator for Web	1.0.16.2010
Avaya 9508 Digital Telephone	0.59
Fax device	Ventafax 7.0

Table 2: Equipment and Software Versions

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

5. Avaya IP Office Configuration

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

Avaya IP Office is configured via the Avaya IP Office Manager program. For more information on provisioning Avaya IP Office Manager, consult reference [3]. 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 and Licensing

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 TCM8, and COMBO6210/ATM4 module.

IP Offices	Control Unit	IP 500 V2																								
<ul style="list-style-type: none"> BOOTP (14) Operator (3) SIL-ATT System (1) Line (6) Control Unit (3) Extension (23) User (25) Group (5) Short Code (76) Service (0) RAS (1) Incoming Call R WAN Port (0) Directory (0) Time Profile (0) Firewall Profile (0) 	<table border="1"> <thead> <tr> <th>Dev No.</th> <th>Dev Type</th> </tr> </thead> <tbody> <tr> <td>1</td> <td>IP 500 V2</td> </tr> <tr> <td>2</td> <td>TCM8</td> </tr> <tr> <td>3</td> <td>COMBO6210/ATM4</td> </tr> </tbody> </table>	Dev No.	Dev Type	1	IP 500 V2	2	TCM8	3	COMBO6210/ATM4	<table border="1"> <thead> <tr> <th colspan="2">Unit</th> </tr> </thead> <tbody> <tr> <td>Device Number</td> <td>1</td> </tr> <tr> <td>Unit Type</td> <td>IP 500 V2</td> </tr> <tr> <td>Version</td> <td>10.0.0.0 build 550</td> </tr> <tr> <td>Serial Number</td> <td>00e007058e33</td> </tr> <tr> <td>Unit IP Address</td> <td>10.64.19.70</td> </tr> <tr> <td>Interconnect Number</td> <td>0</td> </tr> <tr> <td>Module Number</td> <td>Control Unit</td> </tr> </tbody> </table>	Unit		Device Number	1	Unit Type	IP 500 V2	Version	10.0.0.0 build 550	Serial Number	00e007058e33	Unit IP Address	10.64.19.70	Interconnect Number	0	Module Number	Control Unit
Dev No.	Dev Type																									
1	IP 500 V2																									
2	TCM8																									
3	COMBO6210/ATM4																									
Unit																										
Device Number	1																									
Unit Type	IP 500 V2																									
Version	10.0.0.0 build 550																									
Serial Number	00e007058e33																									
Unit IP Address	10.64.19.70																									
Interconnect Number	0																									
Module Number	Control Unit																									

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.

The screenshot shows the Avaya IP Office License Management console. The left navigation pane has 'License' selected. The main area displays the license configuration for a 'Remote Server'. The license mode is 'License Normal' and the version is '10.0'. Below this is a table of features and their instance counts.

Feature	Instances	Status	Expiration Date	Source
Avaya Contact Center Select	1	Valid	Never	PLDS Nodal
Avaya IP endpoints	384	Valid	Never	PLDS Nodal
Avaya Mac Softphone	100	Valid	Never	PLDS Nodal
Avaya Softphone Licence	100	Valid	Never	PLDS Nodal
Basic User	384	Obsolete	Never	PLDS Nodal
CTI Link Pro	1	Valid	Never	PLDS Nodal
Devlink3 External Recorder	1	Valid	Never	PLDS Nodal
Essential Edition	1	Valid	Never	PLDS Nodal
Essential Edition Additional Voice...	4	Valid	Never	PLDS Nodal
IP500 Universal PRI (Additional cha...	100	Valid	Never	PLDS Nodal
IP500 Voice Networking Channels	32	Valid	Never	PLDS Nodal
IPSec Tunnelling	1	Valid	Never	PLDS Nodal
Mobile Worker	384	Valid	Never	PLDS Nodal
Office Worker	384	Valid	Never	PLDS Nodal
Power User	384	Valid	Never	PLDS Nodal
R8- Preferred Edition (VM Pro)	1	Valid	Never	PLDS Nodal
Receptionist	4	Valid	Never	PLDS Nodal
SIP Trunk Channels	128	Valid	Never	PLDS Nodal
SM Trunk Channels	128	Valid	Never	PLDS Nodal
Teleworker	384	Valid	Never	PLDS Nodal

5.2. System Settings

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

5.2.1. 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.

The screenshot displays the 'System' configuration page with the following settings:

- Name:** SIL-ATT
- Locale:** United States (US English)
- Location:** 2: SIL
- Contact Information:** Set contact information to place System under special control (with an empty text box below).
- Device ID:** (empty text box)
- TFTP Server IP Address:** 10 . 64 . 19 . 70
- HTTP Server IP Address:** 10 . 64 . 19 . 70
- Phone File Server Type:** Memory Card
- Manager PC IP Address:** 10 . 64 . 19 . 112
- Avaya HTTP Clients Only:**
- Enable Softphone HTTP Provisioning:**
- Automatic Backup:**
- Time Setting Configuration Source:** Voicemail Pro/Manager
- Time Settings:**
 - Time Server Address:** 0 . 0 . 0 . 0
 - Time Offset (hh:mm):** 00:00
- File Writer IP Address:** 10 . 64 . 19 . 105
- AVPP IP Address:** 0 . 0 . 0 . 0
- HTTP Redirection:** Off
- Favor RIP Routes, over static routes:**

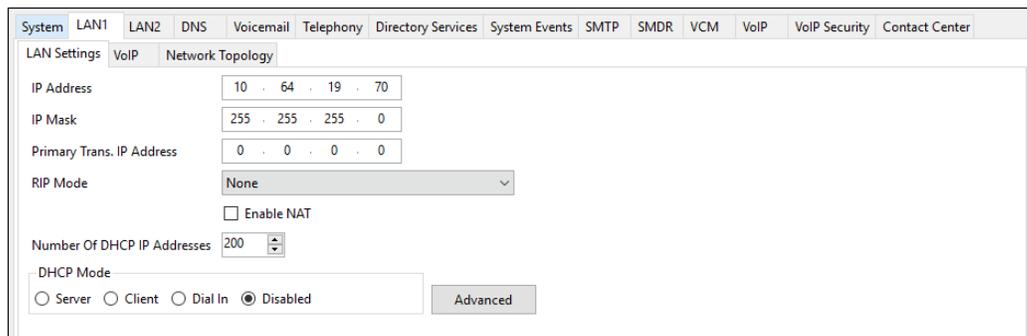
5.2.2.LAN 1 Tab

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

5.2.2.1 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 **10.64.19.70** as used in the reference configuration.
- Click the **OK** button (not shown).



The screenshot shows the 'LAN Settings' configuration window for LAN1. The window has a tabbed interface with 'LAN Settings' selected. The configuration fields are as follows:

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

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

5.2.2.2 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 96x1-Series Telephones used in the reference configuration.
- The **SIP Registrar Enable** box is checked to allow Avaya 11xx (SIP) and Avaya Communicator (SIP) usage.
- The **Domain Name** used in the reference configuration is **silipo.customera.com**.
- In the **Layer 4 Protocol** section, select **UDP/5060** and **TCP/5060**.
- Let all other values default.
- Click the **OK** button (not shown).

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

- H.323 Settings:** Includes checkboxes for H.323 Gatekeeper Enable (checked), Auto-create Extension, Auto-create User, and H.323 Remote Extension Enable. A dropdown menu for H.323 Signaling over TLS is set to Disabled, and the Remote Call Signaling Port is 1720.
- SIP Settings:** Includes checkboxes for SIP Trunks Enable, SIP Registrar Enable (checked), Auto-create Extension/User, and SIP Remote Extension Enable. The SIP Domain Name and SIP Registrar FQDN are both set to silipo.customera.com.
- Layer 4 Protocol:** Includes checkboxes for UDP, TCP, and TLS. The UDP Port is 5060, TCP Port is 5060, and TLS Port is 5061. Remote ports for each are also set to their respective default values (5060 for UDP and TCP, 5061 for TLS).
- Challenge Expiration Time (sec):** Set to 10.
- RTP Settings:** Includes a Port Number Range section with Minimum set to 49152 and Maximum set to 53246. There is a checkbox for Enable RTCP Monitoring on Port 5005 (checked) and an input field for the RTCP collector IP address for phones, which is currently 0.0.0.0.

5.2.2.3 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 (10.64.19.70).
- **Public Port:** Enter **UDP/5060** and **TCP/5060**.

- **Firewall/NAT Type** is set to **Unknown** 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).

The screenshot shows the 'Network Topology Discovery' configuration window. The 'STUN Server Address' is set to '0.0.0.0' and the 'STUN Port' is '3478'. The 'Firewall/NAT Type' is set to 'Unknown'. The 'Binding Refresh Time (sec)' is '60'. The 'Public IP Address' is '0 . 0 . 0 . 0'. There are 'Run STUN' and 'Cancel' buttons. The 'Public Port' section has 'UDP', 'TCP', and 'TLS' all set to '0'. A checkbox for 'Run STUN on startup' is checked.

5.2.3.LAN 2 Tab

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

5.2.3.1 LAN 2 - LAN Settings Tab

- **IP Address:** In the reference configuration the IP Office public address is **192.168.200.26**.
- **IP Mask:** In the reference configuration the IP Office public address subnet mask is **255.255.255.248**.
- Other parameters on this screen are set to the defaults.

The screenshot shows the 'LAN Settings' configuration window for LAN2. The 'IP Address' is '192 . 168 . 200 . 26', 'IP Mask' is '255 . 255 . 255 . 248', and 'Primary Trans. IP Address' is '0 . 0 . 0 . 0'. The 'Firewall Profile' is '<None>' and 'RIP Mode' is 'None'. The 'Enable NAT' checkbox is checked. The 'Number Of DHCP IP Addresses' is '1'. The 'DHCP Mode' is set to 'Disabled' (radio button selected). There is an 'Advanced' button.

5.2.3.2 LAN 2 - VoIP Tab

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

- **RTP Port Number Range:** The AT&T IPTF service requires that the RTP use the port range 16384 to 32767.
 - **16384** entered in the **Port Range (Minimum)** field.
 - **32766** entered in the **Port Range (Maximum)** field, as this field requires even numbers
- To prevent possible issues with network firewalls closing idle RTP channels, it is recommended that **RTP Keepalives** are enabled. Scrolling down to the bottom of the form, enter the following:
 - **Scope:** Select **RTP**
 - **Periodic Timeout:** Enter **30**
 - **Initial keepalives:** Select **Enabled**
- Other parameters on this screen are set to the defaults.
- Click the **OK** button (not shown).

The screenshot displays the configuration interface for VoIP settings. The 'RTP' section is highlighted with a red box. The 'Keepalives' section is also visible, showing the following settings:

- Scope:** RTP
- Periodic timeout:** 30
- Initial keepalives:** Enabled

The 'RTP' section shows the following settings:

- Port Number Range:**
 - Minimum: 16384
 - Maximum: 32766
- Port Number Range (NAT):**
 - Minimum: 49152
 - Maximum: 53246

5.2.3.3 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.
- **Public IP Address:** In the reference configuration the IP Office public address is **192.168.200.26**.
- Set the **Public Port** to **UDP/5060**.
- Click the **OK** button (not shown).

The screenshot shows the 'Network Topology' configuration window. The window has a menu bar at the top with options: System, LAN1, LAN2, DNS, Voicemail, Telephony, Directory Services, System Events, SMTP, SMDR, VCM, VoIP, VoIP Security, and Contact Center. Below the menu bar, there are tabs for 'LAN Settings', 'VoIP', and 'Network Topology'. The 'Network Topology' tab is active. The window contains the following fields and controls:

- Network Topology Discovery** section:
 - STUN Server Address: An empty text input field.
 - STUN Port: A spinner box set to 3478.
 - Firewall/NAT Type: A dropdown menu set to 'Open Internet'.
 - Binding Refresh Time (sec): A spinner box set to 120.
 - Public IP Address: A text input field containing '192 . 168 . 200 . 26'.
 - Buttons: 'Run STUN' and 'Cancel'.
- Public Port** section:
 - UDP: A spinner box set to 5060.
 - TCP: A spinner box set to 0.
 - TLS: A spinner box set to 0.
- At the bottom left, there is a checkbox labeled 'Run STUN on startup' which is currently unchecked.

5.2.4. Voicemail Tab

As described in **Section 3**, Voicemail Pro was used in the reference configuration.

- Set **Voicemail Type** to **Voicemail Line/Pro**.
- Set **Voicemail IP Address** to the IP address of the server hosting voicemail. In the reference configuration, this is **10.64.19.69**.
- 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., **3035559320**) for message retrieval, see **Sections 5.6.3.1** and **5.7**. Note that the **Anonymous** box is checked by default.
- Other parameters on this screen are default. Click the **OK** button (not shown).

The screenshot shows a configuration window with multiple tabs: System, LAN1, LAN2, DNS, Voicemail, Telephony, Directory Services, System Events, SMTP, SMDR, VCM, VoIP, VoIP Security, and Contact Center. The 'Voicemail' tab is active. Key fields include:

- Voicemail Type:** Voicemail Lite/Pro
- Voicemail Destination:** (empty)
- Voicemail IP Address:** 10 . 64 . 19 . 69
- Backup Voicemail IP Address:** 0 . 0 . 0 . 0
- Voicemail Channel Reservation:** Unreserved Channels: 156
- Auto-Attendant:** 0
- Voice Recording:** 0
- Mandatory Voice Recording:** 0
- Announcements:** 0
- Mailbox Access:** 0
- DTMF Breakout:** Reception/Breakout (DTMF 0), Breakout (DTMF 2), Breakout (DTMF 3) (all empty)
- Voicemail Code Complexity:** Enforcement (checked), Minimum length: 0, Complexity (unchecked)
- SIP Settings:** SIP Name: 3035559329, SIP Display Name (Alias): Voicemail, Contact: 3035559329, Anonymous (checked)

5.2.5. 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**.

- Default values are used in the other fields.
- Click the **OK** button (not shown).

The screenshot shows the Avaya configuration interface for the 'Telephony' tab. The 'Analogue Extensions' section includes settings for call sequences, ring types, and dial delay. The 'Companding Law' section is highlighted with a red box and contains two columns: 'Switch' and 'Line'. In the 'Switch' column, 'U-Law' is selected. In the 'Line' column, 'U-Law Line' is selected. Other settings include 'Dial Delay Time (sec)' set to 4, 'Default Name Priority' set to 'Favor Trunk', and 'Inhibit Off-Switch Forward/Transfer' checked.

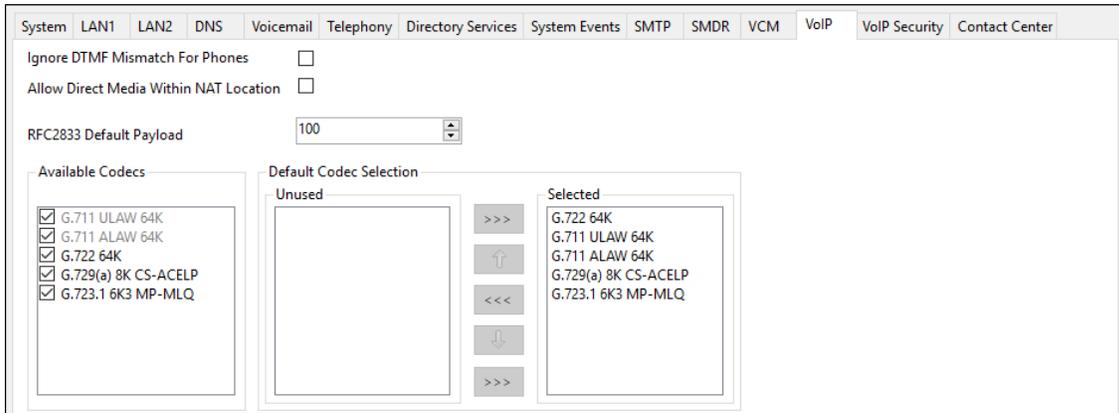
5.2.6. VoIP Tab

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 line or extension.

- 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).

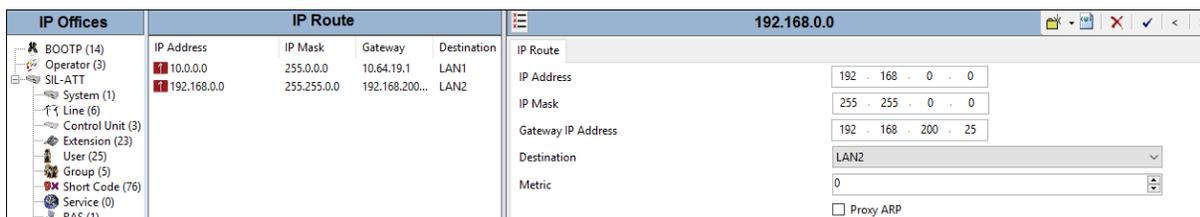
Note - In the reference configuration, the Extension codec lists (see **Section 5.5.2**) also specify G.722 64K, G.711U, G.711A, and G.729(a) (in that order), and the SIP Line (see **Section 5.4.6**) offers G.729(a) and G.711U (in that order). In this manner local Avaya IP Office calls will offer G.711U 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.



5.3. IP Route

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 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 5.2.2** and **5.2.3**.

In order for the Avaya IP Office system to be able to route data to/from the AT&T network, an IP 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 IP route using **Destination** → **LAN2** (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** Error! Reference source not found.. – Error! Reference source not found..

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

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

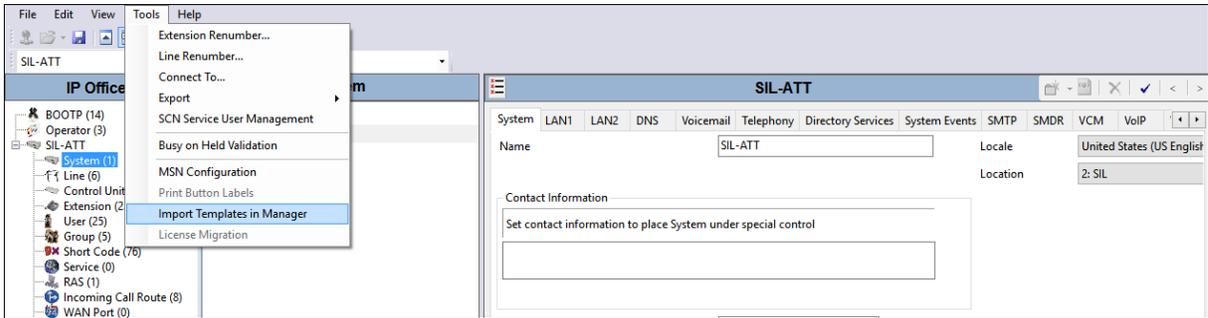
Alternatively, a SIP Line can be created manually. To do so, right-click **Line** in the Navigation Pane and select **New → SIP Line**. Then, follow the steps outlined in **Sections** Error! Reference source not found.. – Error! Reference source not found..

5.4.1.Importing a SIP Line Template

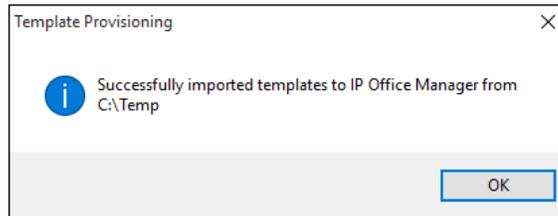
Note – DevConnect generated SIP Line templates are always exported in an XML format. These XML templates do not include sensitive customer specific information and are therefore suitable for distribution. The XML format templates can be used to create SIP trunks on both IP Office Standard Edition (IP500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

1. Copy a previously created template file to a location (e.g., *\temp*) on the same computer where IP Office Manager is installed.

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

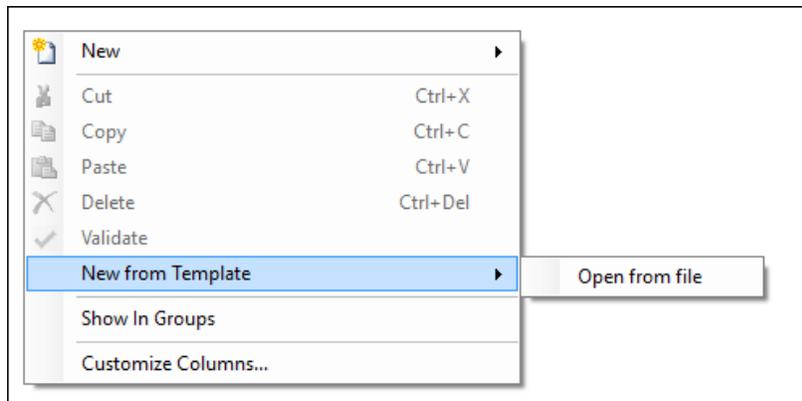


3. A folder browser will open (not shown). Select the directory used in **step 1** to store the template(s) (e.g., `\temp`). In the reference configuration, template file **IPO10TF.xml** was imported. The template files are automatically copied into the IP Office default template location, **C:\Program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

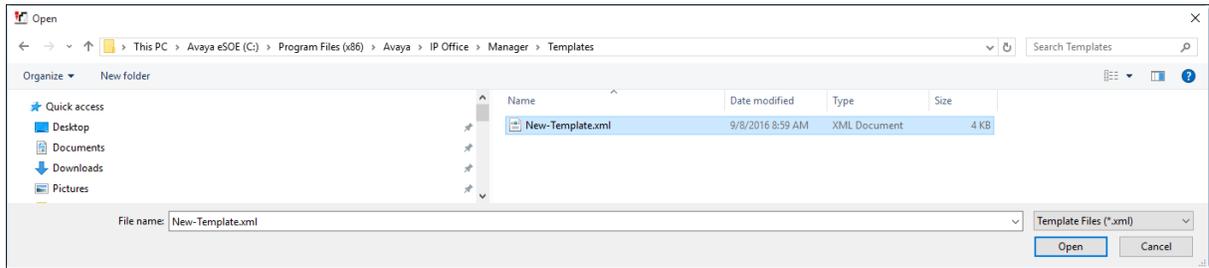


5.4.2. Creating a SIP Trunk from an XML Template

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



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



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

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

Once the SIP Line is created, verify the configuration of the SIP Line with the configuration shown in **Sections** Error! Reference source not found. – Error! Reference source not found..

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 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:** Use the default <blank> to have IP Office send the **ITSP Proxy Address** as the domain name.
- Local Domain Name:** Set to the IP address of the Avaya IP Office LAN2 SIP trunking interface (e.g., **192.168.200.26**).
- 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**).
- Refresh Method:** Set to **Re-Invite**, as AT&T does not support UPDATE
- National Prefix:** Set to the North American country code **1**. See **Section 2.2**.
- International Prefix:** Set to **011**.
- 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 5.2.5**, the **Name Priority** parameter may retain the default **System Default** setting, or can be configured to **Favor Directory**. As shown below, the default **System Default** 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., **192.168.38.69**).
- **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

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

- **Local URI, Contact, and Display Name** fields: Set these fields to **Use Internal Data**.
- Verify **Identity** and **Diversion Header**: Set to the default **None**.
- **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).
- 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 **ARS** configuration (**Section 5.8**).
- **Max Sessions**: In the reference configuration this was set to **10**. This sets the maximum number of simultaneous calls that can use the URI before Avaya IP Office returns busy to any further calls.
- Click **OK**.

The screenshot shows the 'SIP URI' configuration window. At the top, there is a table with columns: URI, Groups, Local URI, Contact, Display Name, Identity, Header, Originator Number, Send Caller ID, Diversion Header, Credential, and Max Calls. Two rows are visible, with the first row selected. Below the table is an 'Edit URI' form with the following fields:

URI	Groups	Local URI	Contact	Display Name	Identity	Header	Originator Number	Send Caller ID	Diversion Header	Credential	Max Calls
1	17 17	<Internal>	<Internal>	<Internal>	None	PAI		Diversion	None	0: <Non...	10
2	17 0	Auto	Auto	Auto	None	PAI		Diversion	None	0: <Non...	10

Edit URI

Local URI: Use Internal Data (dropdown)

Contact: Use Internal Data (dropdown)

Display Name: Use Internal Data (dropdown)

Identity: None (dropdown)

Header: P Asserted ID (dropdown)

Forwarding And Twinning

Originator Number: (text field)

Send Caller ID: Diversion Header (dropdown)

Diversion Header: None (dropdown)

Registration: 0: <None> (dropdown)

Incoming Group: 17 (dropdown)

Outgoing Group: 17 (dropdown)

Max Sessions: 10 (spin box)

Buttons: Add..., Remove, Edit..., OK, Cancel

In the sample configuration, the single SIP URI shown above was sufficient to allow incoming calls for AT&T DID numbers destined for specific IP Office users or IP Office hunt groups. The calls are accepted by IP Office since the incoming number will match the SIP Name configured for the user or group that is the destination for the call. URI 2 will match on any number not associated with users or groups, such as a DID number routed directly to

voicemail or DID used for Mobile Call Control. DID numbers that IP Office should admit can be entered specifically, such as 3035559320, into the **Local URI** and **Contact** fields instead of “Use Internal Data”. To allow IP Office to admit any number, **Auto** can be entered into the **Local URI** and **Contact** fields as shown below. This URI entry will not be used for outbound dialing; therefore, an unused number is specified for the **Outgoing Group**.

URI	Groups	Local URI	Contact	Display Name	Identity	Header	Originator Number	Send Caller ID	Diversion
1	17 17	<Internal>	<Internal>	<Internal>	None	PAI		Diversion	None
2	17 0	Auto	Auto	Auto	None	PAI		Diversion	None

Edit URI	
Local URI	Auto
Contact	Auto
Display Name	Auto
Identity	None
Header	P Asserted ID
Forwarding And Twinning	
Originator Number	
Send Caller ID	Diversion Header
Diversion Header	None
Registration	0: <None>
Incoming Group	17
Outgoing Group	0
Max Sessions	10

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.

- 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**).

The screenshot displays the 'VoIP' configuration tab for an extension. The 'Codec Selection' section features a 'Custom' dropdown menu and two lists: 'Unused' (G.711 ALAW 64K, G.722 64K, G.723.1 6K3 MP-MLQ) and 'Selected' (G.729(a) 8K CS-ACELP, G.711 ULAW 64K). Navigation buttons (>>>, <<<, <-, >=) are positioned between the lists. On the right, several checkboxes are visible: 'VoIP Silence Suppressor' (checked), 'Local Hold Music' (unchecked), 'Re-invite Supported' (checked), 'Codec Lockdown' (unchecked), 'Allow Direct Media Path' (unchecked), 'Force direct media with phones' (unchecked), 'PRACK/100rel Supported' (unchecked), and 'G.711 Fax ECAN' (unchecked). At the bottom, three dropdown menus are shown: 'Fax Transport Support' (T38), 'DTMF Support' (RFC2833), and 'Media Security' (Disabled). Red boxes highlight the 'VoIP Silence Suppressor', 'Re-invite Supported', and 'Fax Transport Support' options.

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).

The screenshot shows the 'T38 Fax' configuration tab. The 'T38 Fax Version' is set to '0'. The 'Disable T30 ECM' checkbox is unchecked. Other settings include Transport: UDPTL, Redundancy: Low Speed 0, High Speed 0, TCF Method: Trans TCF, Max Bit Rate: 14400, EFlag Start Timer: 2600, EFlag Stop Timer: 2300, Tx Network Timeout: 150. A 'Use Default Values' checkbox is at the bottom left.

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**). 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.

- Select **Indicate HOLD**.
- 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.

SIP Line Transport SIP URI VoIP T38 Fax SIP Credentials SIP Advanced Engineering

Addressing

Association Method: By Source IP address

Call Routing Method: Request URI

Suppress DNS SRV Lookups:

Identity

Use "phone-context":

Add user=phone:

Use + for International:

Use PAI for Privacy:

Use Domain for PAI:

Swap From and PAI/Diversion:

Caller ID from From header:

Send From In Clear:

Cache Auth Credentials:

User-Agent and Server Headers:

Send Location Info: Never

Media

Allow Empty INVITE:

Send Empty re-INVITE:

Allow To Tag Change:

P-Early-Media Support: None

Send SilenceSupp=Off:

Force Early Direct Media:

Media Connection Preservation: Disabled

Indicate HOLD:

Call Control

Call Initiation Timeout (s): 4

Call Queuing Timeout (mins): 5

Service Busy Response: 486 - Busy Here

on No User Responding Send: 408-Request Timeout

Action on CAC Location Limit: Allow Voicemail

Suppress Q.850 Reason Header:

Emulate NOTIFY for REFER:

No REFER if using Diversion:

5.5. Users, Extensions, and Hunt Groups

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

5.5.1. User 321 (Digital)

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

The screenshot displays the configuration page for User 321. The interface includes a navigation bar at the top with tabs for various settings: User, Voicemail, DND, Short Codes, Source Numbers, Telephony, Forwarding, Dial In, Voice Recording, Button Programming, Menu Programming, Mobility, and Group. The main configuration area is divided into two sections: a top section for user identification and status, and a bottom section for profile settings. The top section includes fields for Name (Avaya9508), Password (masked with dots), Confirm Password (masked with dots), Unique Identity, Conference PIN, Confirm Audio Conference PIN, Account Status (set to Enabled), Full Name, Extension (321), Email Address, Locale, Priority (5), and System Phone Rights (None). The bottom section, titled 'Profile', shows a dropdown menu set to 'Basic User' and a list of checkboxes for various features: Receptionist, Enable Softphone, Enable one-X Portal Services, Enable one-X TeleCommuter, Enable Remote Worker, Enable Communicator, Enable Mobile VoIP Client, Send Mobility Email, and Web Collaboration.

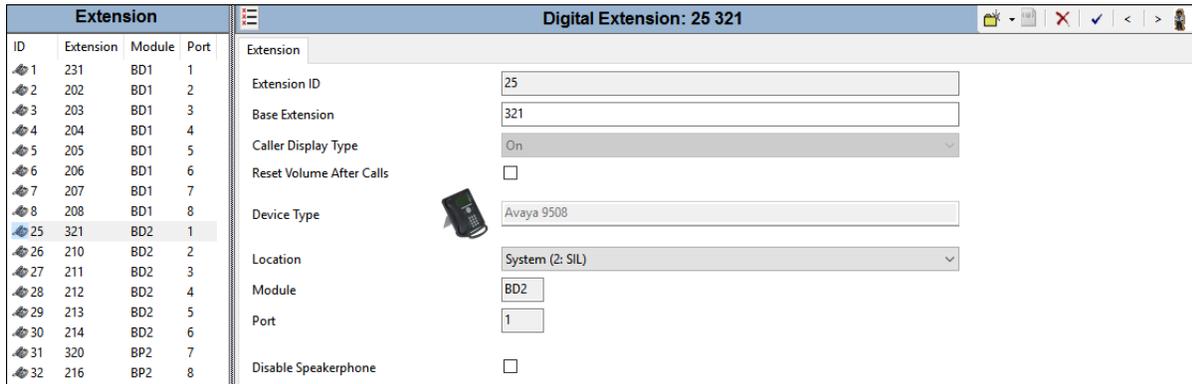
The following screen shows the **SIP** tab for User 321. The **SIP Name** and **Contact** parameters are configured with the DID number of the user, 303-555-9321. These parameters configure the user part of the SIP URI in the From header for outgoing SIP trunk calls, and allow matching of the SIP URI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

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

From **Figure 1**, note that user 321 will use the Mobile Twinning feature. The following screen shows the **Mobility** tab for user 321. The **Mobility Features** and **Mobile Twinning** boxes are checked. The **Twinned Mobile Number** field is configured with the number to dial to reach the twinned mobile telephone, in this case 913055582177. Other options can be set according to customer requirements.

Voicemail	DND	Short Codes	Source Numbers	Telephony	Forwarding	Dial In	Voice Recording	Button Programming	Menu Programming	Mobility
<input type="checkbox"/> Internal Twinning Twinned Handset: <None> Maximum Number of Calls: 1 <input type="checkbox"/> Twin Bridge Appearances <input type="checkbox"/> Twin Coverage Appearances <input type="checkbox"/> Twin Line Appearances <input checked="" type="checkbox"/> Mobility Features <input type="checkbox"/> Mobile Twinning Twinned Mobile Number (including dial access code): 913035552177 Twinning Time Profile: <None> Mobile Dial Delay (sec): 0 Mobile Answer Guard (sec): 0 <input type="checkbox"/> Hunt group calls eligible for mobile twinning <input type="checkbox"/> Forwarded calls eligible for mobile twinning <input type="checkbox"/> Twin When Logged Out <input type="checkbox"/> one-X Mobile Client <input checked="" type="checkbox"/> Mobile Call Control <input checked="" type="checkbox"/> Mobile Callback										

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



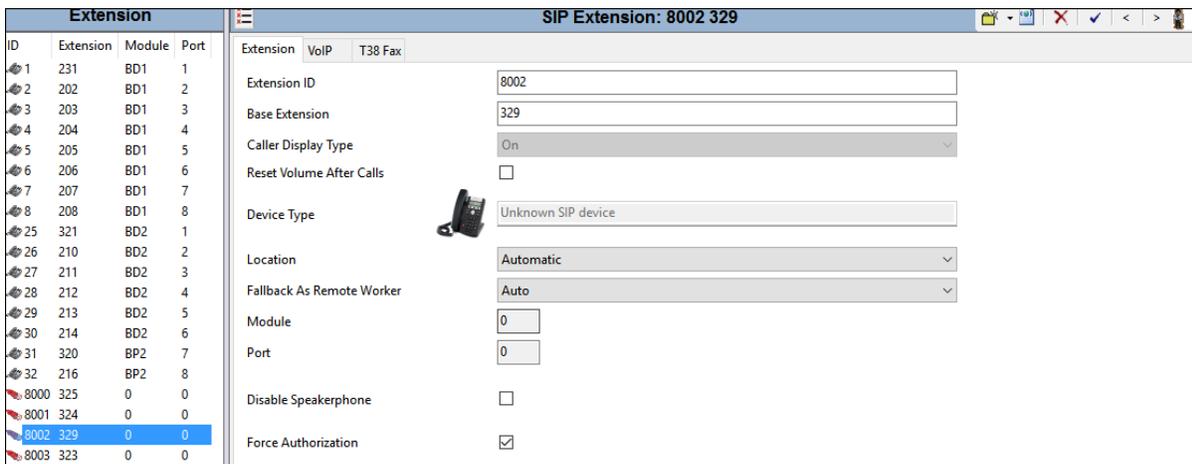
ID	Extension	Module	Port
1	231	BD1	1
2	202	BD1	2
3	203	BD1	3
4	204	BD1	4
5	205	BD1	5
6	206	BD1	6
7	207	BD1	7
8	208	BD1	8
25	321	BD2	1
26	210	BD2	2
27	211	BD2	3
28	212	BD2	4
29	213	BD2	5
30	214	BD2	6
31	320	BP2	7
32	216	BP2	8

Digital Extension: 25 321

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

5.5.2. User 329 (Avaya Communicator for Windows)

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



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

SIP Extension: 8002 329

Extension ID: 8002
 Base Extension: 329
 Caller Display Type: On
 Reset Volume After Calls:
 Device Type: Unknown SIP device
 Location: Automatic
 Fallback As Remote Worker: Auto
 Module: 0
 Port: 0
 Disable Speakerphone:
 Force Authorization:

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

Extension				SIP Extension: 8002 329		
ID	Extension	Module	Port	Extension	VoIP	T38 Fax
1	231	BD1	1	IP Address	0 . 0 . 0 . 0	
2	202	BD1	2	Codec Selection	System Default	
3	203	BD1	3	Reserve License	Reserve Avaya IP endpoint license	
4	204	BD1	4	Fax Transport Support	None	
5	205	BD1	5	TDM->IP Gain	Default	
6	206	BD1	6	IP->TDM Gain	Default	
7	207	BD1	7	DTMF Support	RFC2833	
8	208	BD1	8	3rd Party Auto Answer	None	
25	321	BD2	1	Media Security	Same as System (Disabled)	
26	210	BD2	2			
27	211	BD2	3			
28	212	BD2	4			
29	213	BD2	5			
30	214	BD2	6			
31	320	BP2	7			
32	216	BP2	8			
8000	325	0	0			
8001	324	0	0			
8002	329	0	0			
8003	323	0	0			
8004	236	0	0			
8005	238	0	0			
8006	322	0	0			

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. In the sample configuration, the **Profile** is set to **Power User**, with **Enable Softphone**, and **Enable Communicator** checked.

User	Voicemail	DND	Short Codes	Source Numbers	Telephony	Forwarding	Dial In	Voice Recording	Button Programming	Menu Programming
Name	Mobile									
Password	••••••••									
Confirm Password	••••••••									
Unique Identity										
Conference PIN										
Confirm Audio Conference PIN										
Account Status	Enabled									
Full Name	onex mobile									
Extension	329									
Email Address										
Locale										
Priority	5									
System Phone Rights	None									
Profile	Power User									
	<input type="checkbox"/> Receptionist									
	<input checked="" type="checkbox"/> Enable Softphone									
	<input checked="" type="checkbox"/> Enable one-X Portal Services									
	<input checked="" type="checkbox"/> Enable one-X TeleCommuter									
	<input checked="" type="checkbox"/> Enable Remote Worker									
	<input checked="" type="checkbox"/> Enable Communicator									
	<input checked="" type="checkbox"/> Enable Mobile VoIP Client									
	<input type="checkbox"/> Send Mobility Email									
	<input type="checkbox"/> Web Collaboration									

Select the **Telephony** tab. Then select the **Supervisor Settings** tab as shown below. The **Login Code** will be used by the Avaya Communicator for Windows user as the login password.

The screenshot shows the 'Supervisor Settings' tab in the 'Telephony' section. The 'Login Code' and 'Confirm Login Code' fields are both set to four dots. The 'Login Idle Period (sec)' field is empty. The 'Monitor Group' and 'Coverage Group' are set to '<None>'. The 'Status on No-Answer' is set to 'Logged On (No change)'. The 'Reset Longest Idle Time' section has 'All Calls' selected. The following checkboxes are present: Force Login, Force Account Code, Force Authorization Code, Incoming Call Bar, Outgoing Call Bar, Inhibit Off-Switch Forward/Transfer, Can Intrude, Cannot Be Intruded (checked), Can Trace Calls, and Deny Auto Intercom Calls.

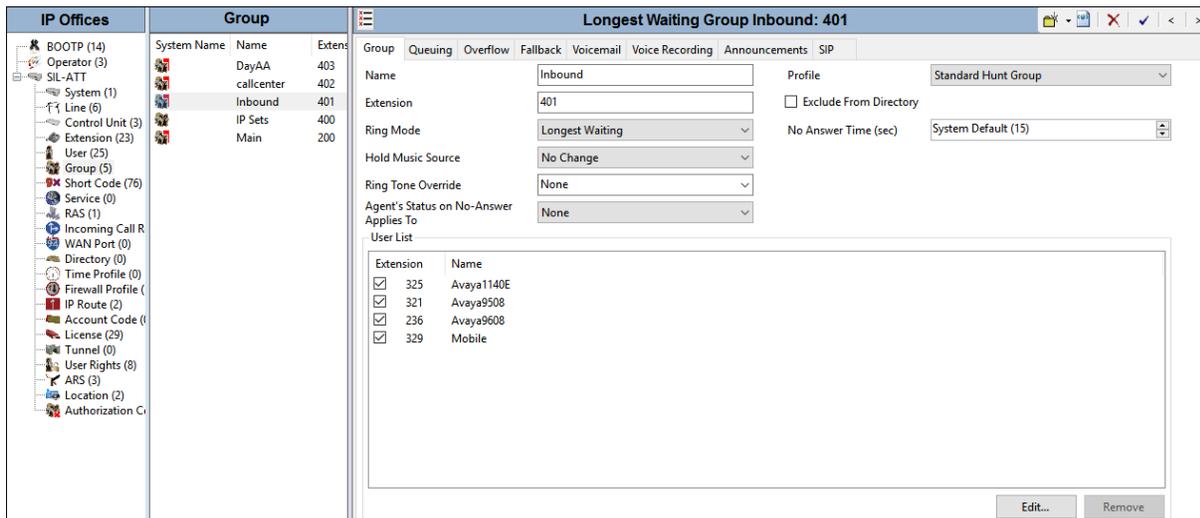
Like other users previously illustrated, the **SIP** tab for the user with extension 321 is configured with a **SIP Name** and **Contact** specifying the user's AT&T DID number.

The screenshot shows the 'SIP' tab configuration. The 'SIP Name' field contains '469554881'. The 'SIP Display Name (Alias)' field contains 'Mobile'. The 'Contact' field contains '469554881'. The 'Anonymous' checkbox is unchecked.

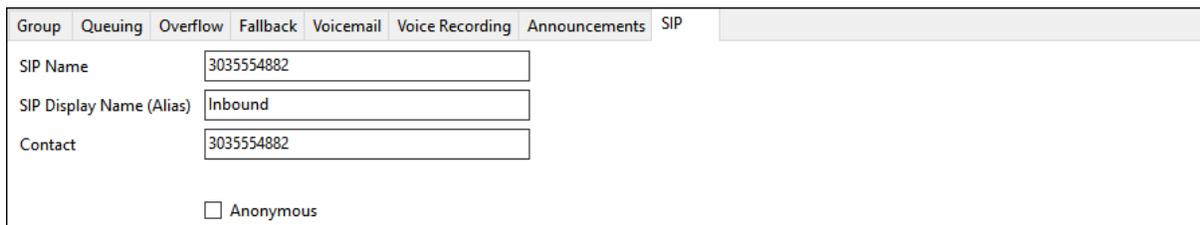
5.5.3.Hunt Groups

During the verification of these Application Notes, users could also receive incoming calls as members of a hunt group. To configure a new hunt group, right-click **Group** from the Navigation pane, and select **New**. To view or edit an existing hunt group, select **HuntGroup** from the Navigation pane, and the appropriate hunt group from the Group pane.

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



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



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 Dialing via Automatic Route Selection (ARS access)

In the screen shown below, the Short Code **9N** is illustrated. 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**
- The **Line Group ID** parameter is set to **60: ATT**, which directs the call to ARS (see **Section 5.8**).
- Click the **OK** button (not shown).

Short Code	
Code	9N
Feature	Dial
Telephone Number	N
Line Group ID	60: ATT
Locale	
Force Account Code	<input type="checkbox"/>
Force Authorization Code	<input type="checkbox"/>

5.6.2. Privacy Dialing

Optionally, a Short Code may be added to access the SIP Line anonymously. In the screen shown below, the Short Code ***67N** is illustrated. This code is similar to the **9N** code. The **Telephone Number** field begins with the letter **W**, which means “withhold the outgoing calling line identification”.

Note - When a user dials ***67** 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 ***67N**
- The **Feature** parameter is set to **Dial**
- The **Telephone Number** parameter is set to **WN**.
- The **Line Group ID** parameter is set to **60: ATT**, which directs the call to ARS (see **Section 5.8**).
- Click the **OK** button (not shown).

Short Code	
Code	*67N
Feature	Dial
Telephone Number	WN
Line Group ID	60: ATT
Locale	
Force Account Code	<input type="checkbox"/>
Force Authorization Code	<input type="checkbox"/>

5.6.3.Feature Dialing

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

5.6.3.1 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**

Short Code	
Code	*17
Feature	Voicemail Collect
Telephone Number	?U
Line Group ID	0
Locale	
Force Account Code	<input type="checkbox"/>
Force Authorization Code	<input type="checkbox"/>

5.6.3.2 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.

5.6.3.2.1 Mobile Callback

The following screen illustrates the **Code FNE33Callback** 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). This code 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 303-555-2177, defined in the **User → Mobility → Twinned Mobile Number** field of associated Avaya IP Office station 321 (see **Section 5.5.1**), calls the DID associated with the Mobile Call Back Short Code **FNE33Callback** (e.g., **3035559326**, 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 1-303-555-2177 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 **913035552177** was defined as the Twinning number for the digital set. Therefore, the inbound calling number must match **3035552177**.

- **Code = FNE33Callback**
- **Feature = FNE Service**
- **Telephone Number = 33**
- **Line Group = 0**

Short Code	
Code	FNE33Callback
Feature	FNE Service
Telephone Number	33
Line Group ID	0
Locale	
Force Account Code	<input type="checkbox"/>
Force Authorization Code	<input type="checkbox"/>

5.6.3.2.2 Mobile Call Control

The following screen illustrates another Mobility Short Code. In this case, the **Code FNE31** 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 **FNE31** 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 303-555-2177, defined in the **User → Mobility → Twinned Mobile Number** field of associated station 321 (see **Section 5.5.1**), calls the DID associated with the Mobile Call Control Short Code FNE31 (e.g., **3035559327**, 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 321).

- **Code = FNE31**
- **Feature = FNE Service**
- **Telephone Number = 31**
- **Line Group = 0**

Short Code	
Code	FNE31
Feature	FNE Service
Telephone Number	31
Line Group ID	0
Locale	
Force Account Code	<input type="checkbox"/>
Force Authorization Code	<input type="checkbox"/>

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, the AT&T IPFR-E service delivered ten DNIS digits in the SIP Invite R-URI. Therefore, incoming calls to Avaya IP Office will match on the ten digit inbound AT&T DNIS string (e.g., **3035559321**). The AT&T IPFR-E service can also be configured to deliver seven DNIS digits. Verify the digits being delivered by AT&T.

In the screen shown below, the incoming call route for **Incoming Number** → **3035559321** 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**.

The screenshot shows the Avaya IP Office configuration interface. On the left is a navigation tree with categories like BOOTP, Operator, SIL-ATT, System, Line, Control Unit, Extension, User, Group, Short Code, Service, RAS, Incoming Call Route, WAN Port, Directory, Time Profile, Firewall Profile, IP Route, Account Code, License, and Tunnel. The main area is titled 'Incoming Call Route' and shows a table of configurations for Line Group ID 17. The selected entry is for Incoming Number 3035559321, with Destination 321 Avaya9508. To the right of the table is a configuration panel for the selected route, showing fields for Bearer Capability (Any Voice), Line Group ID (17), Incoming Number (3035559321), Incoming Sub Address, Incoming CLI, Locale, Priority (1 - Low), Tag, Hold Music Source (System Source), and Ring Tone Override (None).

Line Group ID	Incoming Number	Destination
17	3035559320	VoiceMail
17	3035559xxx	#
17	.	.
17	3035559321	321 Avaya9508
17	4695554882	401 Inbound
17	3035559327	FNE31
17	3035559326	FNE33Callback
17	3035559328	VM:DayAA

Select the **Destinations** tab. From the **Destination** drop-down menu, select the extension to receive the call when AT&T delivers DNIS digits **3035559321**. In the reference configuration DNIS digits **3035559321** is associated with user **321** (the 9508 digital telephone).

The screenshot shows the 'Destinations' tab for the configuration of Incoming Call Route 17 3035559321. It features a table with columns for TimeProfile, Destination, and Fallback Extension. The 'Default Value' row shows 'Default Value' for TimeProfile, '321 Avaya9508' for Destination, and a dropdown arrow for Fallback Extension.

TimeProfile	Destination	Fallback Extension
Default Value	321 Avaya9508	

Repeat this process to route all AT&T DNIS numbers to additional telephone, as well as other Avaya IP Office destinations (Hunt Group (**4695554882**), Voicemail (**3035559320**), Short Codes (**3035559326**), etc.). For example:

TimeProfile	Destination	Fallback Extension
Default Value	401 Inbound	

TimeProfile	Destination	Fallback Extension
Default Value	VoiceMail	

TimeProfile	Destination	Fallback Extension
Default Value	FNE33Callback	

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 5.6.1**, 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., **60: ATT**).
- 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 **ATT** (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 *unchecked*, calls are routed to the ARS route specified in the **Out of Service Route** parameter (in the reference configuration, a second ARS route, **61: ATTbackup**, was defined).

- **Code = 1xxxxxxxxx** This means any dialed string starting with a 1, and 11 digits total will be routed to the specified Line Group.
- **Telephone Number = .**
- **Feature = Dial**
- **Line Group ID = 17** (SIP Line 17).

ARS configuration interface showing the following settings:

- ARS Route ID: 60
- Route Name: ATT
- Dial Delay Time: System Default (4)
- Description: ATT IPFR
- In Service: (highlighted with a red box)
- Out of Service Route: 61: ATTbackup (highlighted with a red box)
- Time Profile: <None>
- Out of Hours Route: <None>

Code	Telephone Number	Feature	Line Group ID
*7N;	.	Dial	17
*9N;	.	Dial	17
0N;	.	Dial	17
11	911	Dial Emergency	17
1xxxxxxxxx	.	Dial	17
303xxxxxxxxx	.	Dial	17
411	.	Dial	17

Alternate Route Priority Level: 3

Alternate Route Wait Time: 30

Alternate Route: 61: ATTbackup (highlighted with a red box)

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., **61:ATTbackup**).

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 ***67N** to access the SIP Line.

Certain Avaya endpoints can also request privacy for a specific call, without dialing a unique Short Code, by accessing the telephone user interface screen **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 header.

5.10. SIP Options

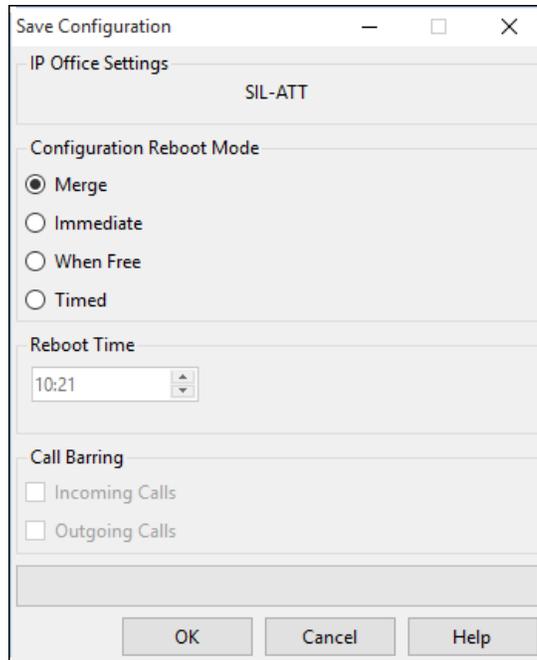
Avaya IP Office periodically checks the health of the SIP Line by sending a SIP OPTIONS message, based on the **Binding Refresh Time** (see **Section 5.2.3**). In the reference configuration, the Binding Refresh Time is set to 120 seconds.

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 10 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 10

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

7.2.1. System Status Application

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

The screenshot shows a 'Logon' dialog box with the following fields and options:

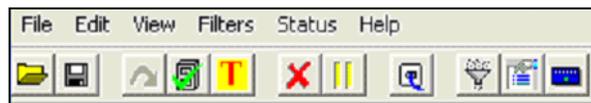
- Control Unit IP Address:** 10.64.19.70
- Services Base TCP Port:** 50804
- Local IP Address:** Automatic
- User Name:** (empty field)
- Password:** (empty field)
- Auto reconnect
- Secure connection
- Logon** button

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 [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 to Avaya IP Office SIP telephone 500.

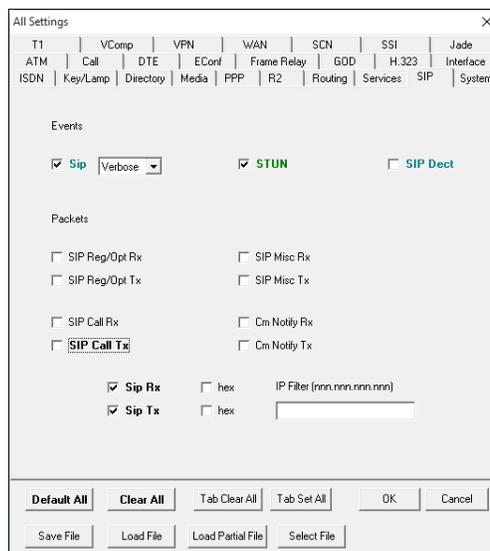
```

13:09:55 697212996ms Sip: Completed f1797be8 ... removing Dialog of CallId 348d1ab162768ff127313e7761d34ea2000aac0@192.168.38.69 and State: SIPDialog::FINAL(28)
13:09:55 697212996ms Sip: (f1797be8) SetUnIncTransactionCondition to Unint_None
13:09:55 697212996ms Sip: SIPDialog f1797be8 deleted, dialogs 0 txn_keys 0
13:09:55 697212997ms Sip: sip_indicateTimeOut txn_keys 0
13:09:56 697213179ms SIP Rx: UDP 192.168.37.149:5060 -> 192.168.200.26:5060
INVITE sip:3035559321@192.168.200.26:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.37.149:5060;branch=z9hG4bKlp24k22018kat7h59nq0.1
From: "AVAYA" <sip:3035552177@192.168.37.149:user=phone>;tag=559904314-1477681796536-
To: "User 3035559321" <sip:3035559321@192.168.200.26:user=phone>
Call-ID: BW190956536281016-1891051343011
CSeq: 106523357 INVITE
Contact: <sip:192.168.37.149:5060;transport=udp>
P-Asserted-Identity: "AVAYA" <sip:3035552177@192.168.37.149:user=phone>
Privacy: none
Supported: timer
Allow: INVITE,ACK,CANCEL,BYE,REFER,INFO,NOTIFY,PRACK,OPTIONS
Recv-Info: x-broadworks-client-session-info
Accept: application/dtmf-relay, application/media_control+xml, application/sdp, multipart/mixed
Min-SE: 60
Session-Expires: 1800;refresher=uac
Max-Forwards: 68
Content-Length: 243
Content-Disposition: session; handling-required
Content-Type: application/sdp

v=0
o=BroadWorks 24011035 1 IN IP4 192.168.37.158
s=
c=IN IP4 192.168.37.158
t=0 0
m=audio 26284 RTP/AVP 18 0 100
a=rtpmap:18 G729/8000
a=rtpmap:0 PCMG/8000
a=rtpmap:100 telephone-event/8000
a=fmtp:100 0-15
a=sendrecv
a=msapline:30
13:09:56 697213183ms Sip: Association found trunk: SIP Line (17)

```

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.



The screenshot shows the 'All Settings' dialog box with the 'SIP' tab selected. The 'Events' section includes checkboxes for 'Sip' (checked, with a 'Verbose' dropdown), 'STUN' (checked), and 'SIP Dect' (unchecked). The 'Packets' section includes checkboxes for 'SIP Reg/Opt Rx', 'SIP Reg/Opt Tx', 'SIP Call Rx', 'SIP Call Tx' (checked), 'SIP Misc Rx', 'SIP Misc Tx', 'Cm Notify Rx', and 'Cm Notify Tx'. At the bottom, there are checkboxes for 'Sip Rx' (checked), 'Sip Tx' (checked), and 'hex' (unchecked), along with an 'IP Filter' field.

8. Conclusion

As illustrated in these Application Notes, Avaya IP Office R10 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 R10 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] *IP Office™ Platform 10.0, Deploying Avaya IP Office™ Platform Servers as Virtual Machines*, Document Number 15-601011, Issue 04d, July 2016
- [2] *IP Office™ Platform 10.0, Deploying Avaya IP Office™ Platform IP500 V2*, Document Number 15-601042, Issue 31h, Aug 2016
- [3] *Administering Avaya IP Office™ Platform with Manager*, Release 10.0, August 2016
- [4] Additional Avaya IP Office information can be found at:
<http://marketingtools.avaya.com/knowledgebase/>

AT&T IPFR-EF Service:

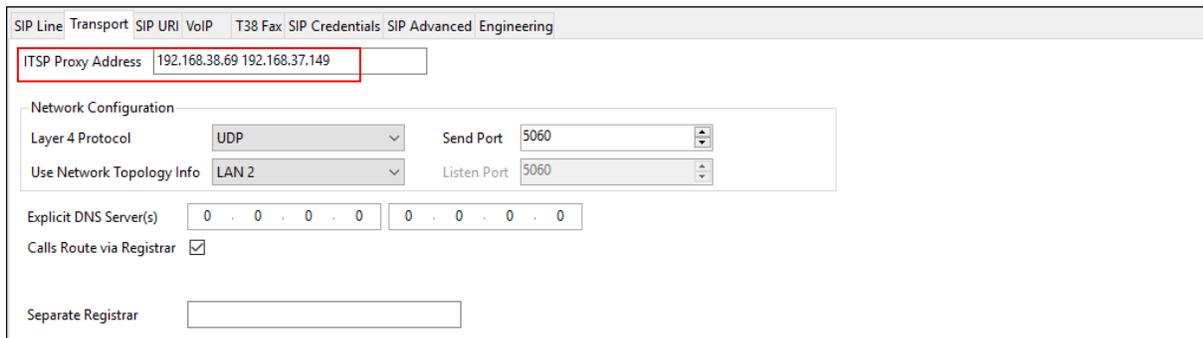
- [5] Information on the AT&T IP Flexible Reach service can be found here:
<http://www.business.att.com/enterprise/Service/voice-services/null/sip-trunking/>

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 **192.168.38.69** (primary) and **192.168.37.149** (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.



SIP Line: Transport SIP URI VoIP T38 Fax SIP Credentials SIP Advanced Engineering

ITSP Proxy Address 192.168.38.69 192.168.37.149

Network Configuration

Layer 4 Protocol UDP Send Port 5060

Use Network Topology Info LAN 2 Listen Port 5060

Explicit DNS Server(s) 0 . 0 . 0 . 0 0 . 0 . 0 . 0

Calls Route via Registrar

Separate Registrar

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 (192.168.38.69). If the OPTIONS fail for the primary connection, then Avaya IP Office will use the second address (192.168.37.149) 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 **192.168.38.69w1 192.168.37.149w1**, will result in a “round-robin” behavior, while a setting of **192.168.38.69w2 192.168.37.149w1**, will result in a “two-to-one” behavior.

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

©2016 Avaya Inc. All Rights Reserved.

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

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