Avaya Solution & Interoperability Test Lab

Application Notes for Avaya IP Office Release 11.1 with AT&T IP Flexible Reach - Enhanced Features – Issue 1.0

Abstract

These Application Notes describe the steps for configuring Avaya IP Office Release 11.1 with the AT&T IP Flexible Reach - Enhanced Features service using AVPN or ADI/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 11.1.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as the observations noted in **Section 2.2**, to ensure that their own use cases are adequately covered by this scope and results.

AT&T is a member of the Avaya DevConnect Service Provider program. Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

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Table of Contents

1. Introduction 4
2. General Test Approach and Test Results 5
   2.1. Interoperability Compliance Testing 6
   2.2. Test Results 7
   2.3. Support 8
3. Reference Configuration 8
   3.1. Illustrative Configuration Information 10
   3.2. Call Flows 11
      3.2.1. Inbound 11
      3.2.2. Outbound 12
      3.2.3. Call Forward 13
      3.2.4. Coverage to Voicemail 14
4. Equipment and Software Validated 15
5. Avaya IP Office Primary Configuration 16
   5.1. Licensing 18
   5.2. System Settings 19
      5.2.1. LAN 1 Tab 19
      5.2.2. LAN 2 Tab 21
      5.2.3. Voicemail Settings 23
      5.2.4. System Telephony Settings 24
      5.2.5. System VoIP Settings 25
      5.2.6. System VoIP Security 26
   5.3. IP Route 26
   5.4. SIP Line 27
      5.4.1. Importing a SIP Line Template 28
      5.4.2. SIP Line – SIP Line tab 29
      5.4.3. SIP Line - Transport tab 30
      5.4.4. SIP Line – Call Details tab 31
      5.4.5. SIP Line - VoIP tab 32
      5.4.6. SIP Line – SIP Advanced Tab 33
   5.5. IP Office Line 34
   5.6. Short Codes 35
   5.7. Users, Extensions, and Hunt Groups 36
      5.7.1. SIP User 36
      5.7.2. Hunt Groups 38
   5.8. Incoming Call Routes 39
   5.9. Automatic Route Selection (ARS) 40
   5.10. Outbound Calling Party Information 42
      5.10.1. Short Code - User 42
      5.10.2. Short Code – ARS 43
   5.11. Saving Configuration Changes to Avaya IP Office 44
6. Avaya IP Office Expansion Configuration 45
   6.1. Expansion System - Physical Hardware 45
   6.2. LAN Settings 46

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6.3. Expansion System - IP Route 46
6.4. Expansion System - IP Office Line 47
6.5. Expansion System - Short Codes 48
6.6. Expansion System - Automatic Route Selection – ARS 49
6.7. Save IP Office Expansion System Configuration 49
7. AT&T IP Flexible Reach – Enhanced Features Configuration 50
8. Verification Steps 50
8.1. AT&T IP Flexible Reach – Enhanced Features 50
8.2. Avaya IP Office Release 11.1 51
8.2.1. System Status Application 51
8.2.2. System Monitor Application 52
9. Conclusion 53
10. Additional References 53
11. Addendum 1 – Multiple AT&T Border Elements 54

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1. Introduction

These Application Notes describe the steps for configuring Avaya IP Office Release 11.1 with the AT&T IP Flexible Reach - Enhanced Features service using AT&T Virtual Private Network (AVPN) or AT&T Dedicated Internet Service (ADI/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 AT&T AVPN[1] or ADI/PNT[2] transport services.

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

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

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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 Release 11.1 (see **Section 3.2** for call flow examples).

The test environment described in these Application Notes consisted of:

- A simulated enterprise with Avaya IP Office Release 11.1, Avaya SIP, H.323 and Analog telephones, as well as a fax machine emulator (Ventafax).
- An IPFR-EF production circuit, to which the simulated enterprise was connected via AVPN transport.

Avaya recommends our customers implement Avaya solutions using appropriate security and encryption capabilities enabled by our products. The testing referenced in this DevConnect Application Note included the enablement of supported encryption capabilities in the Avaya products. Readers should consult the appropriate Avaya product documentation for further information regarding security and encryption capabilities supported by those Avaya products.

Support for these security and encryption capabilities in any non-Avaya solution component is the responsibility of each individual vendor. Readers should consult the appropriate vendor-supplied product documentation for more information regarding those products.

For the testing associated with this Application Note, the interface between Avaya systems and the AT&T Flexible Reach service did not include use of any specific encryption features as requested by AT&T.

Encryption (TLS/SRTP) was used internal to the enterprise between Avaya products wherever possible.

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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.711ULAW.
- 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.729A 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.

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2.2. Test Results

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

1. **Avaya IP Office only supports a packet size (ptime) of 20 msecs –** AT&T recommends that for maximum customer bandwidth utilization, a ptime value of 30 should be specified. Avaya IP Office only supports a ptime value of 20 msecs, and therefore does not specify a ptime value in the SIP SDP (in either requests or responses). No issues were observed during testing as a result.

2. **Fax Support**
   - Inbound T.38 fax calls fail when the CPE fax machine is a Super G3 (SG3) fax device. Avaya IP Office may not send a re-INVITE to renegotiate to T.38 for inbound fax calls in cases where both the PSTN and CPE fax devices support Super Group 3 (SG3) speeds. If the SG3 connection is fully established at the beginning of the call, the CPE fax device may not negotiate down to G3 speeds and IP Office will not detect the G3 tones necessary for T.38.
   - G.711 fax calls behaved unpredictably during the compliance test. Disabling Error Correction Mode on the CPE fax device seemed to modestly improve the G711 fax success rate.
   - T.38 should be the preferred method for fax. If the CPE device supports SG3 speeds (33600 bps), a good practice is to disable SG3 on the CPE fax device, if possible.

3. **SIP OPTIONS** – AT&T responded with "405 Method Not Allowed" to the SIP OPTIONS messages sent from the Avaya IP Office to the network used to monitor the status of the SIP trunk. The IP Office considers this as an acceptable response to keep the trunk in service.

4. **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.

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2.3. Support

For more information on the AT&T IP Flexible Reach service visit: https://www.business.att.com/products/sip-trunking.html. 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

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 the Avaya IP Office Server Edition platform. This solution is extensible to the IP 500 V2 platform as well.
- Avaya endpoints are represented by Avaya 9608 H.323 Deskphones, Avaya J169 SIP Deskphones, an Avaya 1140E SIP Deskphones, Avaya 9508 Digital Telephones, as well as Avaya IX™ Workplace for Windows (SIP) softphones. Fax endpoints are represented by PCs running Ventafax emulation software connected by modem to an Avaya IP Office analog port.
- Voicemail Pro (running on the Primary server) provided the voice messaging capabilities in the reference configuration. This solution is extensible to the Avaya IP Office embedded voice mail as well.
- 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.

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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.

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3.1. Illustrative Configuration Information

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

Note – The Avaya IP Office LAN 2 interface is used on the SIP trunk to AT&T (see **Section 5.2.2**) and communicates with the 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	
Primary Server, LAN2 interface	192.168.200.26
Primary Server, LAN1 interface	10.64.19.170
Expansion System, LAN1 Interface	10.5.5.180
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

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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

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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

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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 5.4.4**).

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, network based 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

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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 Voicemail Pro.

Figure 5: Coverage to Avaya IP Office Voicemail

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4. Equipment and Software Validated

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

Avaya IP Telephony Solution Components	
Equipment	**Software**
Avaya IP Office Server Edition - Avaya IP Office Voicemail Pro	Release 11.1.0.0.0 Build 237 Release 11.1.0.0.0 Build 234
Avaya IP Office 500 V2 Expansion System	Release 11.1.0.0.0 Build 237
Avaya IP Office Manager	Release 11.1.0.0.0 Build 237
Avaya 96x1 Series IP Deskphone (H.323)	Release 6.8304
Avaya 1140E IP Deskphone (SIP)	Release 04.04.23.00
Avaya J169 IP Deskphone (SIP)	Release 4.0.4.0.10
Avaya IX Workplace Client for Windows	Release 3.8.5.41.23
Avaya 9508 Digital Deskphone	Release 0.60
Fax device (Ventafax)	Release 7.10

Table 2: Equipment and Software Versions

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

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5. Avaya IP Office Primary Configuration

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

Avaya IP Office is configured via the Avaya IP Office Manager program. For more information on provisioning Avaya IP Office Manager, consult reference **[4]**. From the Avaya IP Office Manager PC, select **Start →Programs →Avaya IP Office →Manager** to launch the Manager application. Navigate to **File → Open Configuration** (not shown), select the proper Avaya IP Office system from the pop-up window, and log in using the appropriate credentials.

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On Server Edition systems, the Solution View screen will appear, similar to the one shown below. All the Avaya IP Office configurable components are shown in the left pane, known as the Navigation Pane. Clicking the "plus" sign next to the Primary server system name, e.g., **IPOSE-Primary**, on the navigation pane will expand the menu on this server. In the sample configuration, IP users registered to the Primary server. Digital and Analog users are configured on the Expansion System. The SIP trunk to AT&T IPFR is configured on the Primary server.

In the screens presented in the following sections, the View menu was configured to show the Navigation pane on the left side, the Group pane in the center and the Details pane on the right side. These panes will be referenced throughout the rest of this document.

Standard feature configurations that are not directly related to the interfacing with the service provider are assumed to be already in place, and they are not part of these Application Notes.

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5.1. Licensing

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

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

Navigate to **License** in the Navigation Pane. In the Details Pane verify that the **License Status** for **SIP Trunk Channels** is Valid and that the number of **Instances** is sufficient to support the number of channels provisioned for the SIP trunk.

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5.2. System Settings

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

5.2.1. LAN 1 Tab

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

5.2.1.1 LAN 1 – LAN Settings Tab

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

- **IP Address:** Set to **10.64.19.170** as used in the reference configuration.
- **IP Mask:** Set to **255.255.255.0** as used in the reference configuration
- Click the **OK** button (not shown).

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5.2.1.2 LAN 1 - VoIP Tab

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

- The **H323 Gatekeeper Enable** box is checked to allow the use of Avaya IP Telephones using the H.323 protocol, such as the Avaya 96x1-Series Deskphones used in the reference configuration.
- The H.323 Signaling over TLS should be set based on customer needs. In the reference configuration it is set to **Preferred**.
- The **SIP Registrar Enable** box is checked to allow Avaya J100-Series Deskphones, Avaya 1100-Series Deskphones and Avaya IX™ Workplace usage.
- The **Domain Name** and **SIP Registrar FQDN** used in the reference configuration is **silipose.customer.com**.
- Set the **Layer 4 Protocol** section based on customer needs. In the reference configuration **TCP/5055** and **TLS/5056** is configured.
- Let all other values default.
- Click the **OK** button (not shown).

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5.2.1.3 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 (10.64.19.170).
- **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).

5.2.2. LAN 2 Tab

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

5.2.2.1 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.

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5.2.2.2 LAN 2 - VoIP Tab

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

- To prevent possible issues with network firewalls closing idle RTP channels, it is recommended that **RTP Keepalives** are enabled. Scrolling down to the bottom of the form, enter the following:
  - **Scope**: Select **RTP-RTCP**
  - **Periodic Timeout**: Enter **30**
  - **Initial keepalives**: Select **Enabled**
- Other parameters on this screen are set to the defaults.
- Click the **OK** button (not shown).

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5.2.2.3 LAN 2 - Network Topology Tab

In the reference configuration, it was not necessary to configure the Network Topology tab for the purposes of SIP trunking. If a NAT is used between Avaya IP Office and AT&T Border Element, then the SIP Line Use Network Topology Info field in **Section 5.4.3** should be set to the LAN2 interface used by the trunk and the Network Topology tab needs to be configured with the details of the NAT device. The Network Topology tab is summarized here for completeness.

- 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 site.
- **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, **60** is specified.
- The **Public IP Address** and **Public Port** fields are not used for the SIP trunk connection to AT&T. Click the **OK** button (not shown).

5.2.3. Voicemail Settings

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

- Set **Voicemail Type** to **Voicemail Lite/Pro**.
- Set **Voicemail IP Address** to the IP address of the server hosting voicemail. In the reference configuration, this is **10.64.19.170**.
- Other parameters on this screen are default. Click the **OK** button (not shown).

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5.2.4. System Telephony Settings

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

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5.2.5. System VoIP Settings

To view or change system codec settings, select the **VoIP** tab **and VoIP** sub-tab as shown on the screen below. 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.
- Click the **OK** button (not shown).

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5.2.6. System VoIP Security

To view or configure the media encryption settings, select the **VoIP → VoIP Security** tab on the Details pane. For the compliance test, SRTP was used internal to the enterprise wherever possible.

- Set the **Media** drop-down menu to **Preferred** to have IP Office attempt use SRTP for devices that support it, and fall back to RTP for devices that do not support encryption.
- Under **Media Security Options**, select **RTP** for the **Encryptions** and **Authentication** fields.
- Under **Crypto Suites**, select **SRTP_AES_CM_128_SHA1_80**.
- Click **OK** to commit (not shown).

5.3. IP Route

In the reference configuration, the Primary server LAN1 port is physically connected to the local area network switch at the IP Office customer site. The default gateway for this network is **10.64.19.1**. The IP Office LAN2 port is connected to the AT&T network and has a default gateway of **192.168.200.25**. To add an IP Route in the Primary server, right-click **IP Route** from the Navigation pane, and select **New** (not shown). To view or edit an existing route, select **IP Route** from the Navigation pane, and select the appropriate route from the Group pane. The following screen shows the Details pane with the relevant route using **LAN2** as **Destination**.

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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 5.4.1** to create a SIP Trunk from the template.

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

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

Therefore, it is important that the SIP Line configuration be reviewed and updated if necessary, after the SIP Line is created via the template. The resulting SIP Line data can be verified against the manual configuration as shown in **Sections 5.4.2 – 5.4.6**.

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

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

Alternatively, a SIP Line can be created manually. To do so, right-click **Line** in the **Navigation** pane and select **New → SIP Line**. Then, follow the steps outlined in **Sections 5.4.2 – 5.4.6**.

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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.

Copy a previously created template file to the computer where IP Office Manager is installed. To create the SIP Trunk from the template, right-click on **Line** in the Navigation Pane, then navigate to **New →New from Template**. Select **Open from file.**

Navigate to the directory where the template was copied on the local computer (e.g., *\temp*) and select it. Click **Open** (not shown).

The new SIP Line is created, and it will appear on the **Navigation** pane (e.g., SIP Line **2**). The resulting SIP Line data can be verified against the manual configuration shown in **Sections 5.4.2** to **5.4.6**.

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

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5.4.2. SIP Line – SIP Line tab

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

- **ITSP Domain Name**: Use the default <blank> to have IP Office send the **ITSP Proxy Address** as the domain name.
- **Local Domain Name:** Set to the public 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 to check the Out Of Service (OOS) condition, IP Office will use the SIP OPTIONS method to periodically check the SIP Line. The time between SIP OPTIONS sent by IP Office is determined by the Binding Refresh Time for LAN2, as shown in **Section 5.2.2**.
- **Refresh Method:** Set to **Re-Invite**, as AT&T does not support UPDATE.
- Set **Timer (sec)** to **1800** seconds. This field specifies the session expiry time. With this value, a session refresh message is sent every 15 minutes, at the half way point of the expiry time.
- **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 certain SIP telephones and Voicemail call flows**.
- Use the default values for the other fields.
- Click **OK** (not shown).

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5.4.3. 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**).
- **Layer 4 Protocol**: Set to **UDP**.
- **Send Port**: Set to **5060** (default).
- **Listen Port**: Set to **5060** (default).
- **Use Network Topology Info**: Set to **None** since no NAT was used in the test configuration.
- Verify **Calls Route via Registrar**: Enabled (default).
- **Click OK** (not shown).

Note – See Addendum 1 in **Section 11** for a configuration example and considerations when using multiple AT&T Border Elements.

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5.4.4. SIP Line – Call Details tab

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

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

- **Incoming Group:** Set to **2** (SIP Line 2). This value references the table created with **Incoming Call Routes** in **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.
- **Outgoing Group:** Set to **2** (SIP Line 2). This will be used for routing outbound calls to AT&T via the **ARS** configuration (**Section 5.9**).
- Verify **Diversion Header** is checked.
- **Local URI**, **Contact**, and **Diversion Header** fields for the **Display** and **Content** columns: Set these fields to **Auto**. With this configuration, information in the Incoming Call Route (**Section 5.8**) is used determine what call is accepted on the SIP Line. The Incoming Call Route for individual users will also be used to populate the SIP From and Contact headers for outbound calls.
- **Field meaning** section: Set to the values shown in the screenshot below.
- Click **OK**.

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In the reference configuration, the single SIP URI shown was sufficient to allow incoming calls for AT&T DID numbers destined for specific IP Office users, hunt groups or short codes.

5.4.5. 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 with **G729(a) 8K CS-ACELP** and **G.711 ULAW 64K** specified. This causes Avaya IP Office to include these codecs in the Session Description Protocol (SDP) offer, and in the order specified. Note that in the reference configuration G.729A is set as the preferred codec on the SIP trunk to the AT&T IPFR-EF network. During testing, the IP Office configuration was varied such that G.711MU was the preferred or only codec listed, and G.711MU calls were also successfully verified.
- T.38 fax was used in the reference configuration. Set the **Fax Transport Support** drop-down menu to **T.38**. G.711 fax also worked in the reference configuration (T.38 option disabled); however, T.38 is the preferred method.
- Set the **DTMF Support** field to **RFC2833/RFC4733**. This directs IP Office to send DTMF tones using RTP events messages as defined in 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).

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5.4.6. SIP Line – SIP Advanced Tab

Select the **SIP Advanced** tab. Set the parameters as shown below:

- Check the box for **Use PAI for Privacy**. 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.
- Select **Indicate HOLD**. 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 **Emulate NOTIFY for Refer**. The AT&T IPFR-EF service does not support NOTIFY. Some Avaya endpoints (e.g., Avaya 1140E Deskphone) require receipt of a NOTIFY when Refer based call transfers are performed. This option will send a NOTIFY to these endpoints.
- 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.
- Default values may be used for all other parameters.
- Click **OK** to commit (not shown).

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5.5. IP Office Line

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

The screen below shows the IP Office Line, **VoIP Settings** tab. In the reference configuration a fax machine is connected to one of the analog ports on the Expansion System. **Fax Transport Support** is set to **T.38**. The **Media Security** is set to **Same as System (Preferred)** to have IP Office use the system setting for media security set in **Section 5.2.6** to encrypted RTP. Default values were used for all other parameters.

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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.

In the screen shown below, the Short Code **9N** is illustrated. This 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 5.9**. 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 **62: ATT-IPFR**, which directs the call to ARS (see **Section 5.9**).
- Click the **OK** button (not shown).

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5.7. 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.

5.7.1. SIP User

The following screen shows the **User** tab for user 6241. As shown in **Figure 1**, this user corresponds to the Avaya J169 SIP endpoint.

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

The following screen shows the **VoIP** tab for the extension. The **IP Address** field may be left blank. Check the **Reserve Avaya IP endpoint license** box. The **Codec Selection** parameter may retain the default setting "**System Default**" to follow the system configuration shown in **Section 5.2.5**. The Media Security parameter may also retain the default setting "**Same as System (Preferred)**" to follow the system configuring shown in **Section 5.2.5**.

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5.7.2. 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 **Group** 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 the extension that has been unused for the longest period, due to the **Ring Mode** setting "**Longest Waiting**" (i.e., "longest waiting", most idle user receives next call). Click the **Edit** button to change the **User List**.

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5.8. 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 – When the destination is a user's extension, the **Incoming Number** can be used to construct the From and Contact headers in place of the extension number in the outgoing SIP INVITE message for the user. See **Section 5.4.4**.

In the screen shown below, the incoming call route for **Incoming Number → 3035559322** is illustrated. The **Line Group ID** is set to **2**, matching the **Incoming Group** field configured in the **Call Details** tab for the SIP Trunk to AT&T in **Section 5.4.4**.

Select the **Destinations** tab. From the **Destination** drop-down menu, select the extension to receive the call when AT&T delivers DNIS digits **3035559322**. In the reference configuration DNIS digits **3035559322** is associated with user 6**322**.

Repeat this process to route all AT&T DNIS numbers to additional telephone, as well as other Avaya IP Office destinations (Hunt Group, Voicemail, Short Codes, etc.). Incoming Call Routes for other direct mappings of DID numbers to IP Office users are omitted here but can be configured in the same fashion.

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5.9. Automatic Route Selection (ARS)

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**, 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., **62: ATT-IPFR**).

The following screen shows an example ARS configuration for the route **ATT-IPFR** (ARS Route ID 62). The list of codes defined below is simply an example and not intended to be prescriptive. Other dialing codes may be appropriate for different customer networks.

- 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).

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The example below shows the entry for 11 (1+10) digit calls.

- Set **Code** to **1xxxxxxxxxx**
  This means any dialed string starting with a 1, and 11 digits in total will be routed to the specified Line Group.
- **Feature** = **Dial**
- **Telephone Number** = **N**
  With this setting the leading 1 will be excluded, and only the remaining 10 digits will be sent to AT&T.
- **Line Group ID = 2** (SIP Line 2).

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5.10. Outbound Calling Party Information

Following to the SIP URI configuration on **Section 5.4.4**, Avaya IP Office will use the information on the Incoming Number field on the Incoming Call Routes (**Section 5.8**) to populate the From and Contact headers on outbound calls from the IP Office users. These headers are used by the service provider to deliver Caller ID information to the receiving end of the call.

There are cases where the IP Office user may not be associated with an Incoming Call Route. In other situations, it may be desired to send the same outbound caller ID for all the users in the enterprise. Short Codes can be created to construct the From and Contact headers in the outgoing SIP INVITE message.

5.10.1.Short Code - User

In cases where a user does not have an associated Incoming Call Route with a 10-digit number assigned to it, a Short Code can be created for that user to set the outbound caller ID.

To set a Short Code for an existing user, select **User** in the Navigation pane, and select the appropriate user to be configured in the Group pane. Select the **Short Code** tab on the Details pane and click **Add** (not shown).

The following screen shows the **Short Codes** tab for user 6240. This user does not have an associated Incoming Call Route.

- Set **Code** to **9N**
- In the example, the **Telephone Number** is set to "**Ns4695554878**". The number after the "s" will be sent on the From and Contact headers of the SIP INVITE on outbound calls from this user.
- **Feature** = **Dial**
- **Line Group ID = 62**, the ARS route to be used for AT&T.

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5.10.2.Short Code – ARS

There may be other specific cases when it is preferred to send the same caller ID on outbound calls for all the users in the enterprise, for example, the published main number for the location.

Short Code entries on the ARS table (**Section 5.9**) can be created or edited, to modify the From and Contact headers in the outgoing SIP INVITE messages for all outbound calls.

On the screen below, an existing ARS Short Code entry was modified. The **Telephone Number** is set to "**Ns4695554000**". The number after the "s" will be sent on the From and Contact headers of the SIP INVITE on outbound calls from all users in the Avaya IP Office using this ARS route.

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5.11. Saving Configuration Changes to Avaya IP Office

Navigate to **File → Save Configuration** in the menu bar at the top of the screen to save the configuration performed in the preceding sections.

The following will appear, with either **Merge** or **Reboot** selected for the **Change Mode**, based on the nature of the configuration changes made since the last save. Note that clicking **OK** may cause a service disruption.

Click **OK** to execute the save.

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

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6. Avaya IP Office Expansion Configuration

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

6.1. Expansion System - Physical Hardware

In the sample configuration, the IP500 V2 Expansion System contained a PHONE8 analog card and a COMBO6210 card, for the support of analog and digital stations. Also included is a VCM64 (Voice Compression Module). Both the VCM64 and the COMBO6210 cards provide voice compression channels to the control unit. Voice compression channels are needed to support VoIP calls, including IP extensions and or IP trunks.

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6.2. LAN Settings

In the sample configuration, LAN 1 is used to connect the Expansion System to the enterprise network. To view or configure the LAN 1 IP address, select the **LAN 1 → LAN Settings** tab, and enter the following:

- **IP Address:** Set to **10.5.5.180** as used in the reference configuration.
- **IP Mask:** Set to **255.255.255.0** as used in the reference configuration
- Click the **OK** button (not shown).

Defaults were used on the **VoIP** and **Network Topology** tabs (not shown).

6.3. Expansion System - IP Route

Configuration is done similar to the Primary server, as shown in **Section 5.3**. Configure a default route for the Expansion server. The default gateway in the reference configuration is **10.5.5.2**.

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6.4. Expansion System - IP Office Line

The IP Office Lines are automatically created on each server when the Expansion System is added to the solution. Below is the IP Office Line (**Line Number 17**) to the Primary server.

In the reference configuration, a fax machine is connected to one of the analog ports on the Expansion System. To accommodate T.38 fax, select the **VoIP Settings** tab and set **Fax Transport Support** to **T38 Fallback.** The **Media Security** drop-down menu is set to "**Same as System (Preferred)**" to have IP Office use the system setting for media security set in **Section 5.2.6** to encrypted RTP.

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Select the **T38 Fax** tab. The **Use Default Values** box is unchecked, and the **T38 Fax Version** is set to **0**. All other values are left at default.

6.5. Expansion System - Short Codes

Similar to the configuration of the Primary server in **Section 5.6**, create a Short Code to access ARS. In the reference configuration, the **Line Group ID** is set to an ARS route illustrated in the next section.

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6.6. Expansion System - Automatic Route Selection – ARS

The following screen shows an example ARS configuration for the route named "**To-Primary**" on the Expansion System. The **Line Group ID** is set to "**99999**" matching the number of the **Outgoing Group** configured on the IP Office Line 17 to the Primary server (**Section 6.4**).

6.7. Save IP Office Expansion System Configuration

Navigate to **File → Save Configuration** in the menu bar at the top of the screen to save the configuration performed in the preceding sections.

The following will appear, with either **Merge** or **Reboot** selected, based on the nature of the configuration changes made since the last save. Note that clicking **OK** may cause a service disruption. Click **OK** to proceed.

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7. 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 IPFR-EF features, and their associated access numbers, are also assigned by AT&T.

8. Verification Steps

The following procedures may be used to verify the Avaya IP Office Release 11.1 with the AT&T IP Flexible Reach - Enhanced Features service configuration.

8.1. 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 Voicemail Pro). 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.

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8.2. Avaya IP Office Release 11.1

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

8.2.1. System Status Application

The Avaya IP Office System Status application can be used to verify the service state of the SIP line. From the IP Office Manager application, select **File → Advanced → System Status**. Under **Control Unit IP Address** select the IP address of the IP Office system under verification. Log in using the appropriate credentials.

After logging in, select **Trunks → Line: 2** from the left navigation menu. (SIP Line 2 is configured in **Section 5.4**). On the **Status** tab in the right pane, verify that the **Current State** is ***Idle*** for each channel (assuming no active calls at present time).

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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).

8.2.2. System Monitor Application

The Avaya IP Office Monitor application can be used to monitor and troubleshoot signaling messaging on the SIP trunk. Launch the application from **Start → Programs → IP Office → Monitor** on the PC where IP Office Manager was installed. Click the **Select Unit** icon on the taskbar and Select the IP address of the IP Office system under verification.

Clicking the **Trace Options** icon on the taskbar and selecting the **SIP** tab allows modifying the threshold used for capturing events, types of packets to be captured, filters, etc. Additionally, the color used to represent the packets in the trace can be customized by right clicking on the type of packet and selecting to the desired color.

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9. Conclusion

As illustrated in these Application Notes, Avaya IP Office R11.1 can be configured to interoperate successfully with the AT&T IP Flexible Reach - Enhanced Features service using **AVPN** or **ADI/PNT** transport connections, within the limitations described in **Section 2.2**.

This solution provides users of Avaya IP Office R11.1 the ability to support inbound and outbound calls utilizing an AT&T IPFR-EF SIP trunk service connection, via AVPN or ADI/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.

10. Additional References

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

[1] *IP Office, Deploying IP Office Server Edition Solution*, Release 11.1, Issue 14, April 2020
[2] *IP Office™ Platform 11.1, Deploying Avaya IP Office Servers as Virtual Machines,* April 2020
[3] *IP Office™ Platform 11.1, Deploying an IP500 V2 IP Office Essential Edition System,* April 2020.
[4] *Administering Avaya IP Office™ Platform with Manager,* Release 11.1, Issue 1, April 2020.
[5] *Administering Avaya IP Office™ Platform with Web Manager,* Release 11.1, Issue 1, April 2020.
[6] *Planning for and Administering Avaya IX™ Workplace Client for Android, iOS, Mac and Windows,* Release 3.8, Issue 1, March 2020
[7] *Using Avaya IX™ Workplace Client for Android, iOS, Mac and Windows*, Release 3.8, Issue 1, March 2020.

Additional Avaya IP Office information can be found at:
https://ipofficekb.avaya.com/

AT&T IPFR-EF Service:

[8] Information on the AT&T IP Flexible Reach service can be found here: https://www.business.att.com/products/sip-trunking.html.

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11. 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 support this redundant configuration.

Given two AT&T border elements **192.168.38.69** (primary) and **192.168.37.149** (secondary), the SIP Line (Line 2) defined in **Section 5.4**, can be modified to include the secondary Border Element.

1. Navigate to **Line → 2 → Transport Tab** and populate the **ITSP Proxy Address** field with the primary address, followed by the secondary address, separated by a space.

2. Click **OK** (not shown), and save the configuration as shown in **Section 0**.

When completed, Avaya IP Office will send OPTIONS to both of the AT&T Border Elements, as well as respond to OPTIONS received from these addresses.

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.

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