Avaya Solution & Interoperability Test Lab

Application Notes for Avaya IP Office Release 10.0, Avaya Session Border Controller for Enterprise Release 7.1 with AT&T IP Transfer Connect Service – Issue 1.0

Abstract

These Application Notes describe the steps for configuring Avaya IP Office Release 10.0 and Avaya Session Border Controller for Enterprise Release 7.1, with the AT&T IP Transfer Connect service using AVPN or MIS/PNT transport connections.

The AT&T IP Transfer Connect service is a service option available with the AT&T IP Toll Free service, and supports the rerouting of inbound toll free calls to alternate destinations based upon SIP redirection messages from Avaya IP Office R10.0.

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.

Note that these Application Notes are intended to supplement the separate document: *Application Notes for Avaya IP Office Release 10.0, Avaya Session Border Controller for Enterprise Release 7.1 with the AT&T IP Toll Free Service– Issue 1.0.*

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

1. Introduction

These Application Notes describe the steps for configuring Avaya IP Office Release 10.0 and Avaya Session Border Controller for Enterprise Release 7.1, with the AT&T IP Transfer Connect service using **AVPN** or **MIS/PNT** transport connections[1].

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.

These Application Notes describe the steps for provisioning Avaya IP Office Release 10.0 and the Avaya Session Border Controller for Enterprise Release 7.1 (Avaya SBCE), with the AT&T IP Transfer Connect service (IPTC). AT&T IP Transfer Connect is a service option available with the AT&T IP Toll Free service, and supports the rerouting of inbound toll free calls to alternate[2] destinations based upon SIP redirection messages from Avaya IP Office.

Note – These Application Notes are intended to supplement the separate document: *Application Notes for Avaya IP Office Release 10.0, Avaya Session Border Controller for Enterprise Release 7.1 with the AT&T IP Toll Free Service– Issue 1.0*. This document is listed in **Section 9** as reference document **[5]**. It is recommended that this AT&T IP Toll Free service document should be available as a reference during provisioning to the AT&T IP Transfer Connect service.

Note – The AT&T IP Transfer Connect service is referred to in the remainder of the document as *IPTC*.

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 IPTC and the Customer Premises Equipment (CPE) containing the Avaya IP Office Release 10.0 (see **Section 3.2** for call flow examples).

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[1] MIS/PNT transport does not support compressed RTP (cRTP), however AVPN transport does support cRTP
[2] Note that this is NOT the same as the "Alternate Destination Routing (ADR)" service option available with the AT&T IP Toll Free service.

The test environment described in these Application Notes consisted of:

- A simulated enterprise with Avaya IP Office; Avaya SBCE; Voicemail Pro; Avaya SIP, H.323 and Analog telephones.
- Laboratory versions of the IPTC service, to which the simulated enterprise was connected via AVPN/MIS transport.

2.1. Interoperability Compliance Testing

The compliance testing was based on a test plan provided by AT&T, for the functionality required for certification as a solution supported on the IPTC network. Calls were made from the PSTN across the IPTC test network, to the CPE. The interoperability compliance testing focused on verifying inbound call flows (see **Section 3.2**) between Avaya IP Office, Avaya SBCE and the IPTC service.

The following SIP trunking VoIP features were tested with the IPTC service:

- Inbound IPTC calls to Avaya IP Office SIP (1140E, and Communicator for Windows softphone)/H.323 (1616, and 9611)/Analog (6211) telephones; utilizing G.729A codec (IPTC preferred codec).
- Inbound IPTC calls that are immediately redirected by a SIP 302 message, generated by Avaya IP Office, back to the IPTC service for redirection to an alternate destination.
- Inbound IPTC calls that are redirected by a SIP Refer (without Replaces) message, generated by Avaya IP Office/Voicemail Pro, back to the IPTC service for redirection to an alternate destination. However in this case an announcement is played to the caller by Avaya IP Office/Voicemail Pro, prior to the redirection.
- Inbound IPTC INFOPAK data transmission in SIP messaging.
- IPTC International (MOW) inbound calling capabilities.
- Avaya IP Office features such as hold, resume, and local transfer.
- SIP OPTIONS messages used to monitor the health of the SIP trunks between the CPE and AT&T.

2.2. Test Results

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

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

2. **Avaya IP Office does not support transmission of User-to-User (UUI) data**. The IPTC service allows for the optional inclusion of UUI data in both 302 and Refer SIP messages. However Avaya IP Office/Voicemail Pro currently does not support transmission of UUI data.

3. **IPTC service Landline/Mobility test cases could not be executed**. The AT&T supplied IPTC test plan specifies test cases to verify the transmission of Landline/Mobility data by the IPTC service. Due to network provisioning issues, these test cases could not be executed.

4. **Avaya SBCE does not change the host portion of the Contact header URI in the 302 Moved Temporarily from IP Office**. The 302 Moved Temporarily response sent to AT&T included the internal IP address of the Avaya SBCE in the host portion of the Contact header. Although no issues were found during testing, (i.e., AT&T accepted the response), the host should contain the far-end domain/IP address. To correct this anomaly, IP Office was configured to send the AT&T IPTC Border Element IP address in the *ITSP Domain Name* field (see **Section 5.1.1**). With this IP address set, the 302 Moved Temporarily message sent to AT&T will have the correct URI host in the Contact header.

2.3. Support

AT&T customers may obtain support for the AT&T IP Transfer Connect service by calling (800) 325-5555.

Avaya customers may obtain documentation and support for Avaya products by visiting: http://support.avaya.com. In the United States, (866) GO-AVAYA (866-462-8292) provides access to overall sales and service support menus. Customers may also use specific numbers (provided on http://support.avaya.com) to directly access specific support and consultation services based upon their Avaya support agreements.

3. Reference Configuration

The reference configuration used in these Application Notes is shown in **Figure 1** on the next page and consists of the following components:

- An Avaya Server Edition Primary server with an IP500 V2 Expansion System for analog and digital endpoint support. The single Server Edition Primary server provides IP Office Server Edition, Voicemail Pro, and Avaya one-X® Portal for IP Office.
- Avaya IP Office Server Edition provides the voice communications services for a particular enterprise site.
- Voicemail Pro provides the “Modules” required to generate the Refer (without Replaces) SIP messaging (see **Section 5.3**).
- Avaya “desk” telephones are represented with an Avaya 1616 H.323 set, an Avaya 9611 H.323 set, an Avaya 9508 Digital set, an Avaya 1140E SIP set, as well as Avaya Communicator for Windows (SIP).
- In the reference configuration, both the Avaya IP Office (interface “LAN 1”), and the Avaya SBCE (interface “A1”) are connected to the private CPE network. The Avaya SBCE interface “B1” is connected to the AT&T network.
- The AT&T IPTC service requires the following SIP trunk network settings between the Avaya SBCE interface “B1” and the IPTC 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 IPTC service may deliver various digit lengths in the SIP Invite R-URI depending on the circuit order provisioning. In the reference configuration, the IPTC service delivered 15 digits.

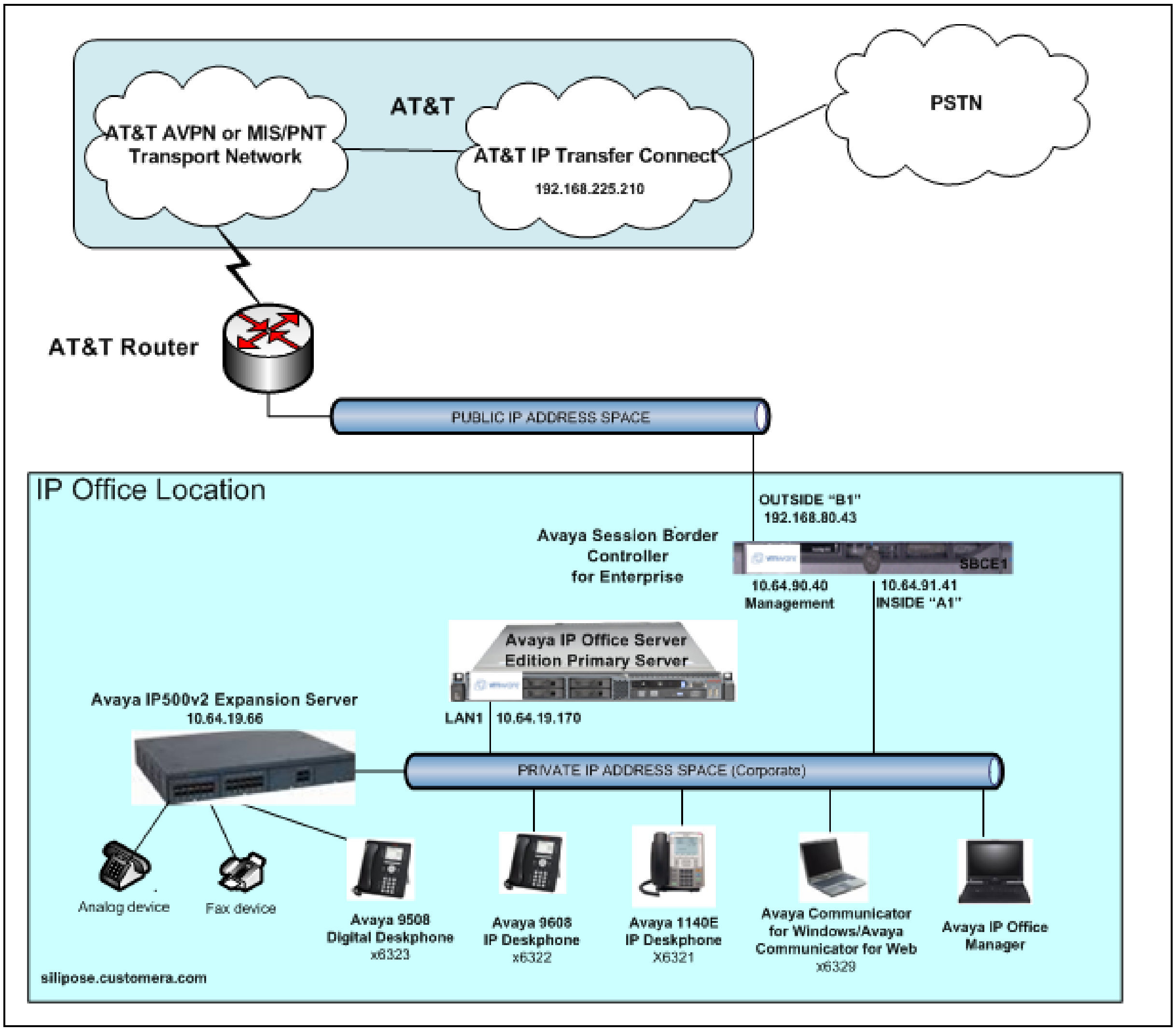


Figure 1: Reference Configuration

3.1. Illustrative Configuration Information

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

Note – The Avaya SBCE "B1" interface communicates with AT&T Border Elements (BEs) located in the AT&T IPTC network. For security reasons, the IP addresses of the AT&T BEs are not included in this document. However as placeholders in the following configuration sections, the IP addresses **192.168.80.42** (Avaya SBCE "B1" IP address), and **192.168.225.210** (AT&T BE IP address), are specified. In addition, AT&T DID/DNIS numbers shown in this document are examples as well. AT&T Customer Care will provide the actual Border Element IP addresses and DID/DNIS numbers as part of the IPTC provisioning process.

Component	Illustrative Value in these Application Notes
Avaya IP Office Server Edition	
Private network LAN1 interface	10.64.19.170
Avaya IP Office Expansion System (IP500 V2)	
Private network LAN1 interface	10.64.19.66
Avaya SBCE	
Private network "A1" interface.	10.64.91.41
Public network "B1" interface.	192.168.80.43
AT&T IPTC Service	
Border Element IP Address	192.168.225.210

Table 1: Illustrative Values Used in these Application Notes

3.2. Call Flows

To understand how inbound AT&T IPTC service calls are handled by Avaya IP Office, three basic call flows are described in this section.

3.2.1. Basic Inbound Call

The first call scenario illustrated in **Figure 2** is an inbound IPTC service call that arrives at Avaya IP Office, and is subsequently routed to an endpoint. Note that no call redirection is performed in this scenario.

1. A PSTN phone originates a call via the IPTC service.
2. The PSTN routes the call to the IPTC service network.
3. The IPTC service routes the call to Avaya IP Office.
4. Depending on the called number, Avaya IP Office routes the call to the associated endpoint.

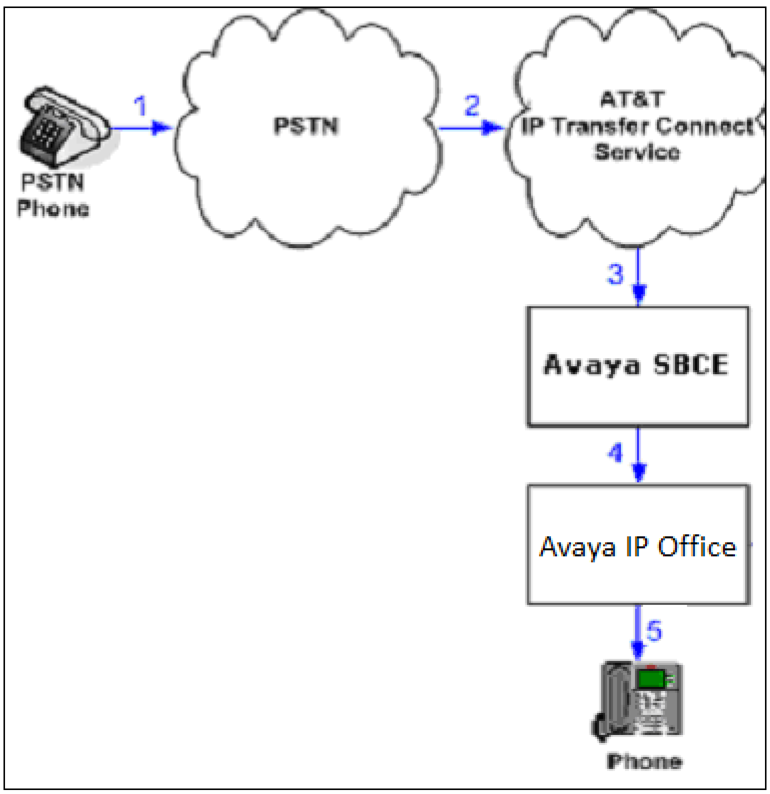


Figure 2: Inbound IPTC Call – No Redirection

3.2.2. 302 Call Redirection

Note: In the call scenarios that follow, the term "alternate destination" does NOT refer to the "Alternate Destination Routing (ADR)" service option of the AT&T IP Toll Free service. ADR and the IPTC service are unrelated.

The second call scenario illustrated in **Figure 3** is an inbound IPTC service call that arrives at Avaya IP Office, which in turn generates the 302 SIP message.

1. Same as the first three steps from the call scenario illustrated in **Section 3.2.1**.
2. Avaya IP Office redirects the call by sending a SIP 302 message back out the SIP trunk (see **Section 5.2.1**). The SIP 302 message is routed back to the IPTC network. Avaya IP Office releases the trunk.
3. The IPTC service places a call to the alternate destination and upon answer, connects the calling party to the target party (alternate destination).

Note that no audio is transmitted between Avaya IP Office and the PSTN caller during the 302 transaction.

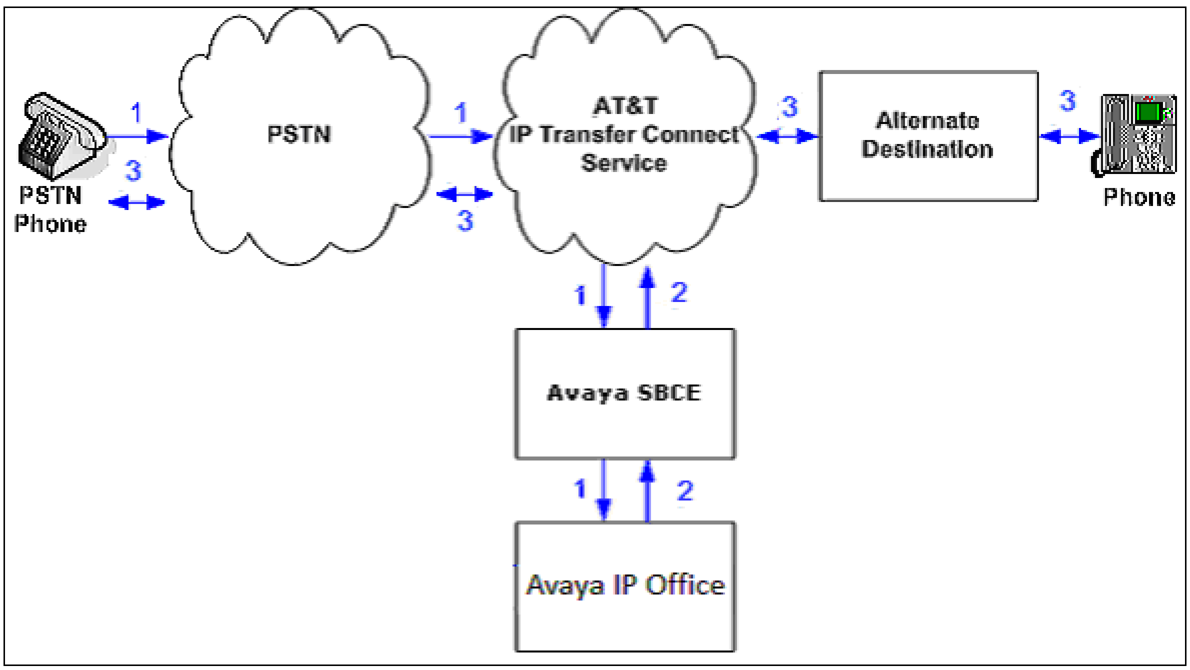


Figure 3: Inbound IPTC Call – SIP 302 Redirection

3.2.3. Refer Call Redirection

The third call scenario illustrated in **Figure 4** is an inbound IPTC service call that is routed to Avaya IP Office, which routes the call to Voicemail Pro. A predefined Voicemail Pro Module redirects the call back to the IPTC service using a Refer[3], for routing to an alternate destination.

1. Same as the first step from the call scenario illustrated in **Section 3.2.2**.
2. Avaya IP Office routes the call to Voicemail Pro.
3. Voicemail Pro executes a corresponding Module (see **Section 5.3**), which plays an announcement back to the PSTN caller, stating that the call is being redirected.
4. Voicemail Pro redirects the call by sending a SIP Refer (without Replaces) message back out on the SIP trunk. The SIP Refer message is sent to the IPTC service network. Avaya IP Office releases the trunk. The IPTC service places a call to the alternate destination and upon answer, connects the calling party to the target party (alternate destination).

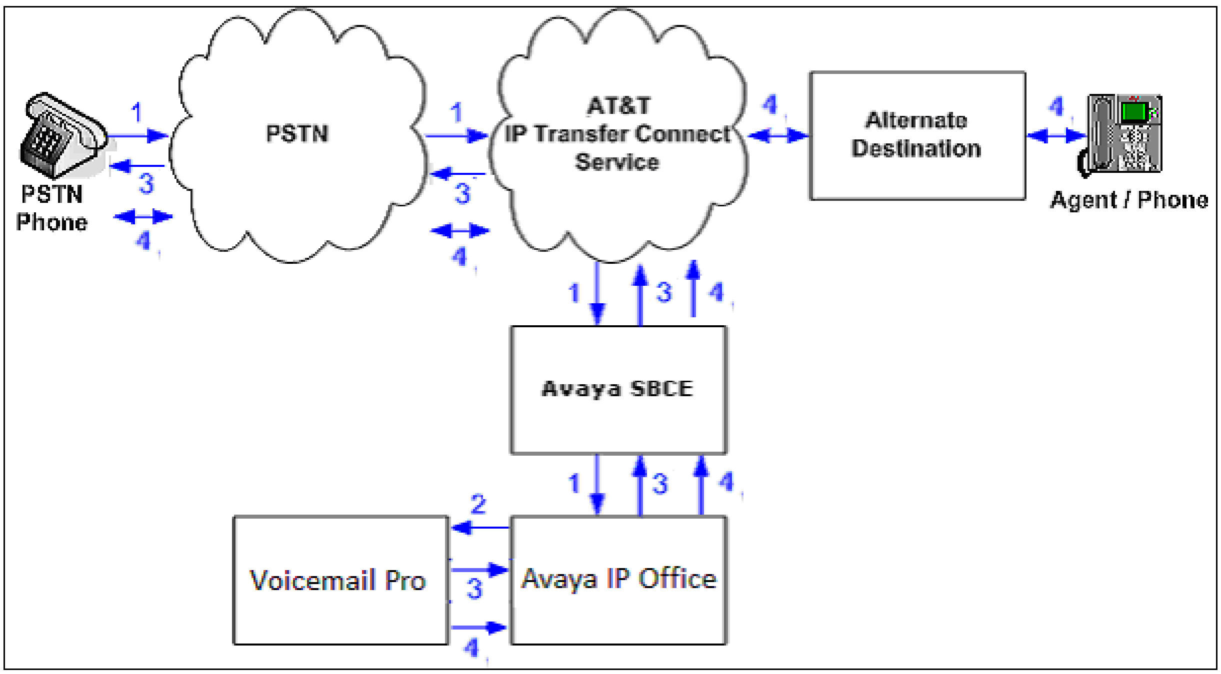


Figure 4: Inbound IPTC Call –SIP REFER Redirection

[3] This is a Refer *without* the Replaces parameter (e.g., a “Blind Refer”).

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 Server Edition ▪ IP Office ▪ Voicemail Pro ▪ Avaya WebRTC Gateway ▪ Avaya one-X® Portal for IP Office	 10.0.0.1.0 build 53 10.0.0.0.0 build 469 10.0.0.1.0 build 3 10.0.0.1.0 build 16
Avaya IP Office 500 V2 ▪ Avaya IP Office TCM 8 ▪ Avaya IP Office COMBO6210/ATM4	 10.0.0.1.0 build 53 10.0.0.1.0 build 53
Avaya IP Office Server Edition Manager	10.0.0.1.0 build 53
Avaya Session Border Controller for Enterprise	7.1.0.1-07-12368
Avaya Communicator for Windows (SIP)	2.1.3.237
Avaya 9641G (H.323) IP Deskphone	6.6302
Avaya 1616 (H.323) Telephone	Ha1616ua1_390A.bin
Avaya 1140E (SIP) Telephone	04.04.23
Avaya 9508 Digital Telephone	0.59
Avaya 6211 Analog Telephone	N/A

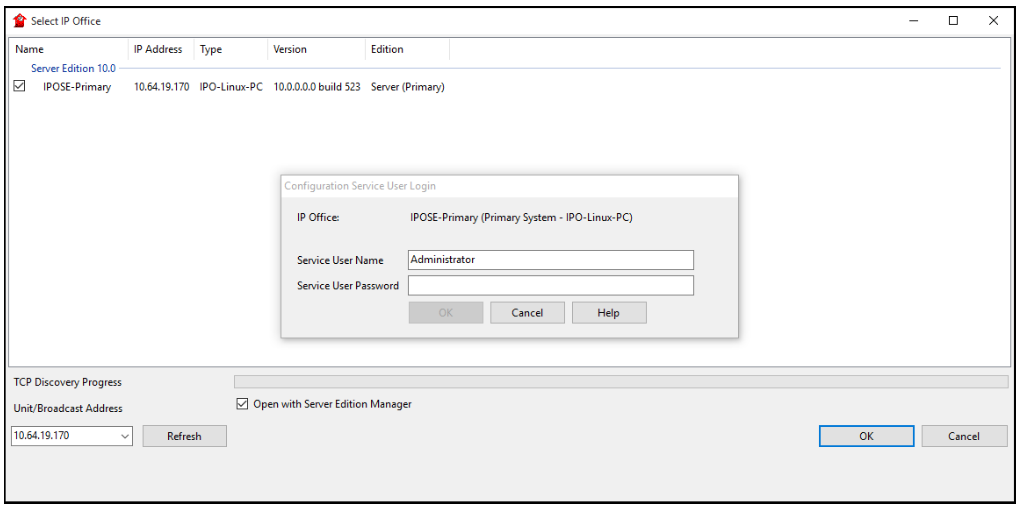
Table 2: Equipment and Software Versions

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.

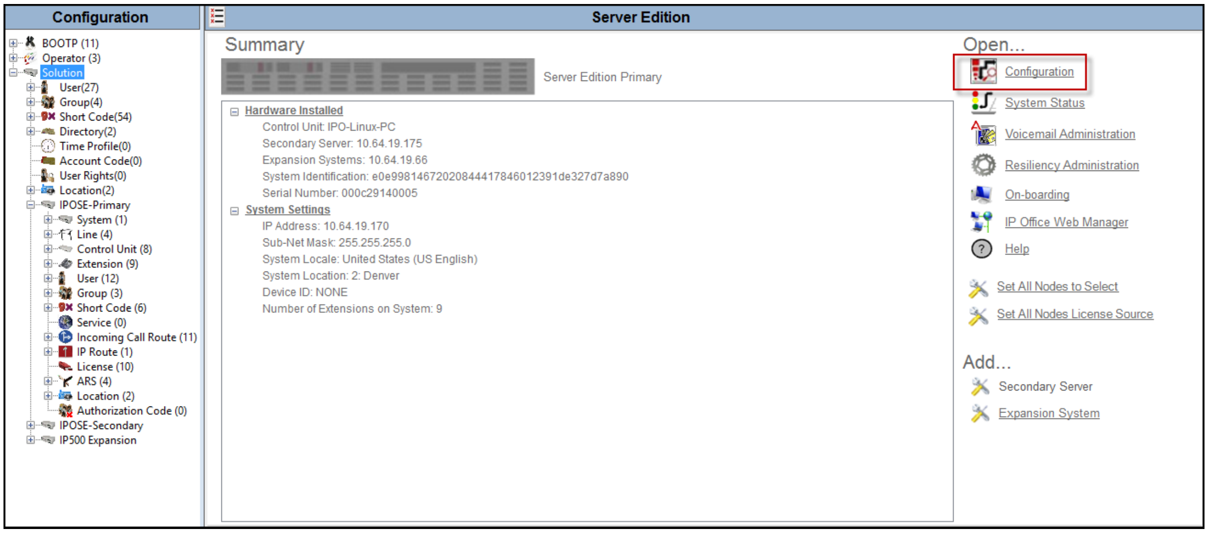
5. Avaya IP Office Primary Configuration

Note – Avaya IP Office administration for interaction with the AT&T IP Toll Free service is described in document **[5]** and is applicable for the IPTC service as well (see the note in **Section 1**). This section describes the additional administration steps on Avaya IP Office necessary for supporting interaction with the IPTC service. It is recommended that the AT&T IP Toll Free service document should be available as a reference during provisioning to the AT&T IP Transfer Connect service.

IP Office is configured via the IP Office Manager program. For more information on IP Office Manager, consult reference **[3]**. From the IP Office Manager PC, select **Start → All Apps → IP Office → Manager** to launch the Manager application. Navigate to **File → Open Configuration** (not shown), select the proper Avaya IP Office system from the pop-up window, and log in using the appropriate credentials.



On Server Edition systems, the Solution View screen will appear, similar to the one shown on the next page. If the left navigation pane does not immediately appear, click on the **Configuration** link as highlighted below. In the reference configuration, IP users registered to the Primary server and failover to the Secondary server. Digital and Analog users are configured on the Expansion System. A SIP trunk to the AT&T IPTC service is configured on the Primary server. Clicking the "plus" sign next to the Primary server system name, e.g., **IPOSE-Primary,** on the left navigation pane will expand the menu on this server.



5.1. SIP Line

The following sections describe the configuration of a SIP Line. The SIP Line terminates the CPE end of the IP Office SIP trunk to the Avaya SBCE, and ultimately to the AT&T IPTC 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 IPTC service.

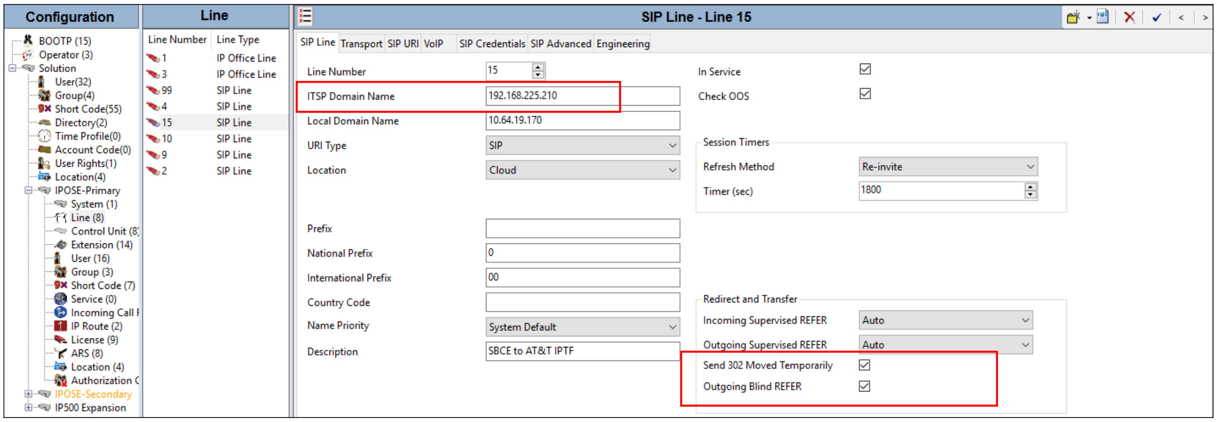
Follow the steps in **Sections 5.4.1** and **5.4.2** of document **[5]** to import/create a SIP Trunk from the template.

Note - In document **[5]**, SIP Line **2** was created for use with the AT&T IP Toll Free service. SIP Line 17 was used for the IPTC testing as well, and is referenced in the following sections.

5.1.1. SIP Line – SIP Line tab

The **SIP Line** tab is shown below for the existing **Line Number 2** (see the note above). This SIP Line form is modified as follows for the IPTC service:

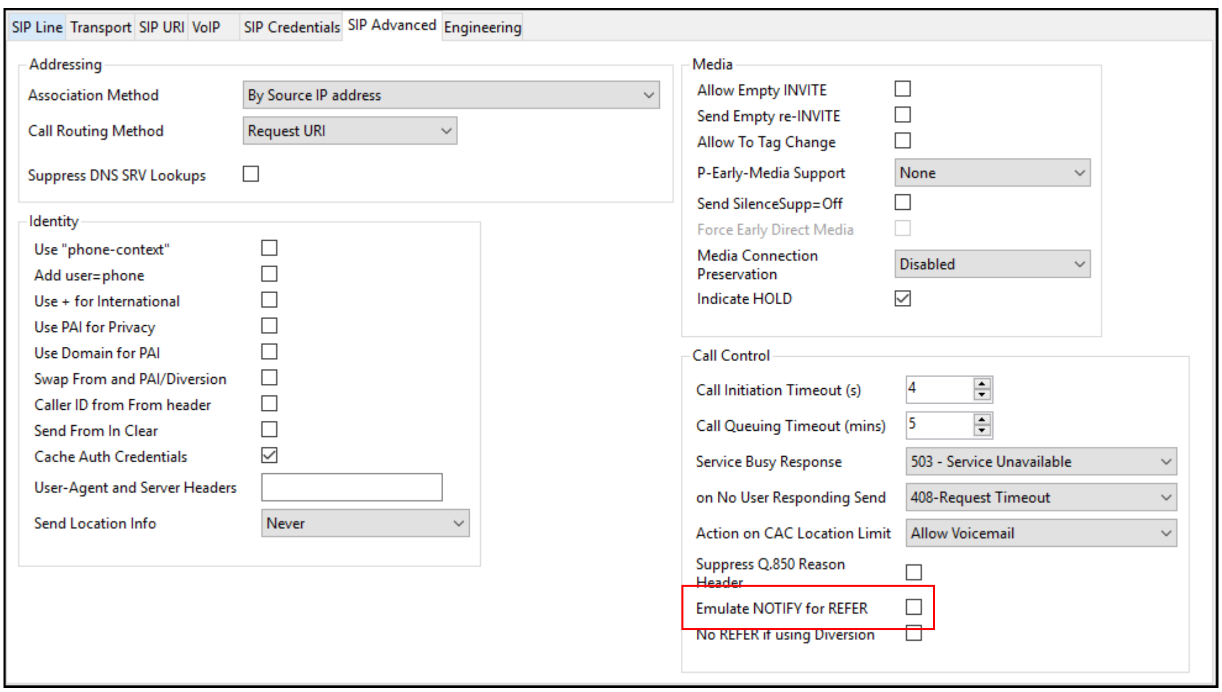
- **ITSP Domain Name**: Set to the IP address of the AT&T Border Element IP address (e.g., **192.168.225.210**). See **Section 2.2** for more details.
- Check the box next to **Send 302 Moved Temporarily**.
- Check the box next to **Outgoing Blind Refer**.
- Click on OK.



5.1.2. SIP Line – SIP Advanced Tab

Navigate to **SIP Line → SIP Advanced** tab.

- Verify that the **Emulate NOTIFY for REFER** is *not* checked.
- Click on OK.



5.2. Incoming Call Routes to Trigger 302 or Refer Call Redirection

Two call redirection methods are supported by the IPTC service; SIP 302 and Refer (without Replaces). While both of these methods utilize the Avaya IP Office Incoming Call Route table, the Destinations specified for each are different. The 302 redirection is triggered by Avaya IP Office, while the Refer redirection is triggered by a Module defined in Voicemail Pro.

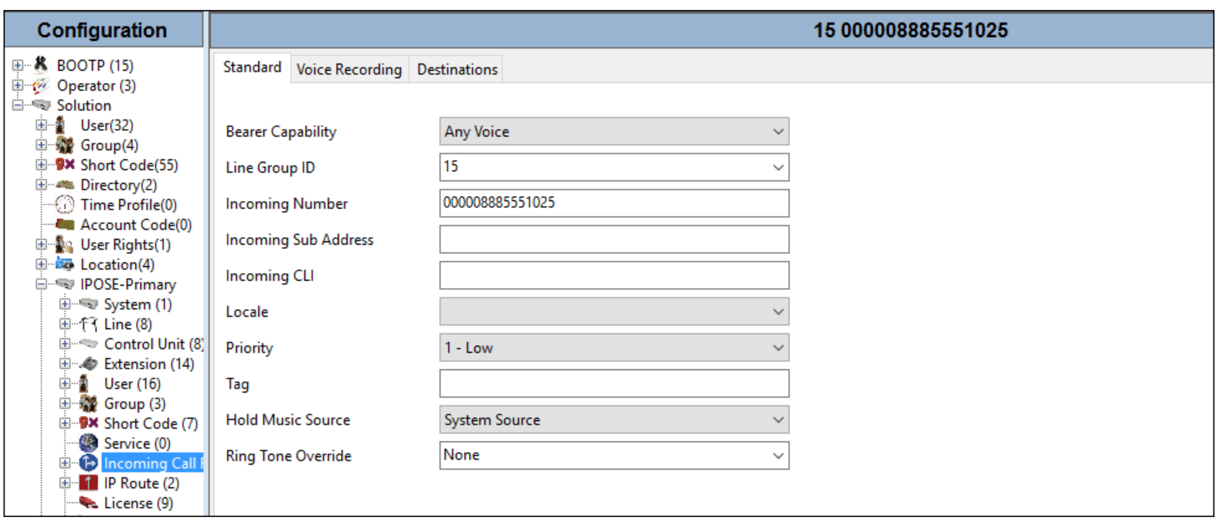
Note – In the reference configuration the IPTC service provided the access number 1012 for use in the 302 and Refer testing.

Note – Although the IPTC is an inbound only service, an outbound Avaya IP Office Short Code must be defined to trigger the 302 and Refer Call Redirections. See **Section 5.4**.

5.2.1. 302 Call Redirection

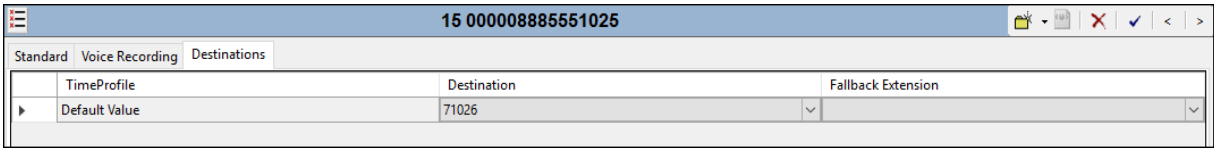
In the example below, the incoming number **000008885551025** is directed to trigger the 302 Call Redirection.

1. From the **Incoming Call Route** page, select the **Standard** tab and enter the following:
    - **Line Group ID**: Enter the SIP Line previously defined in **Section 5.4** of document **[5]** (e.g., **15**).
    - **Incoming Number**: Enter the associated DNIS digits sent by AT&T (e.g., **000008885551025**).
    - Use default values for the remaining fields.



2. Select the **Destinations** tab and enter the following:
    - Enter the string **71026** to the drop down menu, and click **OK** (not shown).

In this example, **7** is the outbound dialing Short Code (see **Section 5.4**), and **1026** is the IPTC defined access number to be used for the call redirection.

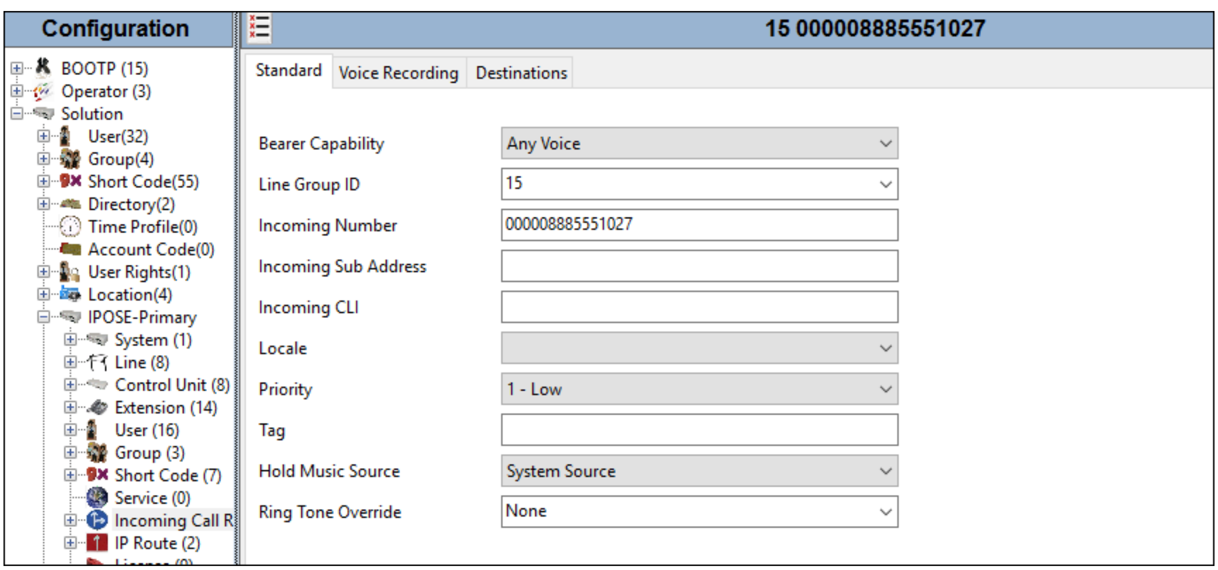


When the 000008885551025 number is received in an Invite, Avaya IP Office will generate a 302 message, with 1026 in the Contact header, back to the IPTC service. The IPTC service will then generate a new Invite to 1026.

5.2.2. Refer Call Redirection

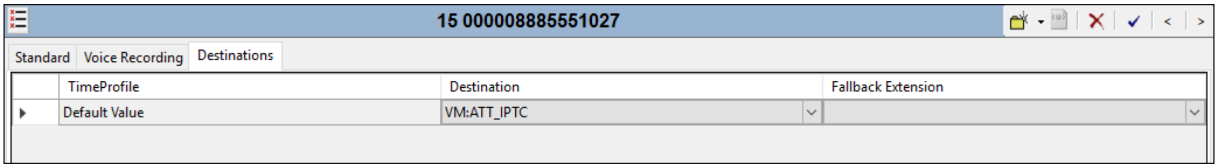
In the reference configuration, Voicemail Pro (running on Primary server), is used to send a Refer (without Replaces) Call Redirection. A Voicemail Pro "Module" is defined with the name **Refer** (see **Section 5.3**). This Module name is defined as a Destination to an inbound call as follows:

1. From the **Incoming Call Route** page, select the **Standard** tab enter the following:
    2. **Line Group ID**: Enter the SIP Line previously defined in **Section 5.4** of document **[5]** (e.g., **15**).
    - **Incoming Number**: Enter the associated DNIS digits sent by AT&T (e.g., **000008885551027**).
    - Use default values for the remaining fields.



3. Select the **Destinations** tab and enter the following:
   - In the **Destinations** column, enter the string **VM:ATT_IPTC** to the drop down menu, and click **OK** (not shown).

In this example, **VM:** specifies that the destination is a Module on Voicemail Pro, and **ATT_IPTC** is the name of the Module (see **Section 5.3**).



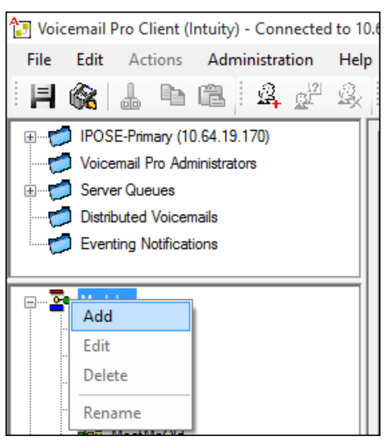
When the 000008885551027 number is received in an Invite, Avaya IP Office/Voicemail Pro will play and announcement to the caller, then generate a Refer (without Replaces) message, (with 1028 in the Refer-To header), back to the IPTC service. The IPTC service will then generate a new Invite to 1028.

5.3. Voicemail Pro Refer Module

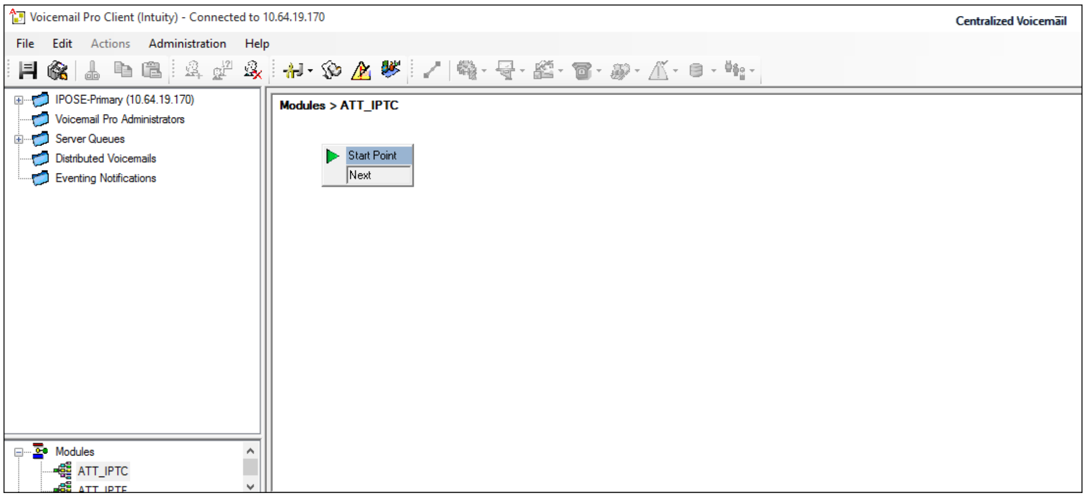
Note - While Avaya Voicemail Pro provisioning and programming is beyond the scope of this document, a sample Module is described below.

The Refer Module is provisioned to play an announcement to the caller, and then generate a Refer (without Replaces) back to the IPTC service. This is accomplished via the following steps via the Voicemail Pro Client interface:

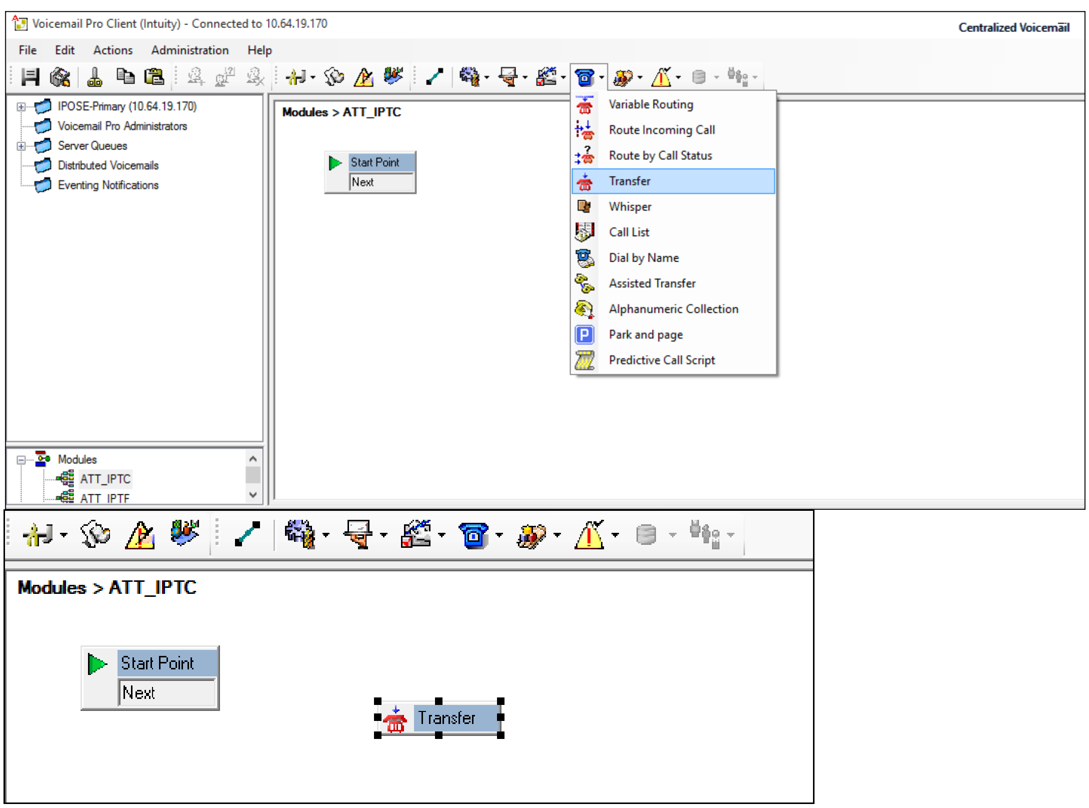
1. Open the **Voicemail Pro Client** application (not shown).
2. Create a **Start Point** by right clicking on **Modules** and selecting **Add**.



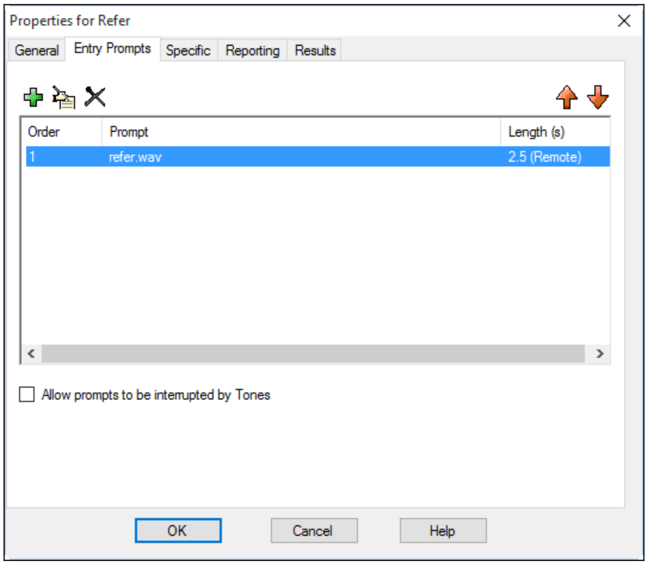
3. Enter a name (e.g., **ATT_IPTC**) and click on **OK** (not shown). The new script “ATT_IPTC” will appear under Modules and a Start Point icon will appear in the work area.



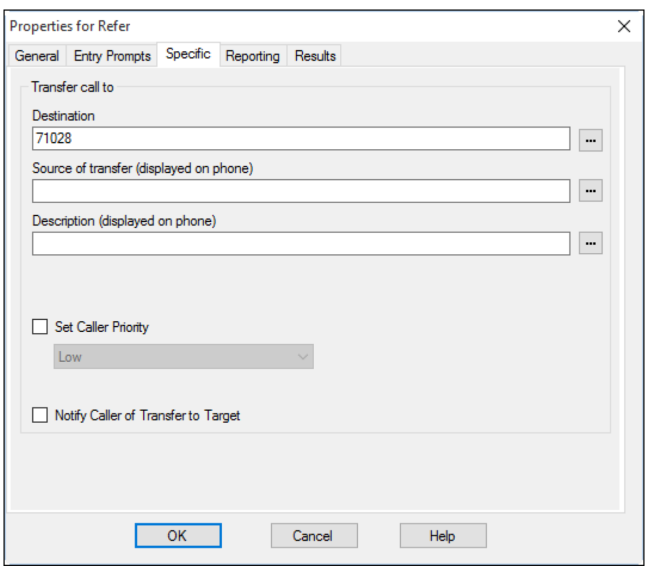
- Click on the **Start Point** icon  to activate the script options at the top of the screen. From the options, **Telephony Actions** icon , select the **Transfer** icon , and click on the work area to place the **Transfer** icon in the work area.



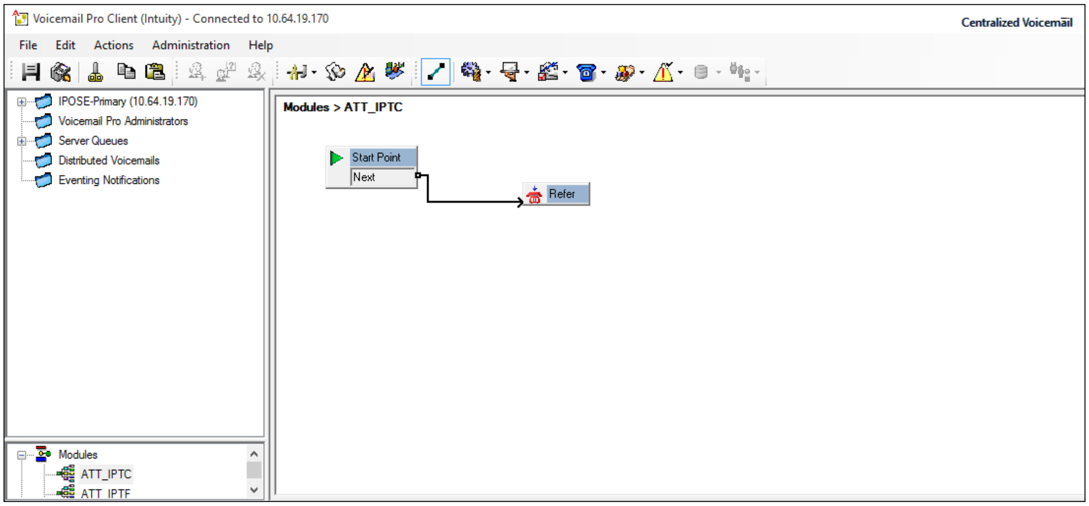
4. Double click on the **Transfer** icon. On the **General** tab → **Token Name** field, enter **Refer** (not shown).
5. Select the **Entry Prompts** tab and select or create an announcement to be played to the caller prior to the Refer (e.g., **refer.wav**). To modify an existing recording, double click on the .wav file and rerecord. If no .wav files exist, double click on the  icon to open the .wav file editor.



6. On the **Specific** tab enter **71028**, where **7** is the Avaya IP Office outbound Short Code, and **1028** is the redirection number specified by the IPTC service (see **Section 5.2.2**).
7. Click on **OK**.



8. From the options bar, select the **Connector** icon and drag a connecting flow line from the **Start Point** box to the **Transfer** box.



9. From the top menu select **File → Save & Make Live**, or select the icon.

When the IPTC DNIS number is received (e.g., **000008885551027**), IP Office sends the call to Voicemail Pro (see **Section 5.2.2**). The caller will hear an announcement (e.g., **refer.wav**), and Voicemail Pro/Avaya IP Office sends a Refer back to the IPTC service, specifying **1028** in the Refer-To header. The IPTC service will then send a new Invite to the 1028 destination.

5.4. Outbound Short Code for 302 and Refer Call Redirection

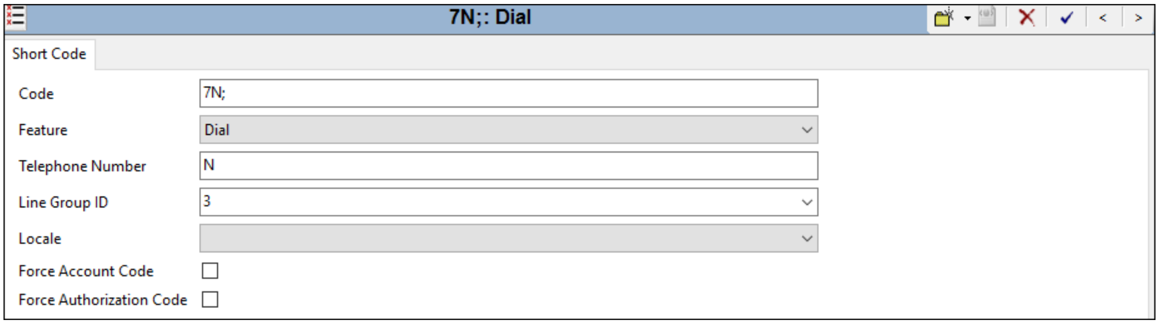
Avaya IP Office provides predefined Short Codes, however new Short Codes may be defined to match number strings to an action. To add a Short Code, right click on **Short Code** in the Navigation pane, and select **New** (not shown). To edit an existing Short Code, click **Short Code** in the Navigation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

Note – Although the IPTC is an inbound only service, an *outbound* Short Code must be defined to trigger the 302 and Refer Call Redirections.

In the following screen, the Short Code **7N;** is illustrated (note the semicolon at the end of the string). This Short Code will allow Avaya IP Office to generate a 302 or Refer message back to the IPTC service (see **Sections 5.2** and **5.3**).

1. Right click on **Short Codes** from the left hand menu and select **New** (not shown).
   - The **Code** parameter is set to **7N;** (note that **7** was used in the reference configuration, however any available number string may be used).
   - The **Feature** parameter is set to **Dial**.
   - The **Telephone Number** parameter is set to **N**.
   - The **Line Group ID** parameter is set to the SIP Line previously defined in **Section 5.4** of document **[5]** (e.g., **3**).

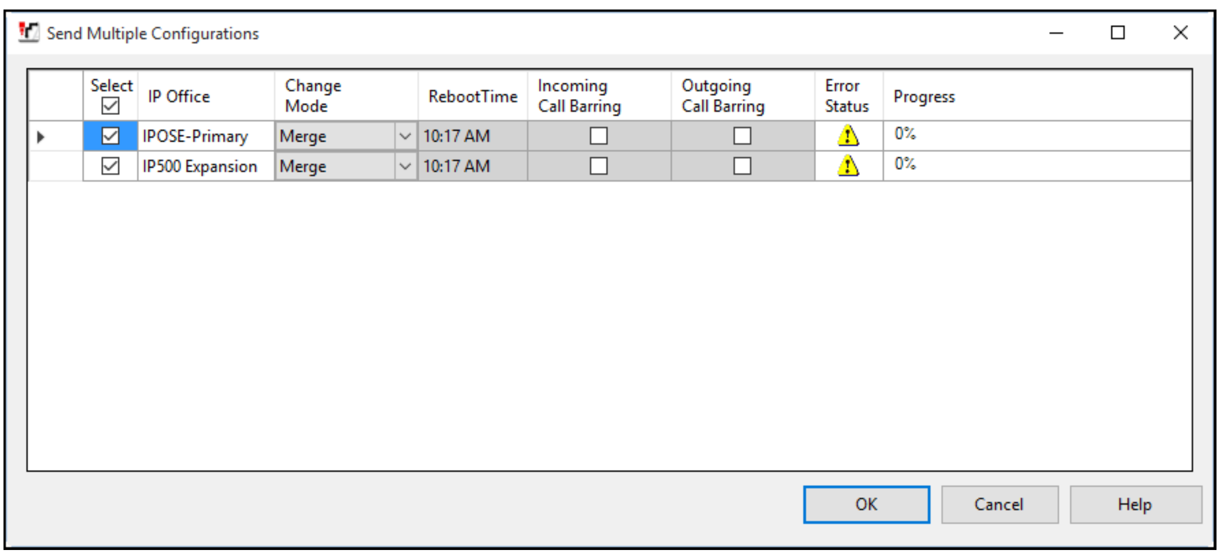
- Click the **OK** button (not shown).



5.5. 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 configuration may be saved to a file at any time by selecting **File → Save Configuration As**.

6. Avaya Session Border Controller for Enterprise

Avaya SBCE configuration for interaction with the AT&T IP Toll Free service provided in document **[7]** should also be followed for interoperability with the IPTC service. No additional administration steps are required on the Avaya SBCE for supporting interaction with the IPTC service.

7. AT&T IP Transfer Connect service Configuration

AT&T provides the IPTC service border element IP address, the access DID numbers, and the associated DNIS digits used in the reference configuration. In addition the AT&T IPTC features, and their associated access numbers, are also assigned by AT&T. AT&T requires that the Avaya IP Office public (LAN2) IP address be provided to the IPTC service, as part of the provisioning process.

8. Verification Steps

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

8.1. Call Verification Tests

The call verification steps and troubleshooting tools described for the AT&T Toll Free service described in document **[5]**, also apply to the IPTC service. However additional verification steps specific to the IPTC service are described below.

1. Place an inbound call to an IPTC service line enabled with Redirect features. Verify that Avaya IP Office redirects the call back to the IPTC service for redirection to an alternate destination using 302. Verify that the 302 message contains the redirection number in the Contact header. Verify two-way talk path and transmission between the caller and the redirected destination.
2. Place an inbound call to an IPTC service line enabled with Refer features. Verify that Avaya IP Office directs the call to Voicemail Pro, which then redirects the call back to the IPTC service using Refer (without Replaces) for redirection to an alternate destination. Verify that the caller hears an announcement prior to the call redirection. Verify that the Refer message contains the redirection number in the Refer-To header, and that the Refer-To header *does not* contain a "Replaces" parameter. Verify two-way talk path and transmission between the caller and the redirected destination.

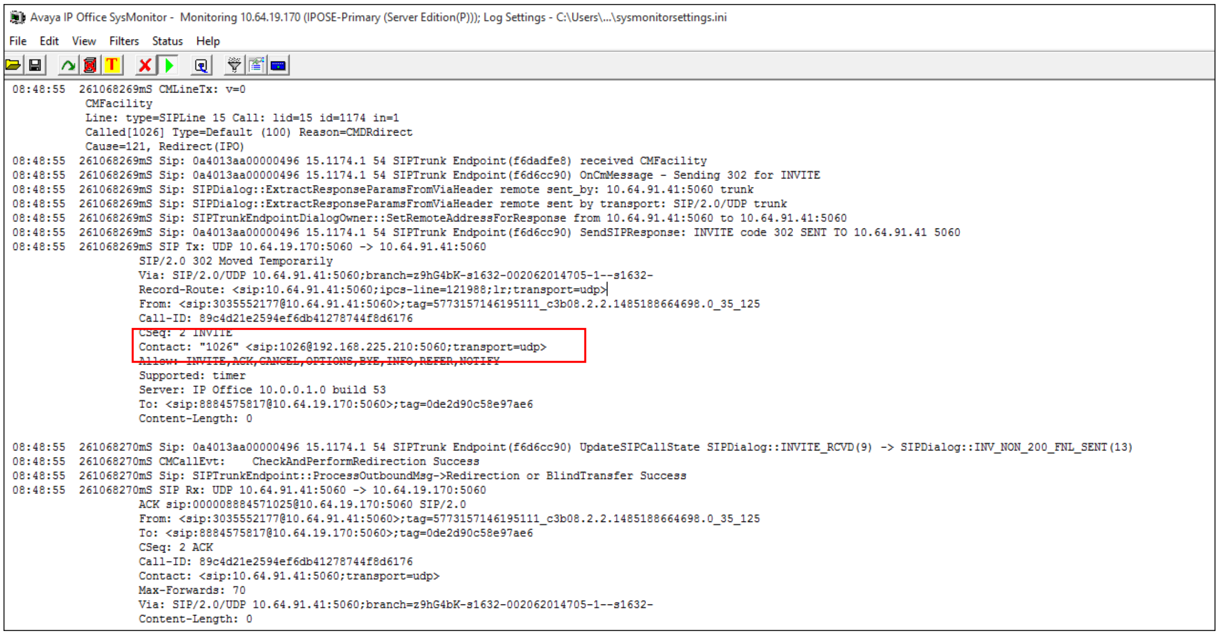
8.2. System Monitor Traces

Monitor the SIP traffic at the connection to the IPTC service, using IP Office System Monitor. The System Monitor application can typically be accessed from **Start → Programs → Avaya IP Office → Monitor**.

8.2.1.302 Redirection

The following is an example of a 302 redirection.

- The Contact header contains the new destination number (1026) as defined in the Avaya IP Office Incoming Call Route Destination field (see **Section 5.2.1)**.



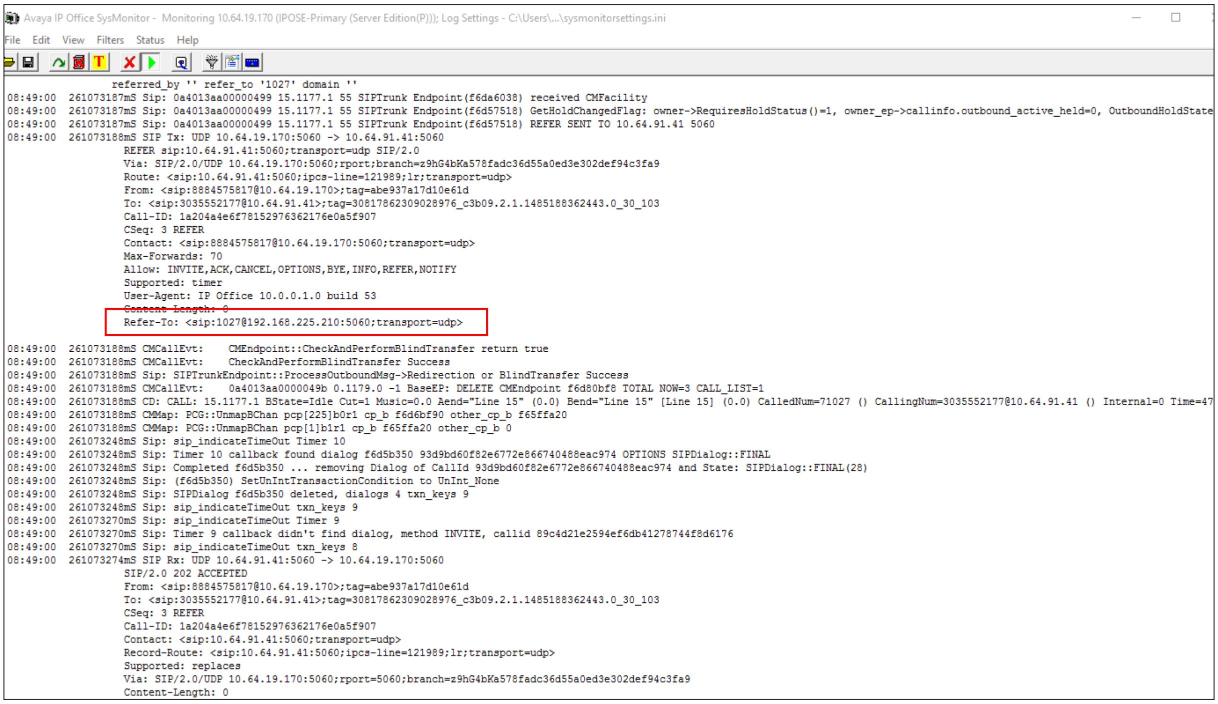

```
08:48:55  261068269mS CMLineTx: v=0
           CMFacility
           Line: type=SIPLine 15 Call: lid=15 id=1174 in=1
           Called[1026] Type=Default (100) Reason=CMDRdirect
           Cause=121, Redirect(IPO)
08:48:55  261068269mS Sip: 0a4013aa00000496 15.1174.1 54 SIPTrunk Endpoint(f6dadfe8) received CMFacility
08:48:55  261068269mS Sip: 0a4013aa00000496 15.1174.1 54 SIPTrunk Endpoint(f6d6cc90) OnCmMessage - Sending 302 for INVITE
08:48:55  261068269mS Sip: SIPDialog::ExtractResponseParamsFromViaHeader remote sent_by: 10.64.91.41:5060 trunk
08:48:55  261068269mS Sip: SIPDialog::ExtractResponseParamsFromViaHeader remote sent by transport: SIP/2.0/UDP trunk
08:48:55  261068269mS Sip: SIPTrunkEndpointDialogOwner::SetRemoteAddressForResponse from 10.64.91.41:5060 to 10.64.91.41:5060
08:48:55  261068269mS Sip: 0a4013aa00000496 15.1174.1 54 SIPTrunk Endpoint(f6d6cc90) SendSIPResponse: INVITE code 302 SENT TO 10.64.91.41 5060
08:48:55  261068269mS SIP Tx: UDP 10.64.19.170:5060 -> 10.64.91.41:5060
                  SIP/2.0 302 Moved Temporarily
                  Via: SIP/2.0/UDP 10.64.91.41:5060;branch=z9hG4bK-s1632-002062014705-1--s1632-
                  Record-Route: <sip:10.64.91.41:5060;ipcs-line=121988;lr;transport=udp>
                  From: <sip:3035552177@10.64.91.41:5060>;tag=5773157146195111_c3b08.2.2.1485188664698.0_35_125
                  Call-ID: 89c4d21e2594ef6db41278744f8d6176
                  CSeq: 2 INVITE
                  Contact: "1026" <sip:1026@192.168.225.210:5060;transport=udp>
                  Allow: INVITE,ACK,CANCEL,OPTIONS,BYE,INFO,REFER,NOTIFY
                  Supported: timer
                  Server: IP Office 10.0.0.1.0 build 53
                  To: <sip:8884575817@10.64.19.170:5060>;tag=0de2d90c58e97ae6
                  Content-Length: 0

08:48:55  261068270mS Sip: 0a4013aa00000496 15.1174.1 54 SIPTrunk Endpoint(f6d6cc90) UpdateSIPCallState SIPDialog::INVITE_RCVD(9) -> SIPDialog::INV_NON_200_FNL_SENT(13)
08:48:55  261068270mS CMCallEvt:    CheckAndPerformRedirection Success
08:48:55  261068270mS Sip: SIPTrunkEndpoint::ProcessOutboundMsg->Redirection or BlindTransfer Success
08:48:55  261068270mS SIP Rx: UDP 10.64.91.41:5060 -> 10.64.19.170:5060
                  ACK sip:000008884571025@10.64.19.170:5060 SIP/2.0
                  From: <sip:3035552177@10.64.91.41:5060>;tag=5773157146195111_c3b08.2.2.1485188664698.0_35_125
                  To: <sip:8884575817@10.64.19.170:5060>;tag=0de2d90c58e97ae6
                  CSeq: 2 ACK
                  Call-ID: 89c4d21e2594ef6db41278744f8d6176
                  Contact: <sip:10.64.91.41:5060;transport=udp>
                  Max-Forwards: 70
                  Via: SIP/2.0/UDP 10.64.91.41:5060;branch=z9hG4bK-s1632-002062014705-1--s1632-
                  Content-Length: 0
```

8.2.2. Refer

The following is an example of a Refer redirection.

- The Refer-To header contains the new destination number (1027) as defined in the Voicemail Pro Refer Module (see **Section 5.3)**. Also note that the Refer-To header *does not* contain a “Replaces” parameter.



```
Avaya IP Office SysMonitor -  Monitoring 10.64.19.170 (IPOSE-Primary (Server Edition(P))); Log Settings - C:\Users\...\sysmonitorsettings.ini
File  Edit  View  Filters  Status  Help
                 referred_by '' refer_to '1027' domain ''
08:49:00  261073187mS Sip: 0a4013aa00000499 15.1177.1 55 SIPTrunk Endpoint(f6da6038) received CMFacility
08:49:00  261073187mS Sip: 0a4013aa00000499 15.1177.1 55 SIPTrunk Endpoint(f6d57518) GetHoldChangedFlag: owner->RequiresHoldStatus()=1, owner_ep->callinfo.outbound_active_held=0, OutboundHoldState
08:49:00  261073187mS Sip: 0a4013aa00000499 15.1177.1 55 SIPTrunk Endpoint(f6d57518) REFER SENT TO 10.64.91.41 5060
08:49:00  261073188mS SIP Tx: UDP 10.64.19.170:5060 -> 10.64.91.41:5060
                   REFER sip:10.64.91.41:5060;transport=udp SIP/2.0
                   Via: SIP/2.0/UDP 10.64.19.170:5060;rport;branch=z9hG4bKa578fadc36d55a0ed3e302def94c3fa9
                   Route: <sip:10.64.91.41:5060;ipcs-line=121989;lr;transport=udp>
                   From: <sip:8884575817@10.64.19.170>;tag=abe937a17d10e61d
                   To: <sip:3035552177@10.64.91.41>;tag=30817862309028976_c3b09.2.1.1485188362443.0_30_103
                   Call-ID: 1a204a4e6f78152976362176e0a5f907
                   CSeq: 3 REFER
                   Contact: <sip:8884575817@10.64.19.170:5060;transport=udp>
                   Max-Forwards: 70
                   Allow: INVITE,ACK,CANCEL,OPTIONS,BYE,INFO,REFER,NOTIFY
                   Supported: timer
                   User-Agent: IP Office 10.0.0.1.0 build 53
                   Content-Length: 0
                   Refer-To: <sip:1027@192.168.225.210:5060;transport=udp>

08:49:00  261073188mS CMCallEvt:    CMEndpoint::CheckAndPerformBlindTransfer return true
08:49:00  261073188mS CMCallEvt:    CheckAndPerformBlindTransfer Success
08:49:00  261073188mS Sip: SIPTrunkEndpoint::ProcessOutboundMsg->Redirection or BlindTransfer Success
08:49:00  261073188mS CMCallEvt:    0a4013aa0000049b 0.1179.0 -1 BaseEP: DELETE CMEndpoint f6d80bf8 TOTAL NOW=3 CALL_LIST=1
08:49:00  261073188mS CD: CALL: 15.1177.1 BState=Idle Cut=1 Music=0.0 Aend="Line 15" (0.0) Bend="Line 15" [Line 15] (0.0) CalledNum=71027 () CallingNum=3035552177@10.64.91.41 () Internal=0 Time=47
08:49:00  261073188mS CMMap: PCG::UnmapBChan pcp[225]b0r1 cp_b f6d6bf90 other_cp_b f65ffa20
08:49:00  261073188mS CMMap: PCG::UnmapBChan pcp[1]b1r1 cp_b f65ffa20 other_cp_b 0
08:49:00  261073248mS Sip: sip_indicateTimeOut Timer 10
08:49:00  261073248mS Sip: Timer 10 callback found dialog f6d5b350 93d9bd60f82e6772e866740488eac974 OPTIONS SIPDialog::FINAL
08:49:00  261073248mS Sip: Completed f6d5b350 ... removing Dialog of CallId 93d9bd60f82e6772e866740488eac974 and State: SIPDialog::FINAL(28)
08:49:00  261073248mS Sip: (f6d5b350) SetUnIntTransactionCondition to UnInt_None
08:49:00  261073248mS Sip: SIPDialog f6d5b350 deleted, dialogs 4 txn_keys 9
08:49:00  261073248mS Sip: sip_indicateTimeOut txn_keys 9
08:49:00  261073270mS Sip: sip_indicateTimeOut Timer 9
08:49:00  261073270mS Sip: Timer 9 callback didn't find dialog, method INVITE, callid 89c4d21e2594ef6db41278744f8d6176
08:49:00  261073270mS Sip: sip_indicateTimeOut txn_keys 8
08:49:00  261073274mS SIP Rx: UDP 10.64.91.41:5060 -> 10.64.19.170:5060
                   SIP/2.0 202 ACCEPTED
                   From: <sip:8884575817@10.64.19.170>;tag=abe937a17d10e61d
                   To: <sip:3035552177@10.64.91.41>;tag=30817862309028976_c3b09.2.1.1485188362443.0_30_103
                   CSeq: 3 REFER
                   Call-ID: 1a204a4e6f78152976362176e0a5f907
                   Contact: <sip:10.64.91.41:5060;transport=udp>
                   Record-Route: <sip:10.64.91.41:5060;ipcs-line=121989;lr;transport=udp>
                   Supported: replaces
                   Via: SIP/2.0/UDP 10.64.19.170:5060;rport=5060;branch=z9hG4bKa578fadc36d55a0ed3e302def94c3fa9
                   Content-Length: 0
```

9. Conclusion

As illustrated in these Application Notes, Avaya IP Office Release 10.0, and the Avaya Session Border Controller for Enterprise Release 7.1 can be configured to interoperate successfully with the AT&T IP Transfer Connect service, within the limitations described in **Section 2.2**.

The sample 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. 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 04g, Jan 2017

[2] *IP Office ™ Platform 10.0, Deploying Avaya IP Office ™ Platform IP500 V2,* Document Number 15-601042, Issue 31q, Feb 2017

[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/

Avaya Application Notes (available at www.avaya.com/devconnect)

[5] *Application Notes for Avaya IP Office Release 10.0, Avaya Session Border Controller for Enterprise Release 7.1 with AT&T IP Toll Free Service – Issue 1.0*

AT&T IP Transfer Connect Service:

[6] AT&T IP Transfer Connect service description - http://www.business.att.com/enterprise/Service/voice-services/contact-center-solutions/ip-toll-free/

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.