Avaya Solution & Interoperability Test Lab

Application Notes for Avaya IP Office Release 11.1 and Avaya Session Border Controller for Enterprise Release 8.1 with AT&T IP Transfer Connect Service – Issue 1.0

Abstract

These Application Notes describe the steps for configuring Avaya IP Office Release 11.1 and Avaya Session Border Controller for Enterprise Release 8.1, with the AT&T IP Transfer Connect service AVPN or ADI/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 Release.

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 11.1 and Avaya Session Border Controller for Enterprise Release 8.1 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.

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1. Introduction

These Application Notes describe the steps for configuring Avaya IP Office Release 11.1 (Avaya IP Office) and the Avaya Session Border Controller for Enterprise (Avaya SBCE) Release 8.1 with the AT&T IP Transfer Connect service using AT&T Virtual Private Network (AVPN) or AT&T Dedicated Internet Service (ADI/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.

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 11.1 and Avaya Session Border Controller for Enterprise Release 8.1 with the AT&T IP Toll Free Service – Issue 1.0*. This document is listed in **Section 10** as reference document **[9]**. 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 call flows between IPTC and the Customer Premises Equipment (CPE) containing the Avaya IP Office Release 11.1 and Avaya SBCE 8.1 (see **Section 3.2** for call flow examples).

[1] ADI/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.

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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, Digital and Analog telephones.
- Laboratory versions of the IPTC service, 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 IPTC 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.

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, J169 and Avaya Equinox softphone), H.323 (9608) and Digital (9508) 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.
- Verify reception of IPTC SIP Multipart/NSS headers, including SDP and XML content.
- 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**. Although no issues were found during testing, AT&T recommends that for maximum customer bandwidth utilization, a ptime value of 30 msecs should be specified.

2. **Avaya SBCE private IP address is sent on the Contact header of 302 Moved Temporarily messages**. During the tests it was observed that the Contact header of outbound 302 Moved Temporarily messages sent to AT&T included the private IP address of the Avaya SBCE private interface. Even though call redirections were successful, in order to prevent the private IP address from being sent across the AT&T network, the ITSP Domain Name on the IP Office SIP Line (**Section 5.1.1**) was set to a fictitious domain name, as a workaround. With this setting, this domain is used to populate the host part of the Contact header of outbound 302 Moved Temporarily messages. This issue is under investigation by Avaya.

3. **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. Although Avaya IP Office 11.0 supports passing UUI headers information on transit calls across a SIP trunk, it did not send any UUI data in the configuration tested. This issue is being investigated by Avaya IP Office development.

4. **Enhanced CID – NSS feature**. The inbound calls to Avaya IP Office are not exercising the Enhanced CID feature. Although Avaya IP Office is accepting SIP Multipart/NSS headers, it is neither passing nor acting upon it, this data is simply being ignored.

5. **IPTC services IP InfoPack** and **Landline/Mobility test cases could not be executed**. The AT&T supplied IPTC test plan specifies test cases to verify the inbound transmission of INFOPAK and Landline/Mobility data by the IPTC service. Due to network provisioning and support issues, these test cases could not be executed.

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.

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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:

- 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. Note that this solution is extensible to deployments using the standalone IP500 V2 platform as well.
- In the sample configuration, the Primary server runs the Avaya IP Office Server Edition Linux software. Avaya Voicemail Pro, running as a service on the Primary Server, provided the "Modules" required to generate the Refer (without Replaces) SIP messaging (see **Section 5.3**).
- The Expansion System (V2) is used for the support of digital, analog and additional IP stations. It consists of an Avaya IP Office 500v2 with analog and digital extension expansion modules, as well as a VCM64 (Voice Compression Module).
- Avaya endpoints are represented with an Avaya 9608 H.323 deskphone, an Avaya 9508 Digital deskphone, Avaya J169 and 1140E SIP deskphones, as well as Avaya Workplace Client for Windows (SIP) softphone.
- 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 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 10 digits.
- An Avaya Remote Worker endpoint (Avaya Workplace Client for Windows) was used in the reference configuration. The Remote Worker endpoint resides on the public side of an Avaya SBCE (via a TLS connection), and registers/communicates with IP Office as though it was an endpoint residing in the private CPE space.

Note – The configuration of the Remote Worker environment is beyond the scope of this document. Refer to [**5**] on the **Additional References** section for information on Remote Worker deployments.

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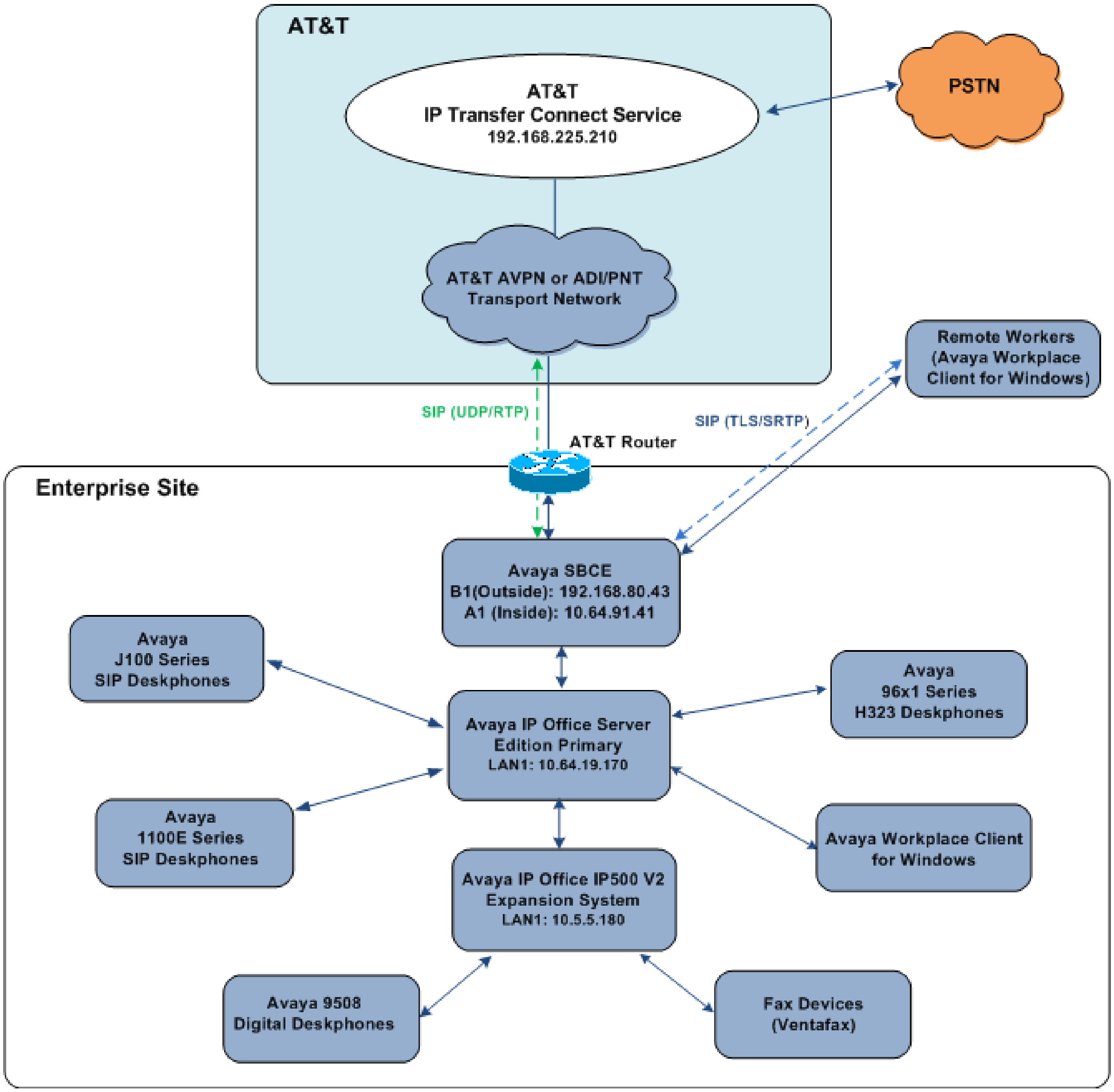

Figure 1: Reference Configuration

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3.1. Illustrative Configuration Information

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

Note – The Avaya 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 **192.168.80.43** (Avaya SBCE "B1"), 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	
Private network LAN1 interface	10.64.19.170
Expansion System, LAN1 Interface	10.5.5.180
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

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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 the Avaya SBCE.
4. The Avaya SBCE performs SIP Network Address Translation (NAT) and any necessary SIP header modifications, and routes the call to Avaya IP Office.
5. Depending on the called number, Avaya IP Office routes the call to the associated endpoint.

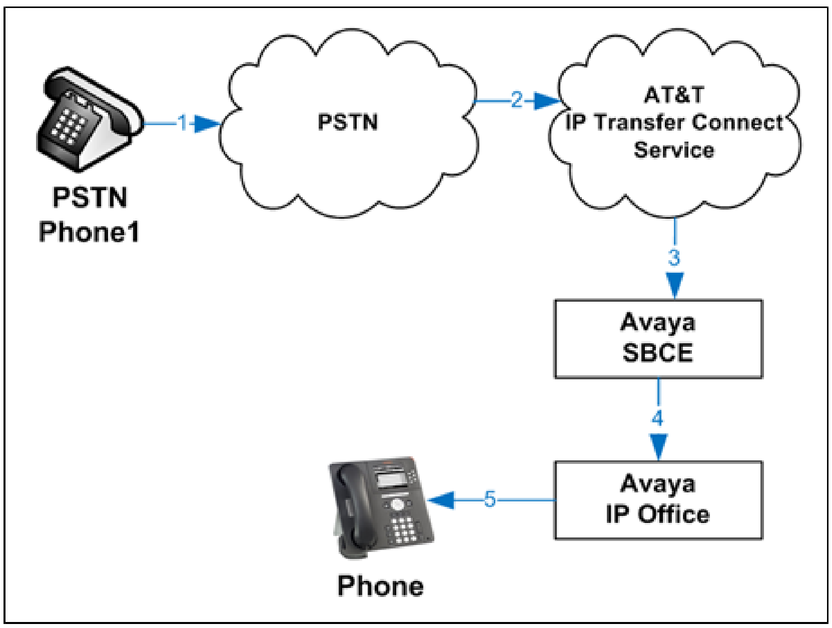

Figure 2: Inbound IPTC Call – No Redirection

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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 a 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.

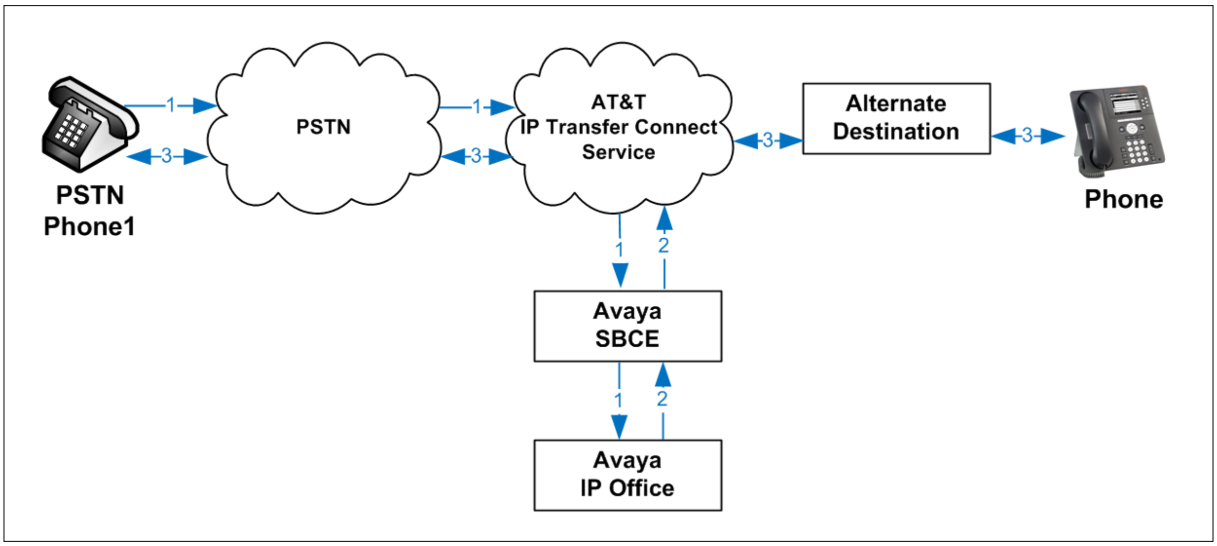

Figure 3: Inbound IPTC Call – SIP 302 Redirection

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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).

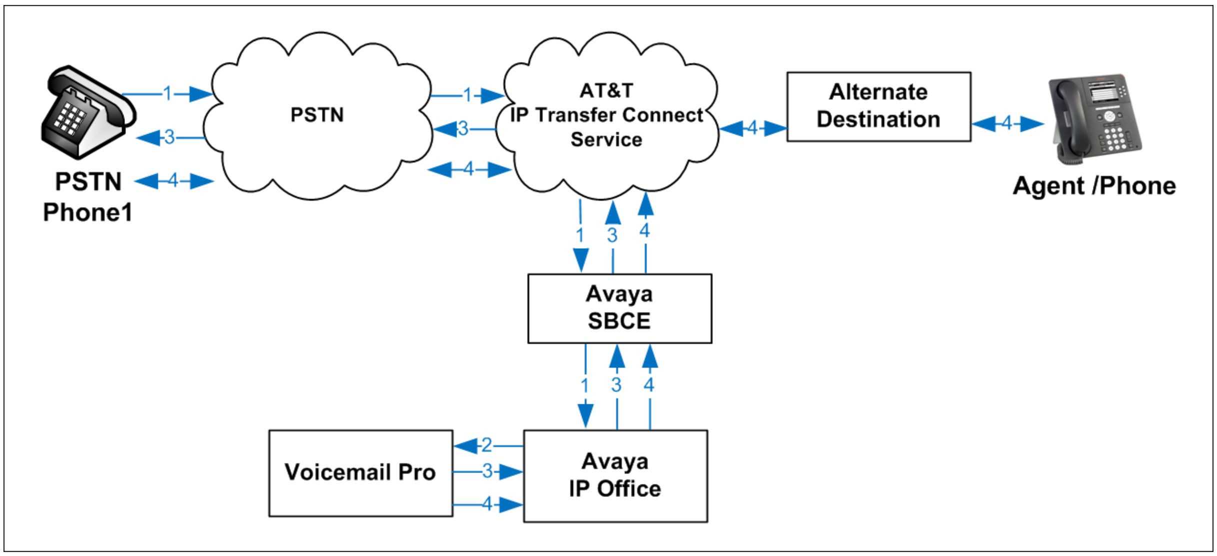

Figure 4: Inbound IPTC Call –SIP REFER Redirection

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

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4. Equipment and Software Validated

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

Equipment/Software	**Release/Version**
Avaya	
Avaya IP Office Server Edition - Avaya IP Office Voicemail Pro	Release 11.1.0.1.0 Build 95 Release 11.1.0.1.0 Build 14
Avaya IP Office 500 V2 Expansion System	Release 11.1.0.1.0 Build 95
Avaya IP Office Manager	Release 11.1.0.1.0 Build 95
Avaya Session Border Controller for Enterprise	Release 8.1.1.0-26-19214 Patch 19242
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 4.0.6.0.7
Avaya Workplace Client for Windows	Release 3.11.0.44.25
Avaya 9508 Digital Deskphone	Release 0.60

Table 2: Equipment and Software Versions

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

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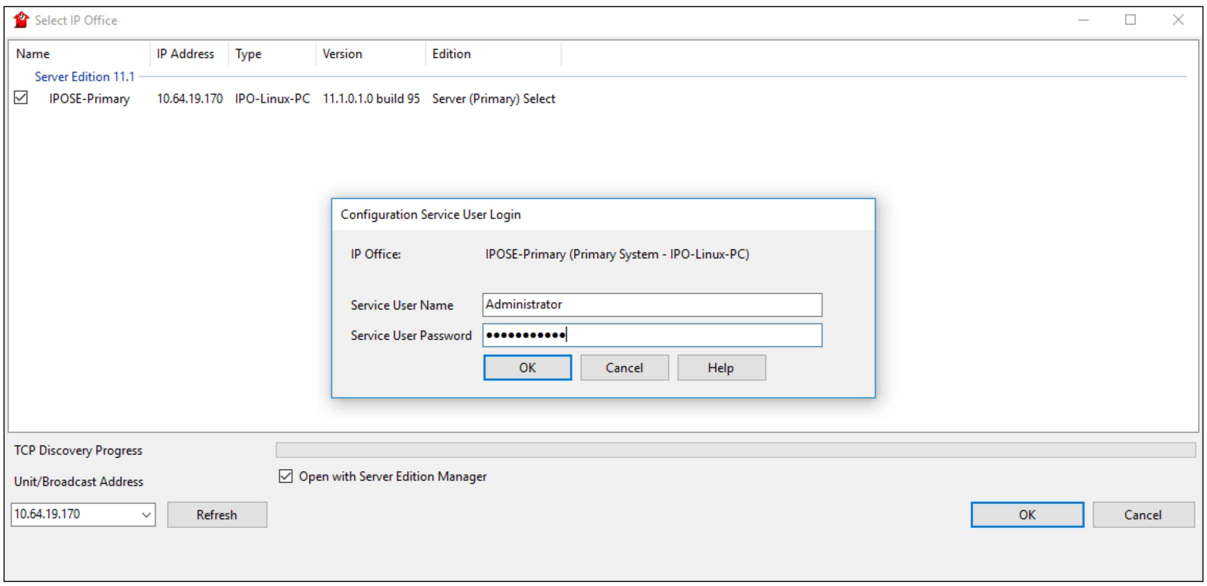
5. Avaya IP Office Configuration

Note – Avaya IP Office administration for interaction with the AT&T IP Toll Free service is described in document **[9]** 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 **[4]**. 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.



The appearance of the IP Office Manager can be customized using the **View** menu. In the screens presented in this section, 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.

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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.

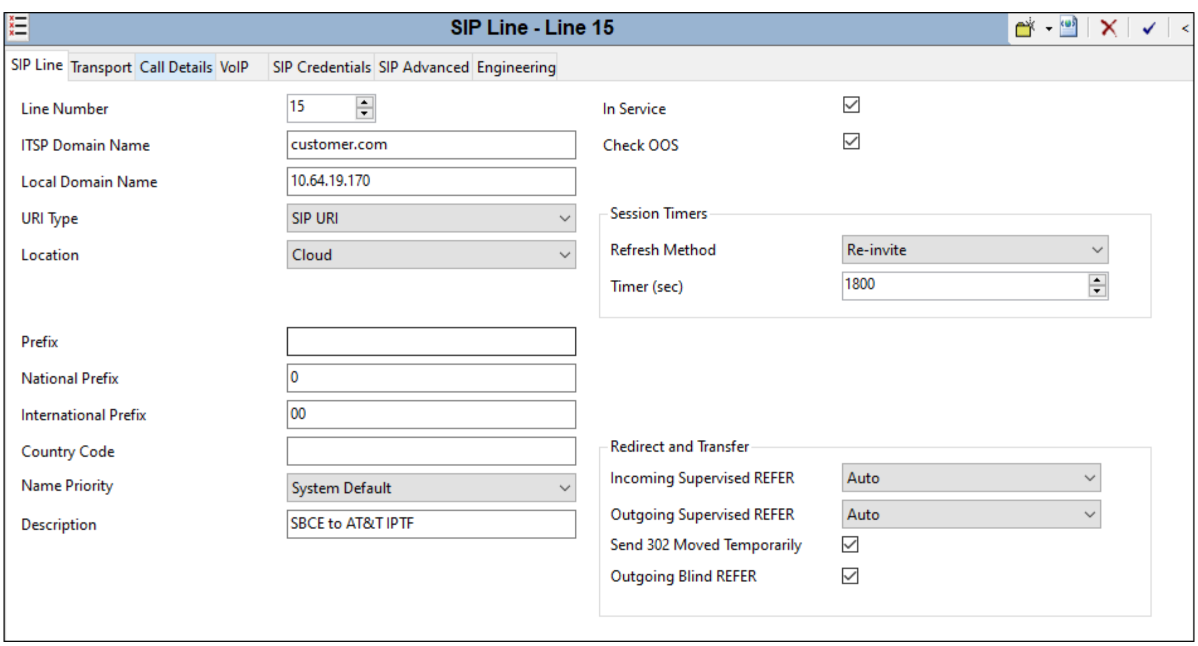
Follow the steps in **Sections 5.5.1** of document **[9]** to import/create a SIP Trunk from the template.

Note - In document **[9]**, SIP Line **15** was created for use with the AT&T IP Toll Free service. SIP Line 15 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 15** (see the note above). This SIP Line form is modified as follows for the IPTC service:

- Under **ITSP Domain Name**, a fictitious domain was entered. See **Section 2.2** item **2** for details.
- Check the box next to **Send 302 Moved Temporarily**.
- Check the box next to **Outgoing Blind Refer**.
- Click on **OK**.



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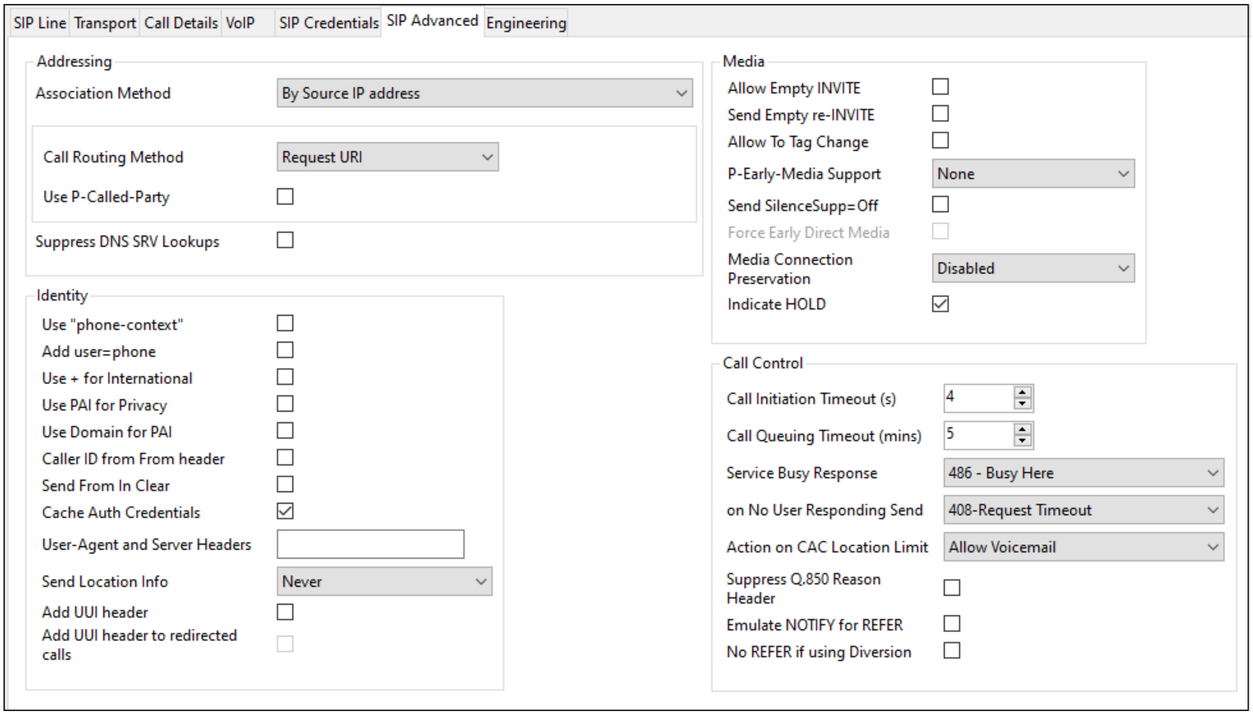
5.1.2. SIP Line – SIP Advanced Tab

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

- Under the **Call Control** section, verify that the **Emulate NOTIFY for REFER** is *not* checked.
- Click on **OK**.



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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.

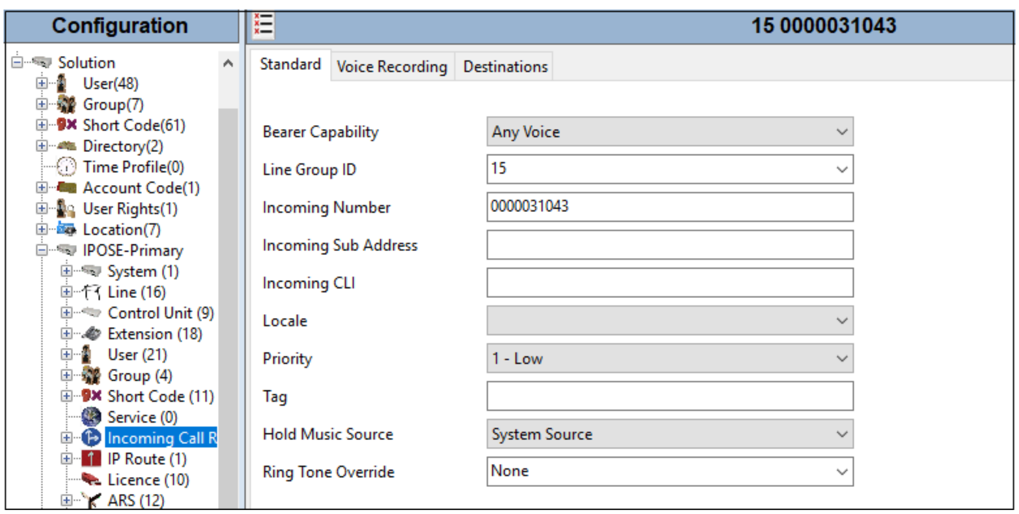
Note – In the reference configuration the IPTC service provided the access number 11041 for use in the 302 and Refer testing.

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 **0000031043** 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 5.5** of document **[9]** (e.g., **15**).
   - **Incoming Number**: Enter the associated DNIS digits sent by AT&T (e.g., **0000031043**).
   - Use default values for the remaining fields.



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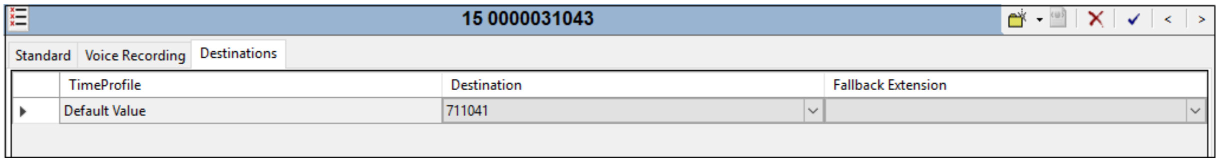
2. Select the **Destinations** tab and enter the following:
   - Enter the string **711041** to the drop down menu, and click **OK** (not shown).

In this example, **7** is the outbound dialing Short Code (see **Section 5.4**), and **11041** is the IPTC defined access number to be used for the call redirection.

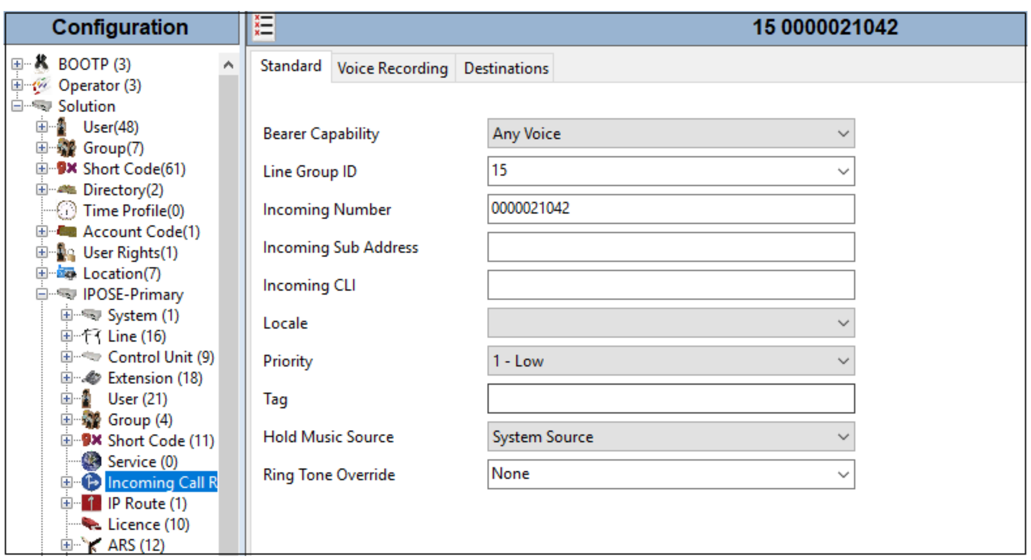


When the 0000031043 number is received in an Invite, Avaya IP Office will generate a 302 message, with 11041 in the Contact header, back to the IPTC service. The IPTC service will then generate a new Invite to the 11041 destination.

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:
   - **Line Group ID**: Enter the SIP Line previously defined in **Section 5.5** of document **[9]** (e.g., **15**).
   - **Incoming Number**: Enter the associated DNIS digits sent by AT&T (e.g., **0000021042**).
   - Use default values for the remaining fields.



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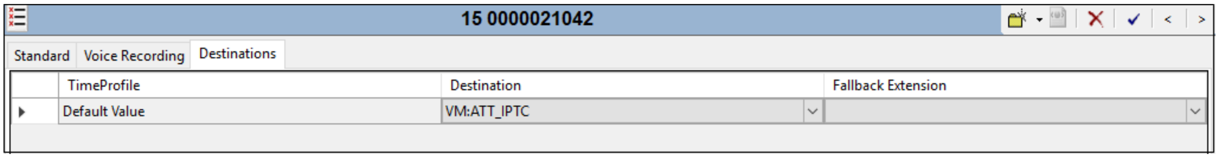
2. Select the **Destinations** tab and enter the following:
   - In the **Destinations** column, enter the string **VM:ATT_IPTC** to the drop down menu, and click **OK** (not shown).

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**).



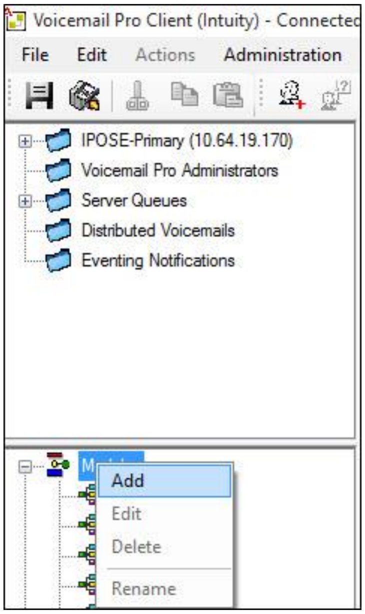
When the 0000021042 number is received in an Invite, Avaya IP Office/Voicemail Pro will play and announcement to the caller, then generate a Refer (without Replaces) message, (with 11041 in the Refer-To header), back to the IPTC service. The IPTC service will then generate a new Invite to the 11041 destination.

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**.



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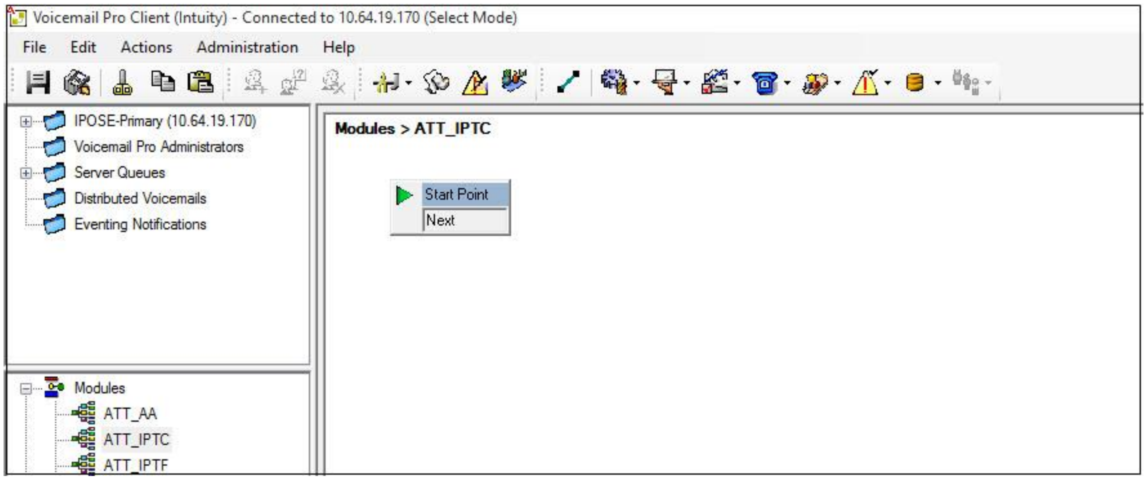
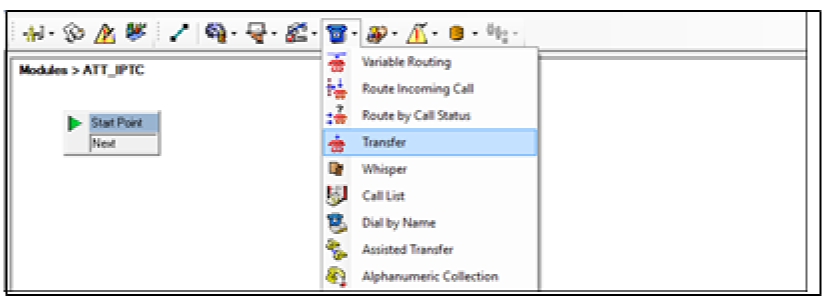
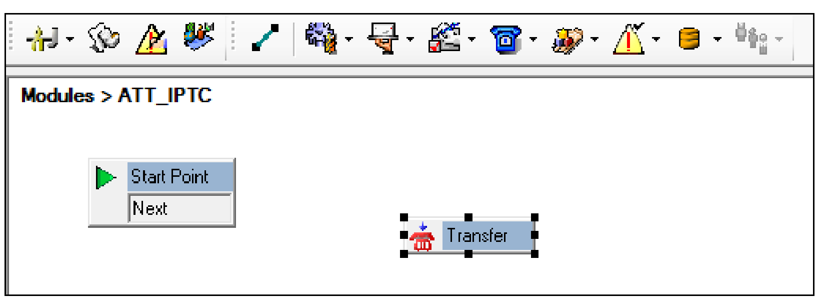
3. Enter a name (e.g., **ATT_IPTC**) and click on **OK** (not shown). The new script "ATT_IPTC" will appear under Modules and a Start Point icon will appear in the work area.



- 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.





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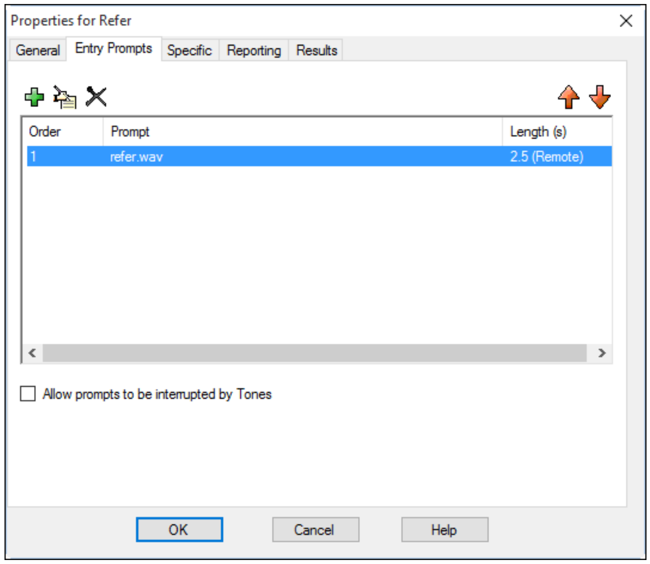
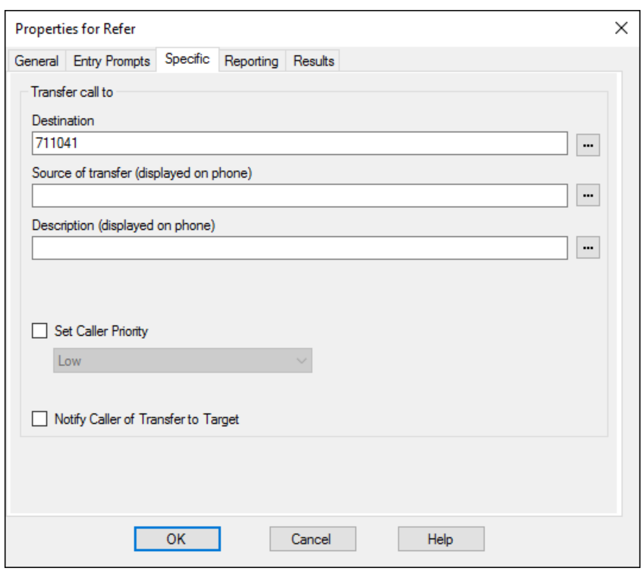
4. Double click on the **Transfer** icon. On the **General** tab → **Token Name** field, enter **Refer** (not shown).
5. Select the **Entry Prompts** tab and select or create an announcement to be played to the caller prior to the Refer (e.g., **refer.wav**). To modify an existing recording, double click 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 **711041**, where **7** is the Avaya IP Office outbound Short Code, and **11041** is the redirection number specified by the IPTC service (see **Section 5.2.2**).
7. Click on **OK**.



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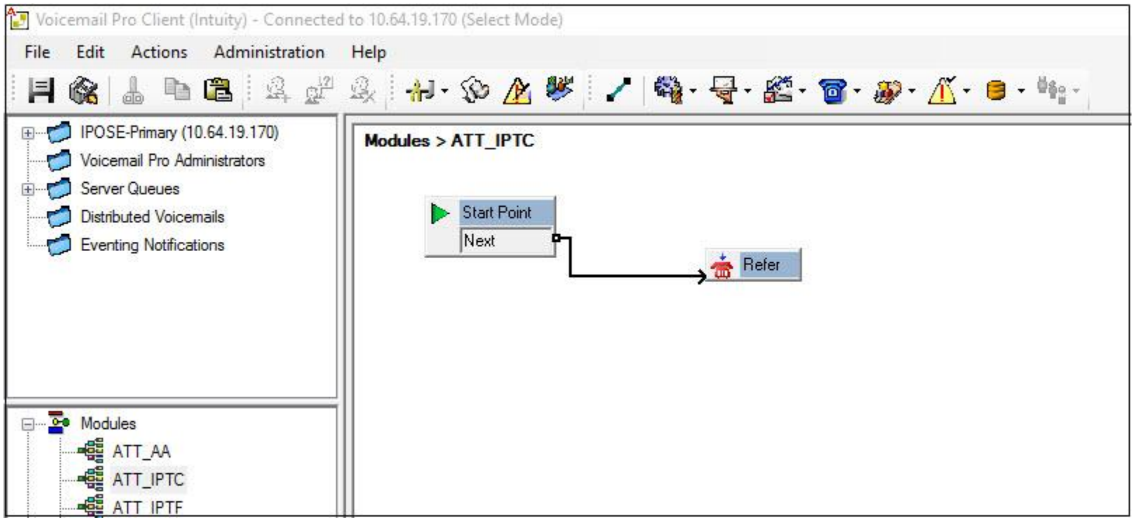
8. From the options bar, select the **Connector** icon  and drag a connecting flow line from the **Start Point** box to the **Transfer** box.



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

When the IPTC DNIS number is received (e.g., **0000021042**), 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 **11041** in the Refer-To header. The IPTC service will then send a new Invite to the 11041 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**.

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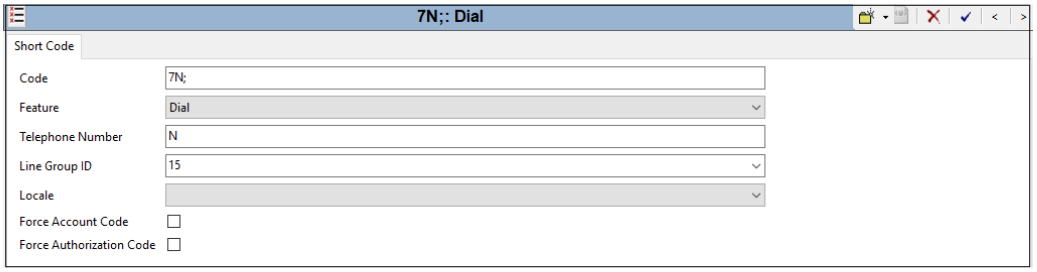
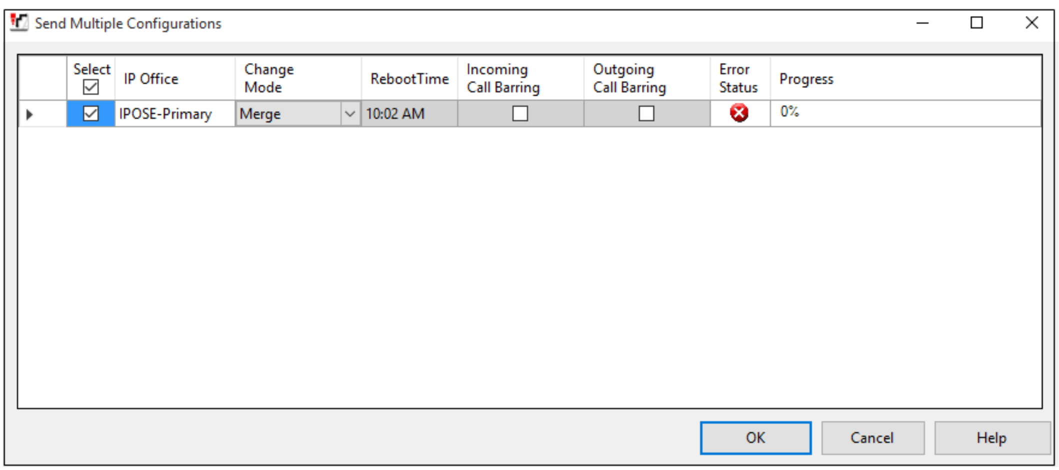
- The **Line Group ID** parameter is set to the SIP Line previously defined in **Section 5.4** of document **[9]** (e.g., **15**).
- Click the **OK** button (not shown).



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 **Reboot** 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 **Reboot** option will save the configuration and cause the Avaya IP Office server to reboot. Click **OK** to execute the save.



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6. Avaya Session Border Controller for Enterprise

Avaya SBCE configuration for interaction with the AT&T IP Toll Free service provided in document **[9]** 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 SBCE public (B1) 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 R11.1 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 **[9]**, 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**.

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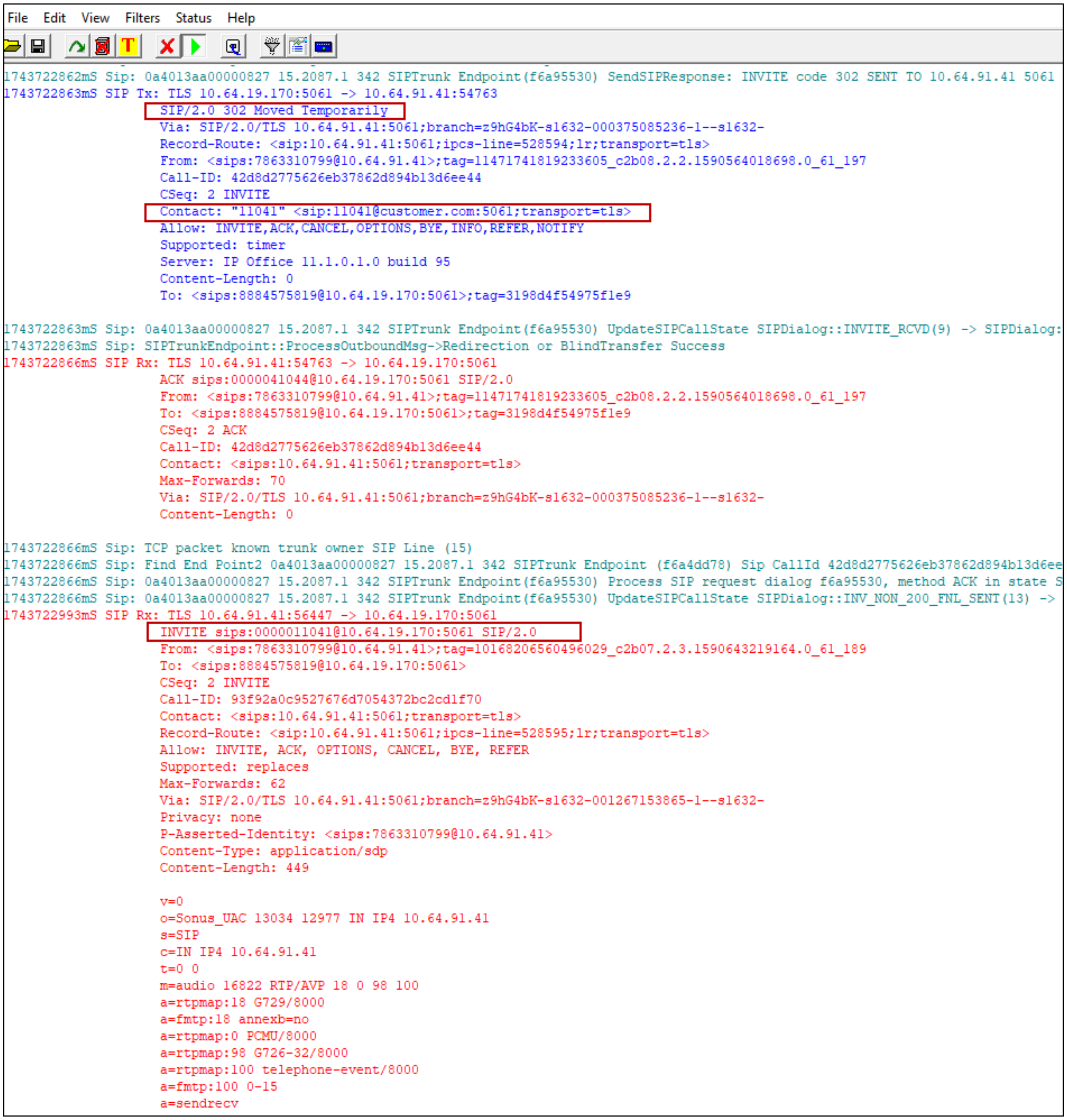
8.2.1. 302 Redirection

The following is an example of a 302 redirection captured in System Monitor..
The Contact header contains the new destination number (11041) as defined in the Avaya IP Office Incoming Call Route Destination field (see **Section 5.2.1)**. The IPTC service will then generate a new Invite to the 11041 destination.




```
1743722862mS Sip: 0a4013aa00000827 15.2087.1 342 SIPTrunk Endpoint(f6a95530) SendSIPResponse: INVITE code 302 SENT TO 10.64.91.41 5061
1743722863mS SIP Tx: TLS 10.64.19.170:5061 -> 10.64.91.41:54763
                  SIP/2.0 302 Moved Temporarily
                  Via: SIP/2.0/TLS 10.64.91.41:5061;branch=z9hG4bK-s1632-000375085236-1--s1632-
                  Record-Route: <sip:10.64.91.41:5061;ipcs-line=528594;lr;transport=tls>
                  From: <sips:7863310799@10.64.91.41>;tag=11471741819233605_c2b08.2.2.1590564018698.0_61_197
                  Call-ID: 42d8d2775626eb37862d894b13d6ee44
                  CSeq: 2 INVITE
                  Contact: "11041" <sip:11041@customer.com:5061;transport=tls>
                  Allow: INVITE,ACK,CANCEL,OPTIONS,BYE,INFO,REFER,NOTIFY
                  Supported: timer
                  Server: IP Office 11.1.0.1.0 build 95
                  Content-Length: 0
                  To: <sips:8884575819@10.64.19.170:5061>;tag=3198d4f54975f1e9

1743722863mS Sip: 0a4013aa00000827 15.2087.1 342 SIPTrunk Endpoint(f6a95530) UpdateSIPCallState SIPDialog::INVITE_RCVD(9) -> SIPDialog:
1743722863mS Sip: SIPTrunkEndpoint::ProcessOutboundMsg->Redirection or BlindTransfer Success
1743722866mS SIP Rx: TLS 10.64.91.41:54763 -> 10.64.19.170:5061
                  ACK sips:0000041044@10.64.19.170:5061 SIP/2.0
                  From: <sips:7863310799@10.64.91.41>;tag=11471741819233605_c2b08.2.2.1590564018698.0_61_197
                  To: <sips:8884575819@10.64.19.170:5061>;tag=3198d4f54975f1e9
                  CSeq: 2 ACK
                  Call-ID: 42d8d2775626eb37862d894b13d6ee44
                  Contact: <sips:10.64.91.41:5061;transport=tls>
                  Max-Forwards: 70
                  Via: SIP/2.0/TLS 10.64.91.41:5061;branch=z9hG4bK-s1632-000375085236-1--s1632-
                  Content-Length: 0

1743722866mS Sip: TCP packet known trunk owner SIP Line (15)
1743722866mS Sip: Find End Point2 0a4013aa00000827 15.2087.1 342 SIPTrunk Endpoint (f6a4dd78) Sip CallId 42d8d2775626eb37862d894b13d6ee
1743722866mS Sip: 0a4013aa00000827 15.2087.1 342 SIPTrunk Endpoint(f6a95530) Process SIP request dialog f6a95530, method ACK in state S
1743722866mS Sip: 0a4013aa00000827 15.2087.1 342 SIPTrunk Endpoint(f6a95530) UpdateSIPCallState SIPDialog::INV_NON_200_FNL_SENT(13) ->
1743722993mS SIP Rx: TLS 10.64.91.41:56447 -> 10.64.19.170:5061
                  INVITE sips:0000011041@10.64.19.170:5061 SIP/2.0
                  From: <sips:7863310799@10.64.91.41>;tag=10168206560496029_c2b07.2.3.1590643219164.0_61_189
                  To: <sips:8884575819@10.64.19.170:5061>
                  CSeq: 2 INVITE
                  Call-ID: 93f92a0c9527676d7054372bc2cd1f70
                  Contact: <sips:10.64.91.41:5061;transport=tls>
                  Record-Route: <sip:10.64.91.41:5061;ipcs-line=528595;lr;transport=tls>
                  Allow: INVITE, ACK, OPTIONS, CANCEL, BYE, REFER
                  Supported: replaces
                  Max-Forwards: 62
                  Via: SIP/2.0/TLS 10.64.91.41:5061;branch=z9hG4bK-s1632-001267153865-1--s1632-
                  Privacy: none
                  P-Asserted-Identity: <sips:7863310799@10.64.91.41>
                  Content-Type: application/sdp
                  Content-Length: 449

                  v=0
                  o=Sonus_UAC 13034 12977 IN IP4 10.64.91.41
                  s=SIP
                  c=IN IP4 10.64.91.41
                  t=0 0
                  m=audio 16822 RTP/AVP 18 0 98 100
                  a=rtpmap:18 G729/8000
                  a=fmtp:18 annexb=no
                  a=rtpmap:0 PCMU/8000
                  a=rtpmap:98 G726-32/8000
                  a=rtpmap:100 telephone-event/8000
                  a=fmtp:100 0-15
                  a=sendrecv
```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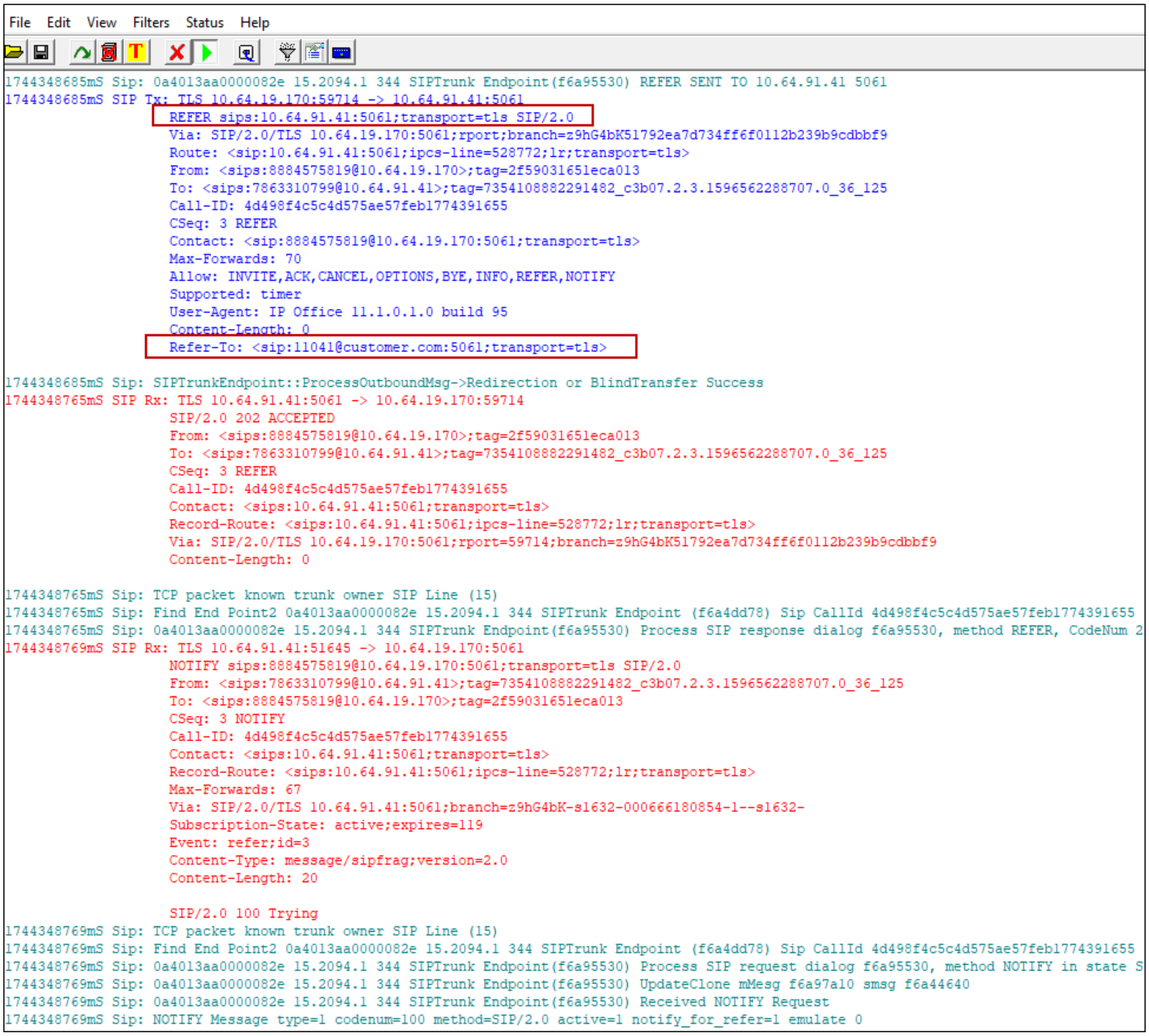
8.2.2. Refer

The following is an example of a Refer redirection.

- The Refer-To header contains the new destination number (11041) 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.



```
File  Edit  View  Filters  Status  Help
1744348685mS Sip: 0a4013aa0000082e 15.2094.1 344 SIPTrunk Endpoint(f6a95530) REFER SENT TO 10.64.91.41 5061
1744348685mS SIP Tx: TLS 10.64.19.170:59714 -> 10.64.91.41:5061
                    REFER sips:10.64.91.41:5061;transport=tls SIP/2.0
                    Via: SIP/2.0/TLS 10.64.19.170:5061;rport;branch=z9hG4bK51792ea7d734ff6f0112b239b9cdbbf9
                    Route: <sip:10.64.91.41:5061;ipcs-line=528772;lr;transport=tls>
                    From: <sips:8884575819@10.64.19.170>;tag=2f59031651eca013
                    To: <sips:7863310799@10.64.91.41>;tag=7354108882291482_c3b07.2.3.1596562288707.0_36_125
                    Call-ID: 4d498f4c5c4d575ae57feb1774391655
                    CSeq: 3 REFER
                    Contact: <sip:8884575819@10.64.19.170:5061;transport=tls>
                    Max-Forwards: 70
                    Allow: INVITE,ACK,CANCEL,OPTIONS,BYE,INFO,REFER,NOTIFY
                    Supported: timer
                    User-Agent: IP Office 11.1.0.1.0 build 95
                    Content-Length: 0
                    Refer-To: <sip:11041@customer.com:5061;transport=tls>

1744348685mS Sip: SIPTrunkEndpoint::ProcessOutboundMsg->Redirection or BlindTransfer Success
1744348765mS SIP Rx: TLS 10.64.91.41:5061 -> 10.64.19.170:59714
                    SIP/2.0 202 ACCEPTED
                    From: <sips:8884575819@10.64.19.170>;tag=2f59031651eca013
                    To: <sips:7863310799@10.64.91.41>;tag=7354108882291482_c3b07.2.3.1596562288707.0_36_125
                    CSeq: 3 REFER
                    Call-ID: 4d498f4c5c4d575ae57feb1774391655
                    Contact: <sips:10.64.91.41:5061;transport=tls>
                    Record-Route: <sips:10.64.91.41:5061;ipcs-line=528772;lr;transport=tls>
                    Via: SIP/2.0/TLS 10.64.19.170:5061;rport=59714;branch=z9hG4bK51792ea7d734ff6f0112b239b9cdbbf9
                    Content-Length: 0

1744348765mS Sip: TCP packet known trunk owner SIP Line (15)
1744348765mS Sip: Find End Point2 0a4013aa0000082e 15.2094.1 344 SIPTrunk Endpoint (f6a4dd78) Sip CallId 4d498f4c5c4d575ae57feb1774391655
1744348765mS Sip: 0a4013aa0000082e 15.2094.1 344 SIPTrunk Endpoint(f6a95530) Process SIP response dialog f6a95530, method REFER, CodeNum 2
1744348769mS SIP Rx: TLS 10.64.91.41:51645 -> 10.64.19.170:5061
                    NOTIFY sips:8884575819@10.64.19.170:5061;transport=tls SIP/2.0
                    From: <sips:7863310799@10.64.91.41>;tag=7354108882291482_c3b07.2.3.1596562288707.0_36_125
                    To: <sips:8884575819@10.64.19.170>;tag=2f59031651eca013
                    CSeq: 3 NOTIFY
                    Call-ID: 4d498f4c5c4d575ae57feb1774391655
                    Contact: <sips:10.64.91.41:5061;transport=tls>
                    Record-Route: <sips:10.64.91.41:5061;ipcs-line=528772;lr;transport=tls>
                    Max-Forwards: 67
                    Via: SIP/2.0/TLS 10.64.91.41:5061;branch=z9hG4bK-s1632-000666180854-1--s1632-
                    Subscription-State: active;expires=119
                    Event: refer;id=3
                    Content-Type: message/sipfrag;version=2.0
                    Content-Length: 20

                    SIP/2.0 100 Trying
1744348769mS Sip: TCP packet known trunk owner SIP Line (15)
1744348769mS Sip: Find End Point2 0a4013aa0000082e 15.2094.1 344 SIPTrunk Endpoint (f6a4dd78) Sip CallId 4d498f4c5c4d575ae57feb1774391655
1744348769mS Sip: 0a4013aa0000082e 15.2094.1 344 SIPTrunk Endpoint(f6a95530) Process SIP request dialog f6a95530, method NOTIFY in state S
1744348769mS Sip: 0a4013aa0000082e 15.2094.1 344 SIPTrunk Endpoint(f6a95530) UpdateClone mMesg f6a97a10 smsg f6a44640
1744348769mS Sip: 0a4013aa0000082e 15.2094.1 344 SIPTrunk Endpoint(f6a95530) Received NOTIFY Request
1744348769mS Sip: NOTIFY Message type=1 codenum=100 method=SIP/2.0 active=1 notify_for_refer=1 emulate 0
```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9. Conclusion

As illustrated in these Application Notes, Avaya IP Office Release 11.1, and the Avaya Session Border Controller for Enterprise Release 8.1 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.

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10. References

Avaya:

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

[1] *Deploying IP Office Platform Server Edition Solution*, Release 11.1, Issue 14, April 2020.
[2] *IP Office™ Platform 11.0, Deploying Avaya IP Office Servers as Virtual Machines,* August 2020.
[3] *IP Office™ Platform 11.1, Deploying an IP500 V2 IP Office Essential Edition System*, September 2020.
[4] *Administering Avaya IP Office™ Platform with Manager*, Release 11.1 SP1, July 2020.
[5] *IP Office™ Platform 11.1, Administering Avaya IP Office Platform Voicemail Pro*, September 2020.
[6] *IP Office™ Platform 11.1, IP Office SIP Phones with ASBCE, Issue 04c*, July 2020
[7] *Deploying Avaya Session Border Controller for Enterprise on a Virtualized Environment Platform, Release 8.1.x*, August 2020.
[8] *Administering Avaya Session Border Controller for Enterprise*, Release 8.1.x, August 2020.

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

[9] *Application Notes for Avaya IP Office Release 11.1 and, Avaya Session Border Controller for Enterprise Release 8.1 with AT&T IP Toll Free Service – Issue 1.0*

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

AT&T IP Transfer Connect Service:

[10] AT&T IP Transfer Connect service description -
https://www.business.att.com/products/ip-toll-free.html

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