



Avaya Solution & Interoperability Test Lab

Application Notes for Avaya IP Office Release 10.1, Avaya Session Border Controller for Enterprise Release 7.2 with AT&T IP Transfer Connect Service – Issue 1.0

Abstract

These Application Notes describe the steps for configuring Avaya IP Office Release 10.1 and Avaya Session Border Controller for Enterprise Release 7.2, with the AT&T IP Transfer Connect service 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 Release 10.1.

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.

Note that these Application Notes are intended to supplement the separate document: *Application Notes for Avaya IP Office Release 10.1, Avaya Session Border Controller for Enterprise Release 7.2 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 10.1 and Avaya Session Border Controller for Enterprise Release 7.2, with the AT&T IP Transfer Connect service and **AVPN** or **MIS/PNT** transport connections¹.

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 10.1 and the Avaya Session Border Controller for Enterprise Release 7.2 (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 alternate² 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 10.1, Avaya Session Border Controller for Enterprise Release 7.2 with the AT&T IP Toll Free Service – Issue 1.0*. This document is listed in **Section 10** as reference document [7]. 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 10.1 (see **Section 3.2** for call flow examples).

The test environment described in these Application Notes consisted of:

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.

- 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.

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, and Communicator for Windows softphone)/H.323 (1616, and 9611)/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.
- Inbound IPTC INFOPAK data transmission in SIP messaging.
- Avaya IP Office features such as hold, resume, and local transfer.
- IP 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 msec, 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 (UI) data.** The IPTC service allows for the optional inclusion of UI data in both 302 and Refer SIP messages. Although Avaya IP Office 10.1 supports sending UI data across a SIP trunk, it does not send any UI data in the configuration tested.
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.

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:

- Avaya IP Office provides the voice communications services for a particular enterprise site. In the reference configuration, Avaya IP Office runs on the IP 500 V2 platform. This solution is extensible to the Avaya IP Office Server Edition platform as well.
- Voicemail Pro (running on the Application Server) 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 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.

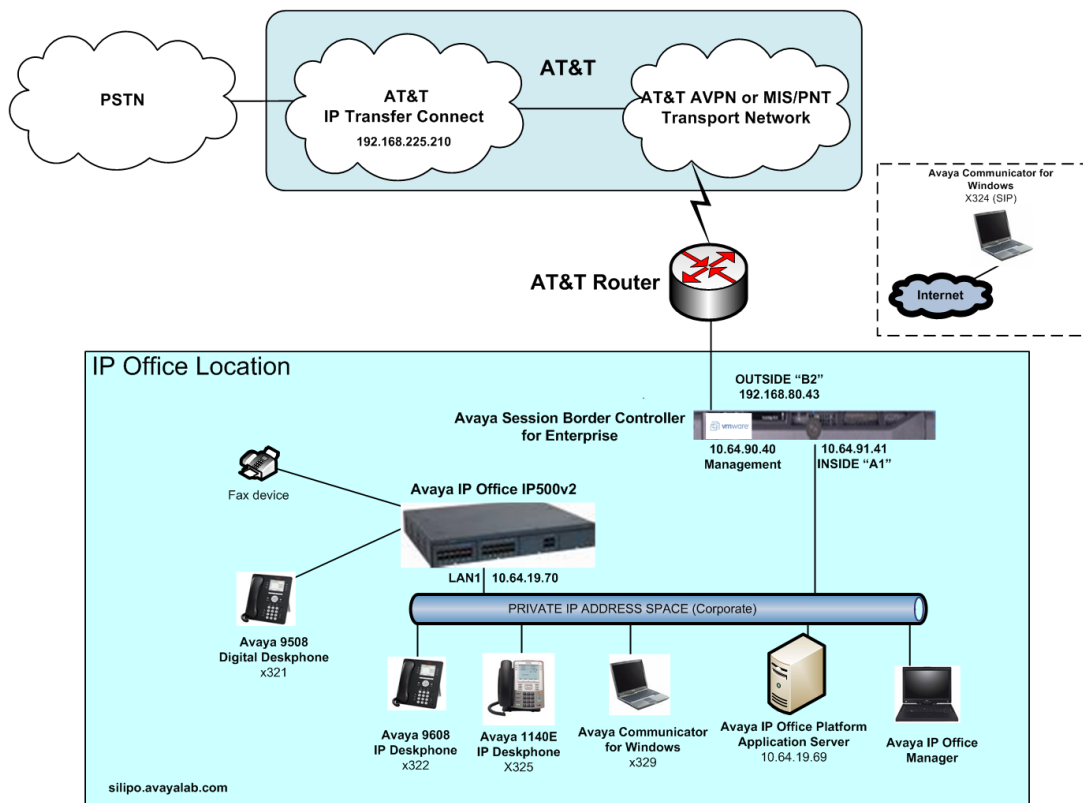


Figure 1: Reference Configuration

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 **192.168.80.43** (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	10.64.19.70
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 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.

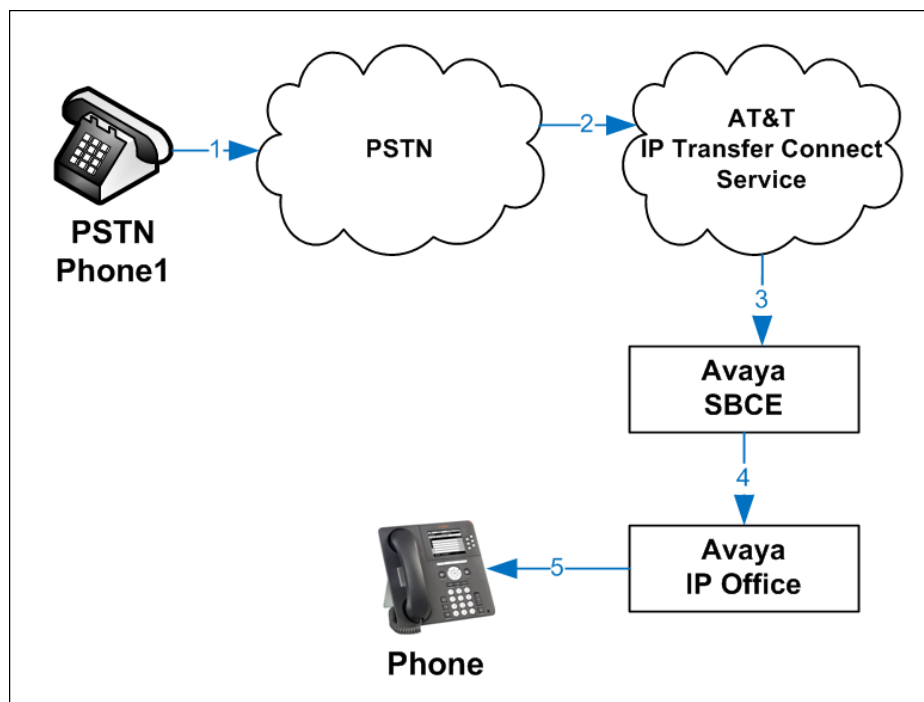


Figure 2: Inbound IPTC Call – No Redirection

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.

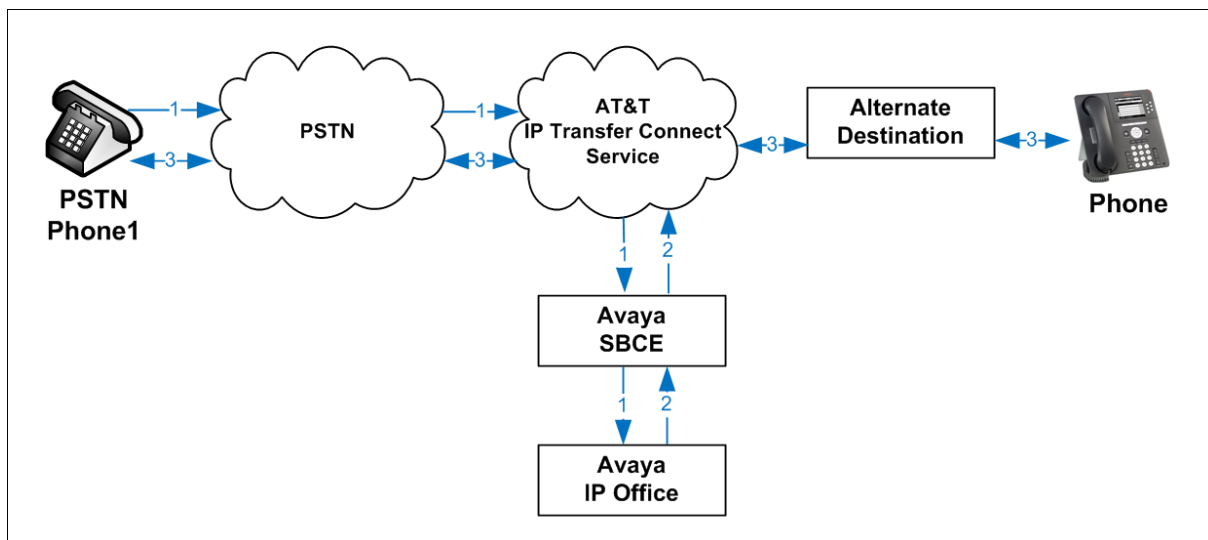


Figure 3: Inbound IPTC Call – SIP 302 Redirection

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 Refer³, 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).

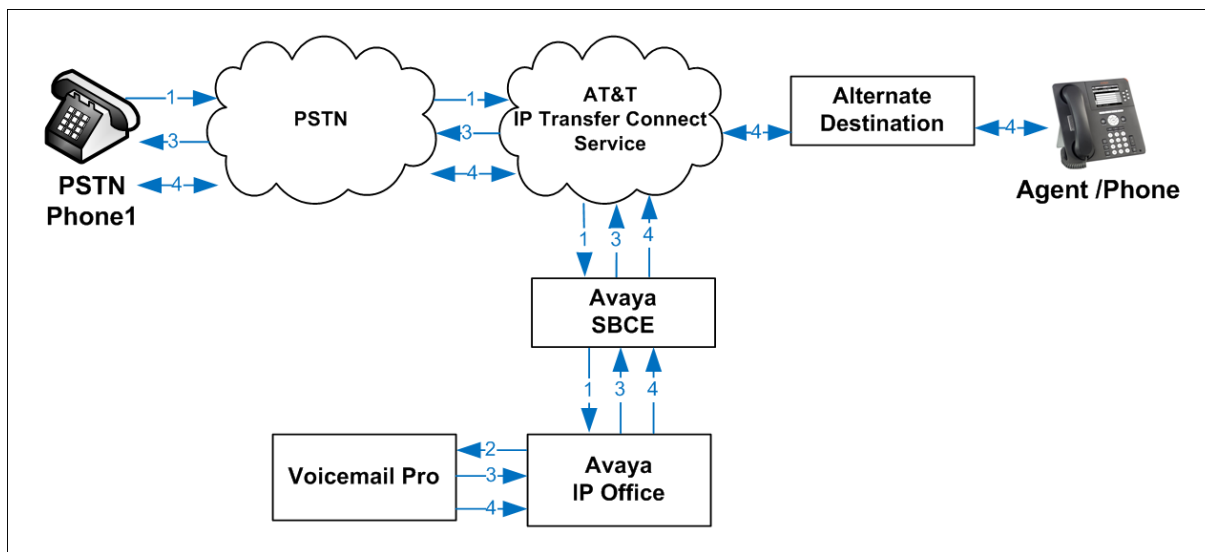


Figure 4: Inbound IPTC Call –SIP REFER Redirection

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

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 Session Border Controller for Enterprise	Release 7.2.1.0-05-14222
Avaya IP Office IP500 V2 <ul style="list-style-type: none">IP OfficeAvaya IP Office TCM 8Avaya IP Office COMBO6210/ATM4	Release 10.1.0.1.0 build 3 Release 10.1.0.1.0 build 3 Release 10.1.0.1.0 build 3
Avaya IP Office Platform Application Server <ul style="list-style-type: none">Voicemail ProAvaya WebRTC GatewayAvaya one-X® Portal for IP Office	Release 10.1.0.1.0 build 3 Release 10.1.0.1.0 build 3 Release 10.1.0.1.0 build 3
Avaya IP Office Manager	Release 10.1.0.1.0 build 3
Avaya 9611SW IP Deskphone (H.323)	Release 6.6506
Avaya 1140E IP Deskphone (SIP)	Release 04.04.23
Avaya 9508 Digital Telephone	Release 0.60
Avaya Communicator for Windows	Release 2.1.4.256
Analog Fax device	Ventafax 7.9

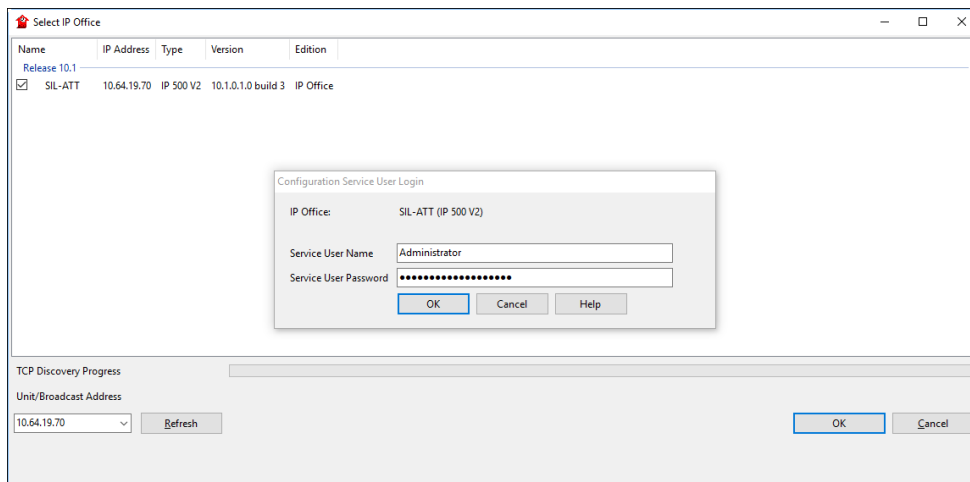
Table 2: Equipment and Software Versions

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

5. Avaya IP Office Configuration

Note – Avaya IP Office administration for interaction with the AT&T IP Toll Free service is described in document [7] 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.



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.

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 [7] to import/create a SIP Trunk from the template.

Note - In document [7], 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:

- Check the box next to **Send 302 Moved Temporarily**.
- Check the box next to **Outgoing Blind Refer**.
- Click on OK.

SIP Line - Line 15

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

Line Number: 15
ITSP Domain Name: 10.64.91.41
Local Domain Name: 10.64.19.70
URI Type: SIP
Location: Cloud
Prefix:
National Prefix: 0
International Prefix: 00
Country Code:
Name Priority: System Default
Description: SBCE to AT&T IPTF

In Service: ☒
Check OOS: ☒

Session Timers
Refresh Method: Re-invite
Timer (sec): 1800

Redirect and Transfer
Incoming Supervised REFER: Auto
Outgoing Supervised REFER: Auto
Send 302 Moved Temporarily: ☒
Outgoing Blind REFER: ☒

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.

The screenshot shows the 'SIP Line - SIP Advanced' configuration window. The 'Addressing' section includes 'Association Method' (By Source IP address), 'Call Routing Method' (Request URI), and 'Suppress DNS SRV Lookups' (unchecked). The 'Identity' section includes various checkboxes for phone context, user agent, and headers, with 'Cache Auth Credentials' checked. The 'Media' section includes checkboxes for empty INVITE, re-INVITE, tag change, and early media support, with 'Media Connection Preservation' set to 'Disabled'. The 'Call Control' section includes timeouts, service busy response, and action on CAC location limit. The 'Emulate NOTIFY for REFER' checkbox is highlighted with a red box.

Section	Option	Value/Status
Addressing	Association Method	By Source IP address
	Call Routing Method	Request URI
	Suppress DNS SRV Lookups	<input type="checkbox"/>
Identity	Use "phone-context"	<input type="checkbox"/>
	Add user=phone	<input type="checkbox"/>
	Use + for International	<input type="checkbox"/>
	Use PAI for Privacy	<input type="checkbox"/>
	Use Domain for PAI	<input type="checkbox"/>
	Swap From and PAI/Diversion	<input type="checkbox"/>
	Caller ID from From header	<input type="checkbox"/>
	Send From In Clear	<input type="checkbox"/>
	Cache Auth Credentials	<input checked="" type="checkbox"/>
	User-Agent and Server Headers	
Send Location Info	Never	
Add UUI header	<input checked="" type="checkbox"/>	
Add UUI header to redirected calls	<input checked="" type="checkbox"/>	
Media	Allow Empty INVITE	<input type="checkbox"/>
	Send Empty re-INVITE	<input type="checkbox"/>
	Allow To Tag Change	<input type="checkbox"/>
	P-Early-Media Support	None
	Send SilenceSupp=Off	<input type="checkbox"/>
	Force Early Direct Media	<input type="checkbox"/>
Media Connection Preservation	Disabled	
Indicate HOLD	<input checked="" type="checkbox"/>	
Call Control	Call Initiation Timeout (s)	4
	Call Queuing Timeout (mins)	5
	Service Busy Response	503 - Service Unavailable
	on No User Responding Send	408-Request Timeout
	Action on CAC Location Limit	Allow Voicemail
	Suppress Q.850 Reason Header	<input type="checkbox"/>
	Emulate NOTIFY for REFER	<input type="checkbox"/>
	No REFER if using Diversion	<input type="checkbox"/>

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 [7] (e.g., **15**).
 - **Incoming Number:** Enter the associated DNIS digits sent by AT&T (e.g., **000008885551025**).
 - Use default values for the remaining fields.

IP Offices		15 000008885551025	
		Standard	Voice Recording Destinations
BOOTP (23)		Bearer Capability	Any Voice
Operator (3)		Line Group ID	15
SIL-ATT		Incoming Number	000008885551025
System (1)		Incoming Sub Address	
Line (9)		Incoming CLI	
Control Unit (3)		Locale	
Extension (33)		Priority	1 - Low
User (35)		Tag	
Group (5)		Hold Music Source	System Source
Short Code (79)		Ring Tone Override	None
Service (0)			
RAS (1)			
Incoming Call			
WAN Port (0)			
Directory (0)			
Time Profile (0)			
Firewall Profile			
IP Route (4)			
Account Code			
License (34)			
Tunnel (0)			
User Rights (8)			

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.

TimeProfile	Destination	Fallback Extension
Default Value	71026	

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 the 1026 destination.

5.2.2.Refer Call Redirection

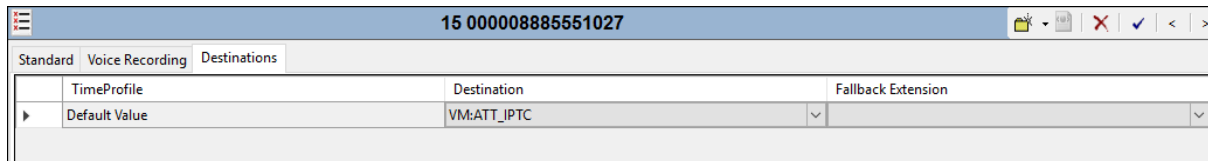
In the reference configuration, Voicemail Pro (running on Primary server), is used to send a Refer (without Replaces) Call Redirection. A Voicemail Pro “Module” is defined with the name **Refer** (see **Section 5.3**). This Module name is defined as a Destination to an inbound call as follows:

1. From the **Incoming Call Route** page, select the **Standard** tab enter the following:
 2. **Line Group ID:** Enter the SIP Line previously defined in **Section 5.4** of document [7] (e.g., **15**).
 - **Incoming Number:** Enter the associated DNIS digits sent by AT&T (e.g., **000008885551027**).
 - Use default values for the remaining fields.

15 000008885551027	
Bearer Capability	Any Voice
Line Group ID	15
Incoming Number	000008885551027
Incoming Sub Address	
Incoming CLI	
Locale	
Priority	1 - Low
Tag	
Hold Music Source	System Source
Ring Tone Override	None

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**).



TimeProfile	Destination	Fallback Extension
Default Value	VM:ATT_IPTC	

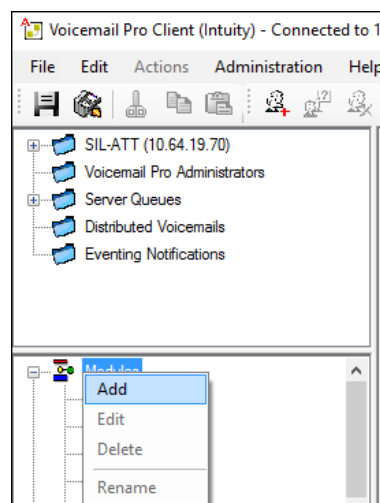
When the 000008885551027 number is received in an Invite, Avaya IP Office/Voicemail Pro will play an announcement to the caller, then generate a Refer (without Replaces) message, (with 1028 in the Refer-To header), back to the IPTC service. The IPTC service will then generate a new Invite to the 1028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

5.3. Voicemail Pro Refer Module

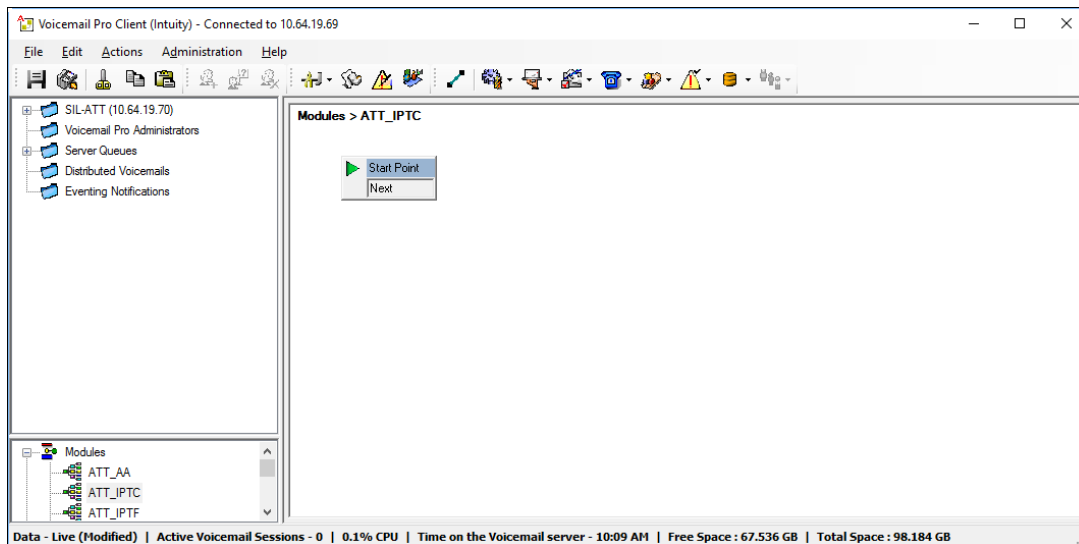
Note - While Avaya Voicemail Pro provisioning and programming is beyond the scope of this document, a sample Module is described below.

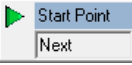


The Refer Module is provisioned to play an announcement to the caller, and then generate a Refer (without Replaces) back to the IPTC service. This is accomplished via the following steps via the Voicemail Pro Client interface:

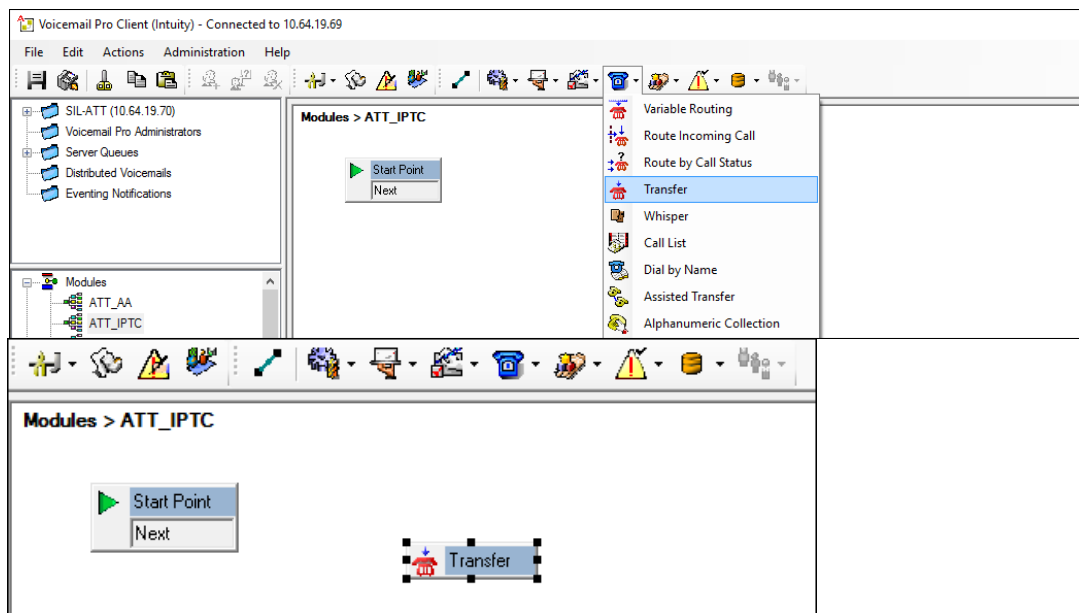
1. Open the **Voicemail Pro Client** application (not shown).
2. Create a **Start Point** by right clicking on **Modules** and selecting **Add**.




- 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.

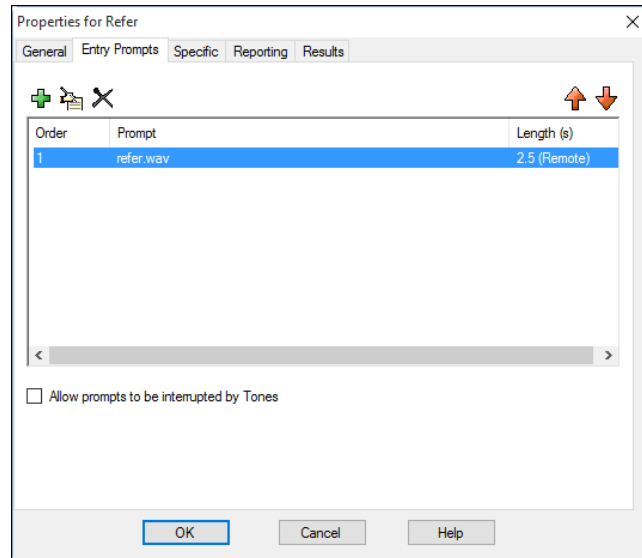


- Click on the **Start Point** icon  to activate the script options at the top of the screen. From the options, **Telephony Actions** icon , select the **Transfer** icon  and click on the work area to place the **Transfer** icon in the work area.

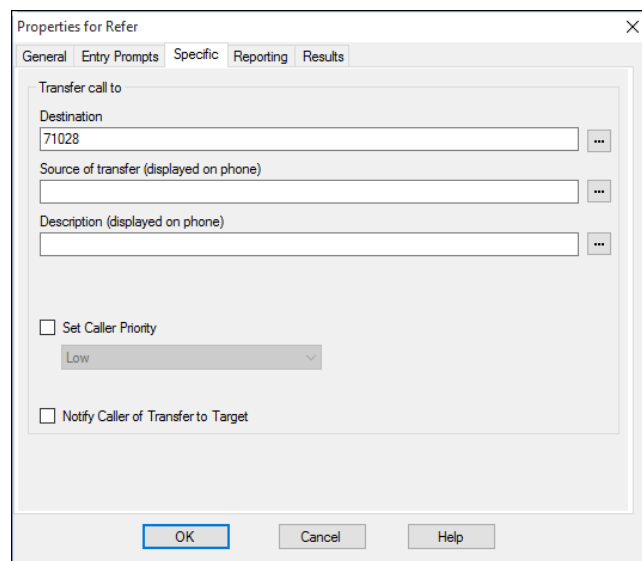


- Double click on the **Transfer** icon. On the **General** tab → **Token Name** field, enter **Refer** (not shown).

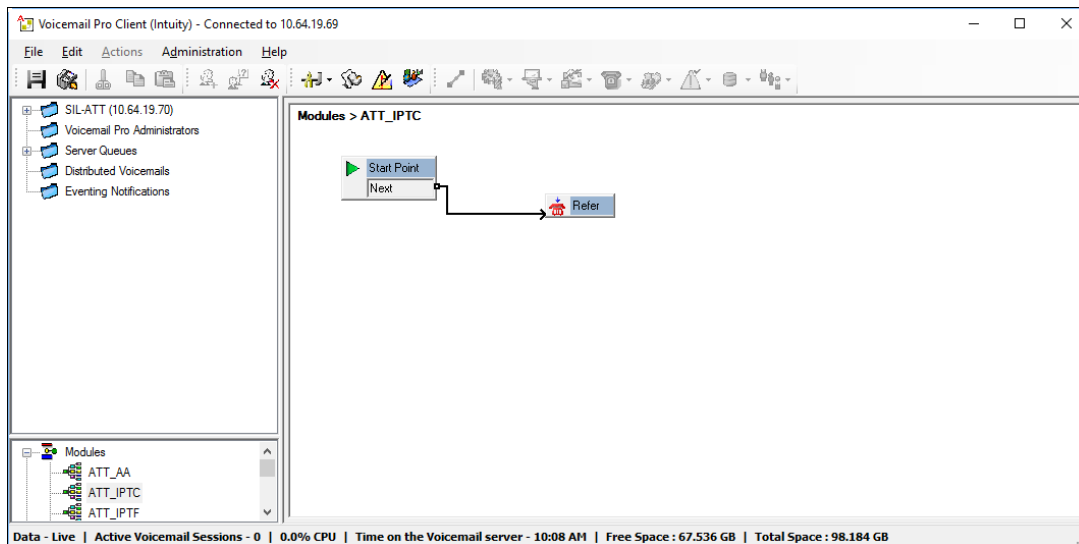
5. Select the **Entry Prompts** tab and select or create an announcement to be played to the caller prior to the Refer (e.g., **refer.wav**). To modify an existing recording, double click on the .wav file and rerecord. If no .wav files exist, double click on the  icon to open the .wav file editor.



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.way**), 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 [7] (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 **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 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 R10.1 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 [7], 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.

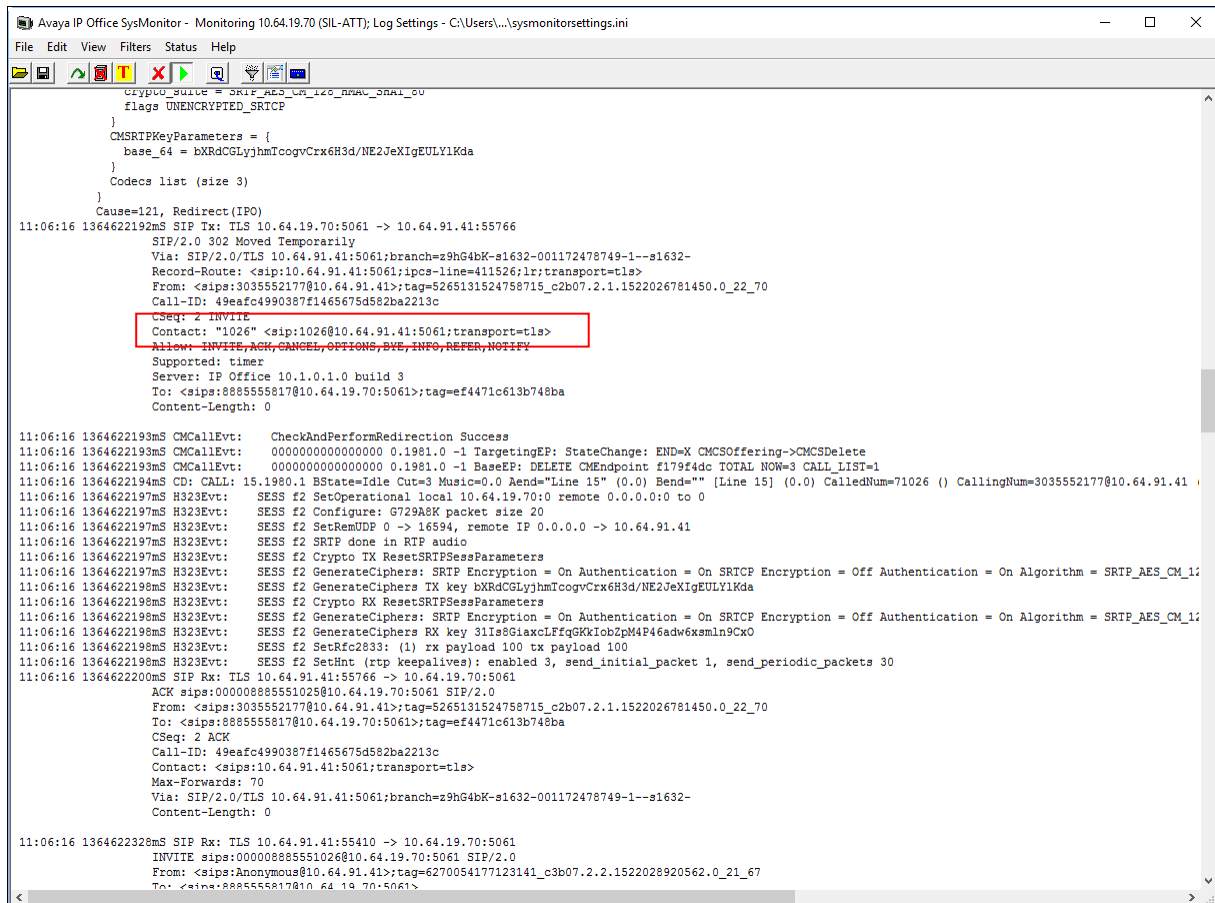
8.2. System Monitor Traces

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**).



```
Avaya IP Office SysMonitor - Monitoring 10.64.19.70 (SIL-ATT); Log Settings - C:\Users\...\sysmonitorsettings.ini
File Edit View Filters Status Help

crypto_suite = SRTP_AES_CM_128_HMAC_SHA1_64
flags UNENCRYPTED_SRTPC

}
CMSRTPKeyParameters = {
  base_64 = bXRdCGLyhmTcogvCrx6H3d/NE2JeXIgEULYlKda
}
Codecs list (size 3)
}
Cause=121, Redirect(IPO)
11:06:16 1364622192mS SIP Tx: TLS 10.64.19.70:5061 -> 10.64.91.41:55766
SIP/2.0 302 Moved Temporarily
Via: SIP/2.0/TLS 10.64.91.41:5061;branch=z9hG4bK-s1632-001172478749-1--s1632-
Record-Route: <sip:10.64.91.41:5061;ipcc=line=411526;lr;transport=tls>
From: <sips:3035552177@10.64.91.41>;tag=5265131524758715_c2b07.2.1.1522026781450.0_22_70
Call-ID: 49eaf04990387f1465675d582ba2213c
CSeq: 2 INVITE
Contact: "1026" <sip:1026@10.64.91.41:5061;transport=tls>
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, INFO, REFER, NOTIFY
Supported: timer
Server: IP Office 10.1.0.1.0 build 3
To: <sips:888555817@10.64.19.70:5061>;tag=ef4471c613b748ba
Content-Length: 0

11:06:16 1364622193mS CMCallEvt: CheckAndPerformRedirection Success
11:06:16 1364622193mS CMCallEvt: 0000000000000000 0.1981.0 -1 TargetingEP: StateChange: END=X CMCSOffering->CMCSDelete
11:06:16 1364622193mS CMCallEvt: 0000000000000000 0.1981.0 -1 BaseEP: DELETE CMEndpoint f179f4dc TOTAL NOW=3 CALL_LIST=1
11:06:16 1364622194mS CD: CALL: 15.1980.1 BState=Idle Cut=3 Music=0.0 Aend="Line 15" (0.0) Bend="" [Line 15] (0.0) CalledNum=71026 () CallingNum=3035552177@10.64.91.41
11:06:16 1364622197mS H323Evt: SESS f2 SetOperational local 10.64.19.70:0 remote 0.0.0.0:0 to 0
11:06:16 1364622197mS H323Evt: SESS f2 Configure: G729A8K packet size 20
11:06:16 1364622197mS H323Evt: SESS f2 SetRemUDP 0 -> 16594, remote IP 0.0.0.0 -> 10.64.91.41
11:06:16 1364622197mS H323Evt: SESS f2 SRTP done in RTP audio
11:06:16 1364622197mS H323Evt: SESS f2 Crypto TX ResetSRTPSessParameters
11:06:16 1364622197mS H323Evt: SESS f2 GenerateCiphers: SRTP Encryption = On Authentication = On SRTPC Encryption = Off Authentication = On Algorithm = SRTP_AES_CM_12
11:06:16 1364622198mS H323Evt: SESS f2 GenerateCiphers TX key bXRdCGLyhmTcogvCrx6H3d/NE2JeXIgEULYlKda
11:06:16 1364622198mS H323Evt: SESS f2 Crypto RX ResetSRTPSessParameters
11:06:16 1364622198mS H323Evt: SESS f2 GenerateCiphers: SRTP Encryption = On Authentication = On SRTPC Encryption = Off Authentication = On Algorithm = SRTP_AES_CM_12
11:06:16 1364622198mS H323Evt: SESS f2 GenerateCiphers RX key 31Is8GiacoLFfGKKlobZpM4P46adw6xsmIn9Cx0
11:06:16 1364622198mS H323Evt: SESS f2 SetRfc2833: (1) rx payload 100 tx payload 100
11:06:16 1364622198mS H323Evt: SESS f2 SetHnt (rtp keepalives): enabled 3, send_initial_packet 1, send_periodic_packets 30
11:06:16 1364622200mS SIP Rx: TLS 10.64.91.41:55766 -> 10.64.19.70:5061
ACK sips:000008885551025@10.64.19.70:5061 SIP/2.0
From: <sips:3035552177@10.64.91.41>;tag=5265131524758715_c2b07.2.1.1522026781450.0_22_70
To: <sips:888555817@10.64.19.70:5061>;tag=ef4471c613b748ba
CSeq: 2 ACK
Call-ID: 49eaf04990387f1465675d582ba2213c
Contact: <sips:10.64.91.41:5061;transport=tls>
Max-Forwards: 70
Via: SIP/2.0/TLS 10.64.91.41:5061;branch=z9hG4bK-s1632-001172478749-1--s1632-
Content-Length: 0

11:06:16 1364622328mS SIP Rx: TLS 10.64.91.41:55410 -> 10.64.19.70:5061
INVITE sips:000008885551026@10.64.19.70:5061 SIP/2.0
From: <sips:Anonymous@10.64.91.41>;tag=6270054177123141_c3b07.2.2.1522028920562.0_21_67
To: <sips:888555817@10.64.19.70:5061>
```

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 10.64.19.70 (SIL-ATT); Log Settings - C:\Users\...\sysmonitorsettings.ini
File Edit View Filters Status Help

call identity 0 redirection number ''
referred_by '' refer_to '1027' domain ''
11:10:24 1364870121mS SIP Tx: TLS 10.64.19.70:4105 -> 10.64.91.41:5061
REFER sip:10.64.91.41:5061;transport=tls SIP/2.0
Via: SIP/2.0/TLS 10.64.19.70:5061;rport=4105;branch=z9hG4bK360ed6b2497623f9c1878e52f72f0ffe
Route: <sip:10.64.91.41:5061;ipcs-line=411596;lr;transport=tls>
From: <sips:888555817@10.64.19.70>;tag=385fad138835c83f
To: <sips:Anonymous@10.64.91.41>;tag=8933012401945208_c2b08.2.2.1522026860657.0_26_93
Call-ID: 53b4085e203fb47817410a2955bc2414
CSeq: 3 REFER
Contact: <sip:888555817@10.64.19.70:5061;transport=tls>
Max-Forwards: 70
Allow: INVITE,ACK,CANCEL,OPTIONS,BYE,INFO,REFER,NOTIFY
Supported: timer
User-Agent: IP Office 10.1.0.1.0 build 3
Content-Length: 0
Refer-To: <sip:1027@10.64.91.41:5061;transport=tls>

11:10:24 1364870123mS CMCallEvt: CMEndpoint::CheckAndPerformBlindTransfer return true
11:10:24 1364870123mS CMCallEvt: CheckAndPerformBlindTransfer Success
11:10:24 1364870123mS CMCallEvt: 0000000000000000 0.1991.0 -1 TargetingEP: StateChange: END=X CMCSOffering->CMCSDelete
11:10:24 1364870123mS CMCallEvt: 0000000000000000 0.1991.0 -1 BaseEP: DELETE CMEndpoint f179f4dc TOTAL NOW=4 CALL_LIST=1
11:10:24 1364870123mS CMCallEvt: 0a01346000007c5 0.1989.0 -1 BaseEP: DELETE CMEndpoint f173c714 TOTAL NOW=3 CALL_LIST=1
11:10:24 1364870125mS CMMap: a=21.11 b=0.0 PCGS CPReserveCodec (pcp[473]b0r1) true
11:10:24 1364870126mS CMMap: PCG:UnmapBChan pcp[473]b0r1 cp_b f5603c7c other_cp_b f55faa64
11:10:24 1364870126mS CMMap: a=21.11 b=2.18 M02
11:10:24 1364870126mS H323Evt: SESS f4 SetOperational local 10.64.19.70:49152 remote 10.64.91.41:16598 to 0
11:10:24 1364870126mS CMMap: PCG:UnmapBChan pcp[23]b1r0 cp_b f55faa64 other_cp_b 0
11:10:24 1364870127mS CD: CALL: 15.1987.1 BState=Idle Cut=3 Music=0.0 Aend="Line 15" (0.0) Bend="" [Line 15] (0.0) CalledNum=71027 () CallingNum=Anonymous@10.64.91.41 ()
11:10:24 1364870166mS RES: Fri 20/4/2018 11:10:24 FreeMem=58228012 Heap=57550076 (7) Cache=677936 MemObjs=12410 (Max 14126) CMMag=7 (9) ASN=0 Buff=5200 1362 1000 7443 5 Lir
1943/01.08%/0/02.85% MCR=0 MCW=0 DEV=0
11:10:24 1364870166mS RES2: IP 500 V2 10.1.0.1.0 build 3 Tasks=55 RTEngine=0 CMRTEngine=0 ExRTEngine=0 Timer=13+91 Poll=0 Ready=0 CMReady=0 CMQueue=0 VPNNQueue=0 Monitor
11:10:24 1364870166mS RES4: XML MemObjs=8 PoolMem=5796992 (3) FreeMem=5784872 (0) HeapUsed=12120
11:10:24 1364870166mS RES5: CLog MemObjs=344 FreePoolMem(Objs)=2912 (56) TotalMem=20800 StringsTotalMem=103600
11:10:24 1364870205mS SIP Rx: TLS 10.64.91.41:5061 -> 10.64.19.70:4105
SIP/2.0 202 ACCEPTED
From: <sips:888555817@10.64.19.70>;tag=385fad138835c83f
To: <sips:Anonymous@10.64.91.41>;tag=8933012401945208_c2b08.2.2.1522026860657.0_26_93
CSeq: 3 REFER
Call-ID: 53b4085e203fb47817410a2955bc2414
Contact: <sips:10.64.91.41:5061;transport=tls>
Record-Route: <sips:10.64.91.41:5061;ipcs-line=411596;lr;transport=tls>
Supported: replaces
Via: SIP/2.0/TLS 10.64.19.70:5061;rport=4105;branch=z9hG4bK360ed6b2497623f9c1878e52f72f0ffe
Content-Length: 0

11:10:24 1364870210mS SIP Rx: TLS 10.64.91.41:55410 -> 10.64.19.70:5061
NOTIFY sips:888555817@10.64.19.70:5061;transport=tls SIP/2.0
From: <sips:Anonymous@10.64.91.41>;tag=8933012401945208_c2b08.2.2.1522026860657.0_26_93
To: <sips:888555817@10.64.19.70>;tag=385fad138835c83f
CSeq: 3 NOTIFY
Call-ID: 53b4085e203fb47817410a2955bc2414
Contact: <sips:10.64.91.41:5061;transport=tls>
```

9. Conclusion

As illustrated in these Application Notes, Avaya IP Office Release 10.1, and the Avaya Session Border Controller for Enterprise Release 7.2 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 10.1, Deploying Avaya IP Office Servers as Virtual Machines*, Document Number 15-601011, Issue 05g, July 2017
- [2] *IP Office™ Platform 10.1, Deploying Avaya IP Office™ Platform IP500 V2*, Document Number 15-601042, Issue 32g, Aug 2017
- [3] *Administering Avaya IP Office™ Platform with Manager*, Release 10.1, June 2017
- [4] *IP Office™ Platform 10.1, IP Office SIP Phones with ASBCE*, Issue 02b, July 2017
- [5] *Deploying Avaya Session Border Controller in Virtualized Environment*, June 2017
- [6] *Administering Avaya Session Border Controller for Enterprise*, June 2017

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

- [7] *Application Notes for Avaya IP Office Release 10.1, Avaya Session Border Controller for Enterprise Release 7.2 with the AT&T IP Toll Free Service – Issue 1.0*

AT&T IP Transfer Connect Service:

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

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